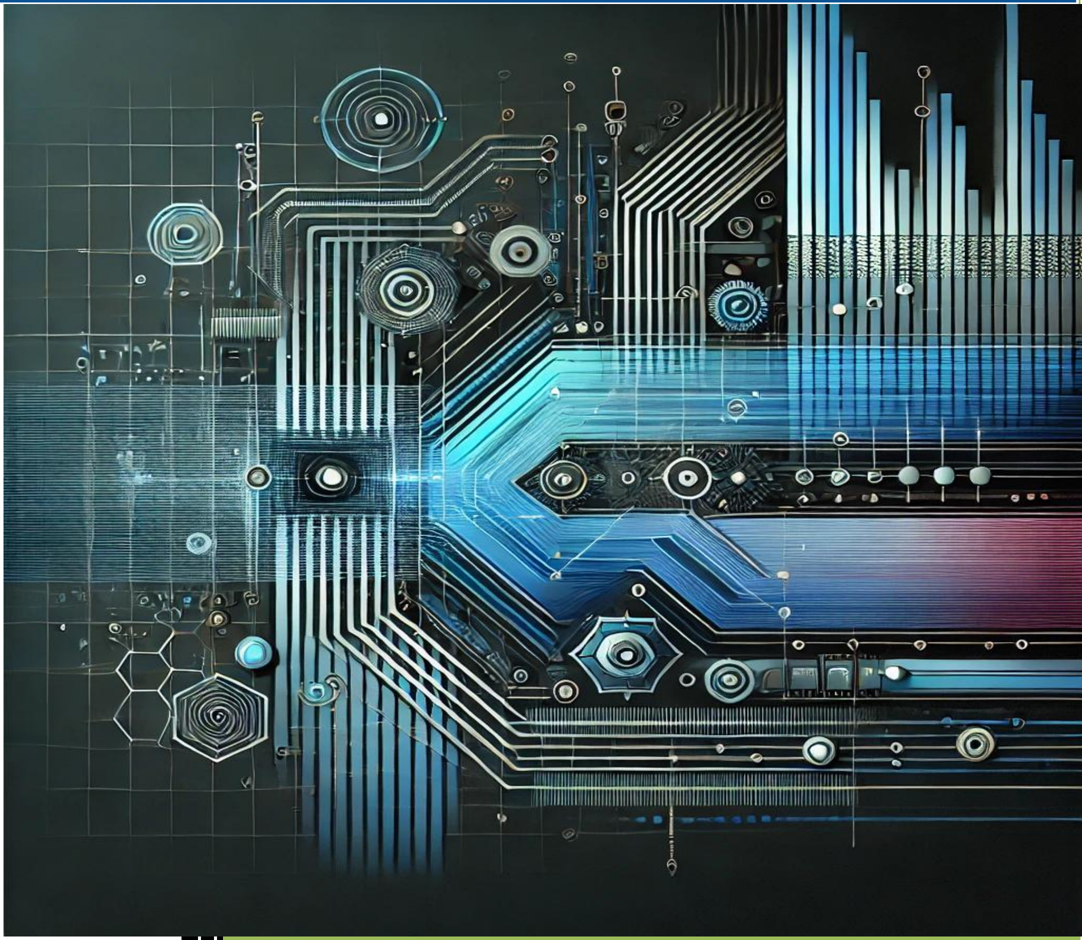
Advanced audio system setup 4.0



Grigorenko Aleksei

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Preface

The impressive evolution of automotive audio systems is associated not only with advancements in hardware but also with the emergence of refined scientific approaches to designing and tuning sound within the limited space of a vehicle cabin. The complex acoustic environment of an automobile—with its heterogeneous surfaces, resonances, and reflections—compels researchers and engineers to continually seek methods for optimizing and enhancing sound quality. This situation necessitates not only the proper selection of speakers and amplifiers but also the development of an integrated concept of sound creation, in which every stage—from the mathematical modeling of sound wave propagation to the final listening experience—is imbued with a scientific dimension rather than merely a practical one. A systematic analysis of the issue reveals the emergence of a unique interdisciplinary field that combines electroacoustic principles, psychoacoustic effects, engineering calculations, and aspects of digital signal processing. Consequently, a new type of automotive audio is being established, one whose objective is not solely to achieve high volume or powerful bass but to create an in-cabin audio stage capable of delivering musical imagery with exceptional precision and depth. The present work aims to systematize modern approaches to addressing this complex set of challenges. The integration of a fundamental understanding of in-cabin processes with the technological and theoretical tools required for precise control of tonal balance, filtering, and time correction represents the key to achieving a new level of performance. The monograph reflects the most current methodologies and experiments, culminating in the development of specialized tuning algorithms that encompass every link in the audio chain.

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Grigorenko Aleksei is a recognized expert in the field of car audio and tuning, with over a decade of experience in implementing large-scale projects in creating, promoting, and distributing car audio brands. His entrepreneurial journey began in St. Petersburg, where he built a solid foundation through his education at School 337 and SPBIGO, nurturing his management and creative skills.

As the Founder and CEO of Talex Audio LLC, Aleksei successfully introduced the Pride Car Audio brand to the US market by developing effective marketing strategies, optimizing supply chains, and establishing a reliable distribution network. His expertise also extends to launching distribution centers in Tallinn and playing a pivotal role in the early development of the brand in the Northwestern Federal District of Russia.

Aleksei continuously drives innovation in audio systems, combining deep technical insight with practical management and strategic planning abilities. His proven track record in developing and implementing audio solutions has enabled him to create and successfully exit multiple ventures, leaving a lasting impact on the industry.

This monograph, “Advanced audio system setup 4.0,” is a comprehensive guide built upon his extensive experience and knowledge, designed for professionals seeking to integrate cutting-edge audio systems and innovative solutions into their projects.

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INTRODUCTION

Modern automotive audio systems constitute highly complex assemblies, in which precise acoustic tuning demands thorough consideration of physical principles, psychological factors of sound perception, and engineering design solutions. Unlike conventional stationary systems, the automotive environment imposes significant constraints: limited cabin volume, intricate geometry (including the body, windshield, door surfaces, and trim), the close proximity of listeners to the speakers, and a high degree of reflections. All these factors render the task of constructing high-quality sound not merely an engineering challenge but also a subject of scientific investigation, requiring a deep understanding of the physics of sound production, computer modeling techniques, and psychoacoustic principles.

The monograph is devoted to advanced tuning of 4.0 audio systems and covers the key stages of working with sound in an automobile—from design and component selection to the fine adjustment of frequency bands and final sound evaluation. The objective of the study is not only to describe the principles and tuning tools but also to present a comprehensive view of how various aspects (system configuration, gain, filtering, time correction, and tonal balance) interact holistically in the creation of a high-caliber automotive audio system.

It is important to recognize that the automotive audio industry is continuously evolving: component manufacturers are introducing new types of speakers, amplifiers, and digital processors; vehicles are becoming more sophisticated in both body construction and in-cabin electronics. This evolution creates a demand for a universal yet highly accurate tuning methodology that accounts for all factors—acoustic, electrical, and ergonomic.

The monograph examines modern design approaches that incorporate not only empirical methods but also elements of theoretical modeling of resonances, reflections, and standing waves within the cabin. Detailed analyses of fine-tuning instruments are provided, and recommendations for channel alignment, gain correction, filtering, and time delays are formulated with consideration of complex psychoacoustic phenomena.

Ultimately, the comprehensive development and implementation of approaches to advanced tuning of 4.0 audio systems enable the realization of the musical potential even under the challenging conditions of an automotive cabin. The chapters sequentially illustrate the path from design decisions and initial installation to the precise adjustment of gains, filters, time delays, and tonal balance. This approach is aimed at achieving an audio stage that closely approximates the ideal, with transparent and detailed sound reproduction, while avoiding critical errors that could lead to the rapid failure of expensive components.

CHAPTER 1: CONFIGURATION IN SYSTEM DESIGN

1.1 Objectives and challenges in designing an automotive audio system

Designing an automotive audio system requires careful consideration of numerous factors that are often absent in conventional stationary acoustic systems. The confined space and complex geometry of a car's interior—comprising plastic, metal, glass, and soft upholstery—introduce distinct acoustic effects:

- Standing waves and localized resonances in the low-frequency range. Due to the short distance between parallel surfaces (such as the floor and roof), bass frequencies may experience significant amplification or attenuation.
- Reflections and multiple reflections from the windshield, side windows, and dashboard. These interactions lead to phase shifts, comb filtering effects, and distortions in soundstage perception.
- Limited listening zone. In most cases, the system is tuned for the driver (and sometimes the front passenger), necessitating specific speaker placement and time alignment adjustments.

Thus, the primary prerequisite for system design is a deep understanding of sound propagation physics within the car cabin. Ideally, this process is based on simulation, considering material properties and the precise dimensions of the vehicle. However, in practice, empirical methods (experiments and measurements) are often combined with theoretical principles.

To assess sound quality within a vehicle, the industry relies on a set of key parameters recognized by professionals, including competitive car audio communities:

1. Accuracy and naturalness of reproduction. This entails the absence of perceptible distortions, a smooth (or predictably adjusted) frequency response, and tonal balance that aligns with reference recordings.

2. Soundstage formation:

○ Perceived distance to the soundstage. The farther the sound source appears to be from the listener, the more comfortable the auditory experience.

○ Width and height of the soundstage. Ideally, the soundstage extends beyond the windshield and aligns with the listener's eye level.

○ Depth (layering). The system should effectively convey the sense of near and distant planes in the music.

○ Localization and focus of sound sources. Each instrument and vocal element should have well-defined boundaries and a stable position within the virtual soundstage.

3. Dynamic range and detail resolution. The system must accurately reproduce both quiet and loud passages. Detail resolution refers to the ability to convey subtle nuances, from the softest cymbal decay to the intricate overtones of a vocal performance.

Each of these criteria is largely determined by the system's architecture, including the selection of speaker types, their placement, as well as the accuracy of preliminary acoustic calculations and subsequent tuning.

In practice, simply installing high-end components is insufficient; without meticulous design planning, even the most expensive speakers and amplifiers may produce significant distortions. The design phase involves:

● Defining the system's configuration (e.g., two-way or three-way setups; see Figures 1, 2, 3).

- Determining the optimal speaker placement (e.g., doors, pillars, dashboard, trunk).
- Anticipating and mitigating cabin-specific acoustic challenges such as resonances, reflections, and vibrations.

Beyond acoustic considerations, system design must also address practical aspects, including ergonomics (avoiding obstructions to visibility and controls), safety (ensuring secure installation of heavy components), and maintaining cabin functionality. In the context of competitive audio systems, design plays a crucial role, as expert evaluations (by judges) take into account the system's ability to create an ideal soundstage and achieve precise tonal balance at the designated listening position.

A well-structured design phase enhances everyday listening comfort and minimizes the need for extensive manual adjustments. Additionally, a professionally executed project lays the groundwork for subsequent fine-tuning stages, including gain structure optimization, filtering, and time alignment (as discussed in later chapters).

1.2 Optimal system configuration: selection and arrangement of acoustic components

For comprehensive reproduction of the full range of audible frequencies—from the sub-bass region (approximately 16 Hz) to the highest overtones (20 kHz)—automotive audio systems incorporate multiple types of speakers. The table 1 below summarizes the basic frequency coverage and functional roles of the primary components in an audio system.

Table 1. Basic frequency coverage and functional roles of primary audio system components

Speaker Type	Approximate Frequency Range	Primary Function
Subwoofer	16–63 Hz	Deep low frequencies (sub-bass)
Woofer	63–250 Hz	Lower bass, fullness in the "warm bass" region
Midwoofer	~60 Hz – 2–3 kHz	Covers bass and part of the midrange
Midrange Speaker	~250 Hz – 4 kHz	Vocal range, instrumental detail
Mid-tweeter	500–1000 Hz – 20 kHz	Combined mid-high range, broadband approach
Tweeter (High-Frequency Speaker)	~3–4 kHz – 20 kHz	Upper frequency range, "airiness," detail resolution

The key differences among speaker types lie not only in their frequency response but also in their structural characteristics:

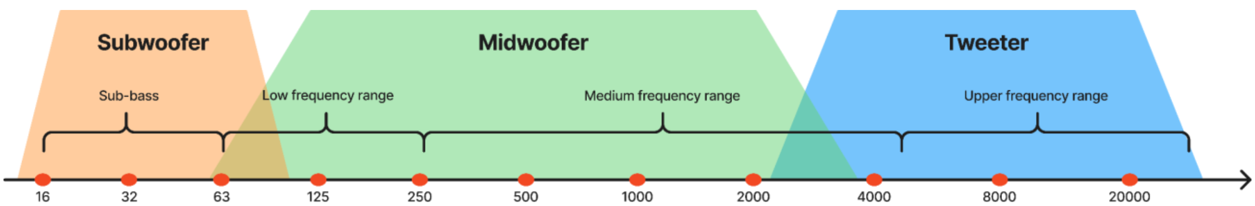
- Rigid cones (typically conical in shape) exhibit high directionality, perform well at lower frequencies, but may introduce dispersion issues in the high-frequency range.
- Soft/dome diaphragms have a broader dispersion pattern and are commonly used in midrange and tweeter applications, where spatial diffusion and sound "airiness" are crucial.

During the design phase, the optimal combination of components is selected based on system objectives and budget. It is essential to consider how high (or low) each speaker can operate without distortion, as well as its output capacity.

A comparative analysis of two-way and three-way configurations is presented below.

1. Two-Way Systems – The most common configuration consists of a midwoofer + tweeter (Figure 1):

- Advantages: Simplified installation, fewer components, and a more cost-effective solution.
- Disadvantages: The midwoofer must cover a broad frequency range (up to 2–3 kHz), while the tweeter must extend significantly below 4–5 kHz, requiring both speakers to exhibit highly linear performance. Additionally, when midwoofers are installed in lower positions (such as in doors), there is often a perceived "collapse" of the soundstage at the edges and reduced accuracy in the midrange.



(Figure 1. Two-Way Systems – Midwoofer + Tweeter)

Other two-way configurations exist, such as a woofer + mid-tweeter setup (Figure 2), where the woofer handles bass frequencies (up to approximately 500–1000 Hz), and the mid-tweeter covers the remaining spectrum. This arrangement provides greater flexibility in the low-frequency range, allowing for the use of a more robust subwoofer designed for deep sub-bass reproduction, while the woofer extends into the region where the subwoofer's response tapers off.

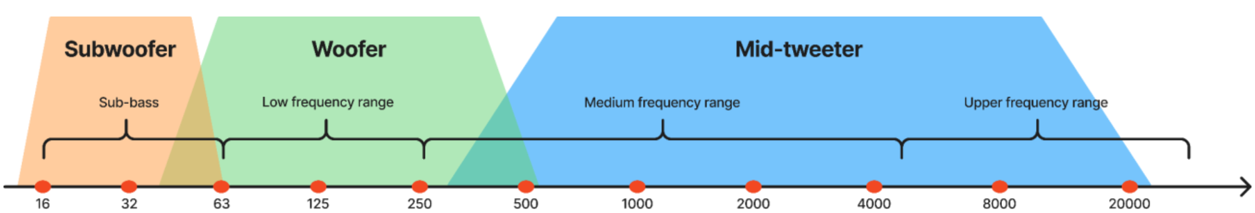
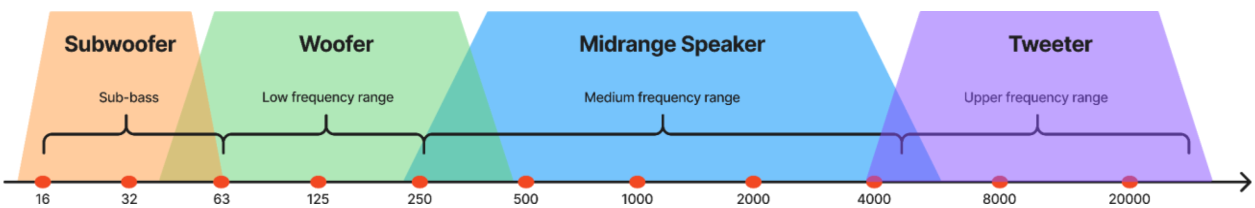


Figure 2. Two-Way Systems – Woofer + Mid-Tweeter

2. Three-Way Systems. A more balanced configuration consists of a subwoofer, woofer, midrange speaker, and tweeter (Figure 3). In some cases, the subwoofer is considered separately, and the front stage is implemented as woofer/midwoofer + midrange speaker + tweeter:

- Advantages: Each speaker operates within a narrower frequency range, reducing the risk of overload and enabling more precise integration of frequency bands. When properly installed, this configuration yields a highly elevated and detailed soundstage.
- Disadvantages: Higher cost, more complex installation (requiring an additional mounting point for the midrange speaker), and increased difficulty in crossover and time-alignment calibration.



(Figure 3. Three-Way Systems – Subwoofer, Woofer, Midrange Speaker + Tweeter)

From a practical standpoint, the choice between two-way and three-way systems is typically determined by two factors: the pursuit of uncompromising sound quality and the availability of resources (time, budget, and willingness to modify the vehicle's interior extensively).

A mid-tweeter is a specialized driver that combines midrange and high-frequency reproduction (typically covering 500–1000 Hz to 20 kHz).

- Advantages: Reduces the number of required speaker mounting points (one driver instead of two), potentially achieving seamless frequency integration without additional crossover transitions.
- Disadvantages: Requires precise placement to maintain a smooth frequency response at higher frequencies and lacks independent level adjustment for mids and highs. If the mid-tweeter exhibits excessive sharpness or dullness in certain regions, it is difficult to correct solely through equalization.

Full-range speakers (covering approximately 100–150 Hz to 15–20 kHz) are sometimes used in "mini mid-high" configurations for compact installations where simplicity and space efficiency are priorities. However, achieving audiophile-grade sound with full-range drivers in a car is challenging due to their limited low-frequency extension, restricted output levels, and higher distortion levels.

1.3 Acoustic characteristics of the vehicle cabin and reflection management

The interior of a vehicle is a highly enclosed space where sound waves undergo multiple reflections from rigid surfaces. This leads to several types of distortions:

1. Standing waves at low frequencies. These occur when reflected waves interfere with the original sound wave, creating alternating peaks and dips in the frequency response. In the compact space of a vehicle, this effect is amplified due to the close proximity of all surfaces, often resulting in "boomy" bass (a pronounced peak at a specific frequency) or a drop-off at another frequency.
2. Comb filtering effect. At mid and high frequencies, reflected waves (e.g., from the windshield) reach the listener with a slight delay relative to the direct sound, causing phase cancellation (interference). As a result, the frequency response

exhibits alternating dips and peaks (resembling the teeth of a comb), which distorts tonal balance, particularly in the critical vocal range (1–4 kHz).

3. Localization distortion. Excessive reflections can cause the soundstage to become blurred or shifted, especially when the listener perceives more reflected than direct sound. This significantly degrades spatial perception, which is a key focus in precise audio tuning.

The vehicle cabin consists of various surfaces, each with specific reflection and absorption properties:

- Glass (windshield, side windows): Highly reflective, particularly for mid and high frequencies.
- Metal body panels: Also serve as reflective surfaces and may resonate at specific frequencies.
- Plastic panels (dashboard, door trims): Partially attenuate midrange frequencies but can generate unwanted vibrations under certain conditions.
- Soft materials (seats, carpeting): More effectively absorb high frequencies, which can create an overall imbalance if not compensated for.

The design of the doors is particularly critical when speakers are mounted within them:

- The internal volume of a door is far from an "ideal" speaker enclosure. Metal vibrations and plastic panels contribute to additional resonances.
- Placing a speaker at the lower part of the door enhances bass response but can lead to soundstage collapse and midrange issues.

Managing cabin acoustics during the design phase involves several key approaches. First, the primary axis of speaker radiation (especially for mid and high-frequency drivers) should be directed toward the listener to minimize initial reflections from the windshield and side panels.

In cases where reflection-based techniques are used (e.g., directing sound toward the ceiling or windshield to create a more distant soundstage), it is important to avoid situations where the listener simultaneously perceives both direct and reflected sound from the same speaker.

The next step involves applying damping materials to doors, the floor, and the roof. This reduces resonances, improves bass control, and minimizes unwanted reflections within the door cavity. Soft materials in the cabin help reduce high-frequency "echo," but this can shift tonal balance, so high-frequency compensation should be considered in the system design, typically through equalization.

Finally, digital correction plays a crucial role. Time alignment (TA) compensates for varying distances between speakers and the listener's ears. During the design phase, these distances should be estimated to ensure that speakers can be time-aligned effectively.

Equalization helps smooth peaks and dips in the frequency response. However, relying solely on an equalizer is not advisable—structural issues, such as early reflections, should be addressed through design modifications rather than being artificially suppressed with EQ adjustments.

1.4 Speaker placement and orientation: soundstage design

Designing a soundstage in a vehicle involves creating a virtual "space" in front of the listener, ensuring that each instrument and vocal element has a distinct position and realistic depth. The soundstage should feel distant yet seamlessly integrated with the cabin and sufficiently wide to extend beyond the windshield pillars rather than being confined between them.

Several key parameters must be considered during the design phase:

- Soundstage depth. The farther the perceived soundstage, the more comfortable it is for long-term listening.

- Width. In high-end systems, the soundstage often extends beyond the cabin's physical boundaries, while in simpler setups, it is typically limited to the area between the windshield pillars.
- Height. The optimal level is generally at or near the listener's eye level to prevent sound from being perceived as coming from the floor or the ceiling.
- Depth and layering. A carefully balanced mix of direct and reflected sound, further refined by DSP adjustments, creates a perception of layering and proper placement of musicians on an imaginary stage.
- Localization and focus. Precise positioning (left, center, right) and clear sound source definition are essential. Vocals should not blur, and instruments should not be duplicated or misplaced.

Achieving all these characteristics simultaneously requires a comprehensive approach to speaker placement and orientation. Any shift or rotation may enhance one aspect of the soundstage while compromising another, such as reducing height accuracy or localization precision.

The vehicle cabin is a highly constrained space where even minor details influence sound wave propagation. During the design phase, compromises must be made between the ideal speaker placement and the actual geometric limitations of the cabin. The table 2 below presents a comparison of three commonly used installation zones for different frequency ranges in the front stage of an audio system.

Table 2. Typical front speaker placement zones and their acoustic characteristics

Parameter	**Lower Door Section**	**Upper Door/Pillars**	**Dashboard (Upward Orientation)**

Primary Use	Midwoofer (especially woofer in a three-way front stage)	Midwoofer in two-way systems, midrange/tweeter in three-way setups	Experimental placement for midwoofers/midrange drivers
Acoustic Advantages	Natural bass reinforcement due to floor coupling, allows larger speaker diameters	Higher soundstage, less side collapse, suitable for mid and high frequencies due to proximity to ear level	Creates a distant and elevated soundstage via windshield reflections, may benefit from large internal cabin volume in some vehicles
Acoustic Disadvantages	Soundstage collapse at the edges, difficulty reproducing midrange accurately	Reduced bass output (speaker farther from the floor), more complex installation	Unpredictable tonal balance due to reflections, challenges in isolating direct sound and avoiding comb filtering
Recommendations	Suitable for woofers/midbass drivers with a wide frequency range, moderate angling is beneficial	Optimal for midrange and tweeters, requires careful dispersion pattern calculations	Requires experimentation with "windshield reflections," suitable only with advanced DSP tuning

As shown in the table, there are significant differences between the conventional placement of midbass drivers in the lower door section and the installation of speakers in the dashboard or pillars. Higher placement naturally elevates the soundstage but typically reduces bass depth, while utilizing windshield reflections can create an exceptionally distant stage but introduces challenges such as comb filtering.

In a two-way system (midwoofer + tweeter), the midwoofer must extend up to 2–3 kHz, while the tweeter must cover frequencies starting at approximately 3–4 kHz. This approach simplifies installation and reduces the number of required components but demands higher quality from both drivers and the amplifier.

In a three-way front stage (woofer + midrange + tweeter), each driver is responsible for a narrower frequency band, leading to more balanced load distribution and generally improved layering of the soundstage. However, the increase in physical speaker locations complicates phase alignment. If the midrange speaker placement is poorly executed, the soundstage may develop gaps or unwanted reflections.

Mid-tweeters, capable of covering frequencies from 800–1000 Hz to 20 kHz, present an alternative solution, effectively replacing the midrange-tweeter combination. While compact, this setup requires precise speaker orientation toward the listener to maintain clarity in upper midrange and high frequencies.

Regardless of the chosen approach, it is crucial to define the system's objectives early in the design process—whether for competitive use, everyday listening, powerful bass reproduction, or a highly articulate midrange—and align these goals with the available budget and cabin constraints.

1.5 Criteria and stages of preliminary "project-level" tuning

The accuracy of the final tuning largely depends on how effectively frequency bands are allocated among the speakers during the design phase. If a speaker can reproduce frequencies from 80 to 3000 Hz without distortion, it is advisable to utilize this range as much as possible while considering that directivity will become significantly narrower at mid frequencies.

When determining frequency cutoffs, the designer considers:

- Real-world operating conditions. For example, rock music often requires increased output in the lower-midrange region (around 200–500 Hz).
- Diaphragm design characteristics. Midwoofers with rigid conical diaphragms often incorporate a phase plug to extend their usable frequency range.

- Excessive frequency overlap. If a midwoofer and midrange driver have significant overlap, phase interference (comb filtering effect) may occur at certain frequencies.

Preliminary frequency boundaries and recommended speaker orientation angles should be documented in technical specifications or at least outlined in sketches to prevent installation from turning into an endless trial-and-error process.

Even before component installation, preliminary calculations of time delays are made based on the cabin geometry. The formula for estimating sound travel time is:

$$t = d / v,$$

where d is the distance from the speaker to the listener's ear, and $v \approx 343$ m/s (at room temperature). If, for example, the right midwoofer is 20 cm farther from the listening position than the left one, the time difference in sound arrival will be approximately 0.58 ms.

Final phase alignment values are determined through actual measurements. However, preliminary estimates help ensure that the DSP processor in the system design has a sufficient delay adjustment range for each channel. If the subwoofer is positioned at a considerable distance, significant delays may be required for the front-stage speakers to maintain phase coherence across the frequency spectrum.

In addition to delays, the choice of crossover filters must be considered. Steeper slopes (e.g., 24 dB per octave) are more likely to introduce phase shifts in the transition zone, which must be compensated for either through speaker positioning or during fine-tuning.

After initial calculations, it is recommended to conduct preliminary test measurements. This is typically done using measurement microphones and specialized software (such as REW, ARTA, or CLIO). The objective at this stage is to identify key frequency response characteristics (such as peaks and dips) and assess the impact of reflections from glass or door panels based on the chosen speaker angles.

Additionally, it is beneficial to conduct an early-stage listening test with the system in a semi-configured state. It is often observed that a minor adjustment of a tweeter's angle—by as little as 5–10 degrees—can drastically alter the perceived soundstage, either improving or degrading the overall result in unpredictable ways. Therefore, preliminary "project-level" tuning provides valuable data for installation adjustments. If measurements indicate a significant dip at 2–2.5 kHz, repositioning the midrange speaker or selecting a different mounting point may be necessary.

1.6 Conclusions and recommendations for further tuning stages

A well-designed system layout significantly simplifies all subsequent steps, including gain adjustments, filter settings, and time alignment. If the system has a smooth or predictably adjusted frequency response by the time final tuning begins, digital signal processing (DSP) will serve only to refine the sound rather than correct fundamental design flaws.

In practical terms, this means that equalization can be minimal, and time delays will not need to exceed 15–20 cm in equivalent physical distance. Systems assembled without a structured design often suffer from numerous deep frequency dips and peaks, requiring aggressive equalization, which not only degrades sound quality but also reduces speaker longevity.

Practical recommendations for different configurations and budget levels:

- Two-way systems. It is advisable to select a midwoofer with an extended upper-frequency range (featuring a phase plug or a specially shaped diaphragm) and a tweeter capable of stable operation from 3 kHz. To preserve bass performance, placing the speaker in the lower section of the door is recommended. However, this placement requires compensation for soundstage collapse at the edges, which can be addressed by a moderate upward or inward speaker tilt.

- Three-way systems. These are optimal for those with sufficient resources (time, budget, and available cabin space) and a focus on high detail resolution. Positioning the woofer in the lower section and placing the midrange and tweeter on the pillars or near the side mirrors helps achieve a highly elevated soundstage. If crossover filters and dispersion patterns are properly accounted for, the result will be both a wide stage and precise localization of instruments.
- Mid-tweeter solutions. These are useful for saving space by eliminating a separate tweeter but require precise directional placement. This approach is suitable for systems where maintaining the original vehicle interior is a priority while improving mid and high-frequency performance.

Regardless of the system's complexity, success depends on a balanced approach that considers acoustic principles, ergonomic constraints, and the owner's aesthetic preferences.

Modern CAD tools (such as EASE and Car Audio Simulator) enable sound behavior modeling within a vehicle cabin, though only to a limited degree. Fully accounting for all reflective surfaces and material types still requires substantial computational power. Nevertheless, such software provides valuable guidance on frequency cutoffs and optimal speaker angles.

A key factor driving advancements in automotive audio in the near future is the improvement of automatic DSP tuning algorithms. However, it is essential to recognize that even the most advanced processor cannot compensate for fundamental design flaws in speaker placement. The most effective approach remains a structured process: accurate system design → high-quality installation → precise digital calibration.

Thus, system design establishes the foundation for future sound performance. When mathematical and engineering decisions are made wisely, the system will deliver clean, balanced, and reliable audio reproduction. Whether for competitive or everyday car audio applications, thorough preliminary planning significantly reduces the time and effort required for fine-tuning while ensuring the highest sound quality within the vehicle cabin.

CHAPTER 2: TUNING TOOLS

2.1 Overview of available tuning tools

A modern automotive audio system requires a set of tools to optimize its performance within a specific vehicle interior. Each tuning tool influences a particular aspect of the acoustic-electrical chain, and together, they form a comprehensive system essential for both protecting components and achieving high-quality sound reproduction. The key tuning parameters are categorized as follows:

1. Levels (at the head unit or processor). Proper level adjustment serves two main purposes: preventing amplifier overload (avoiding clipping) and ensuring appropriate volume balance between different channels (front, subwoofer, etc.). Most head units provide controls for overall volume level, subwoofer level, and in processor-based systems, individual level settings for each frequency band. Some systems also include Source Level Adjustment (SLA) to balance volume levels across different input sources (radio, USB, Bluetooth, etc.), ensuring a consistent listening experience when switching between them.

2. Gain (amplifier input sensitivity). Gain determines the input voltage level at which the amplifier reaches its maximum (or near-maximum) power output. Setting the gain too high results in premature clipping, while setting it too low reduces dynamic range and increases unwanted noise and interference. Correct gain adjustment is crucial for protecting speakers from mechanical and thermal damage while also maintaining tonal balance.

3. Filters (active/passive, HPF/LPF/Bandpass). Every speaker operates within a limited frequency range. To prevent speakers from attempting to reproduce frequencies beyond their capabilities, filters are used: high-pass filters (HPF), low-pass filters (LPF), and bandpass filters. These can be implemented in the head unit's

processor, a DSP, or passive crossovers (using inductors and capacitors). Their primary function is to remove unwanted frequencies from the signal, protecting speakers from overloading and distortion.

4. Time Alignment. Due to the asymmetry of a vehicle's cabin, the distance between each speaker and the listener varies significantly. To synchronize the arrival of sound waves and improve imaging—enhancing depth, width, and precise positioning of vocals and instruments—time alignment is applied. This adjustment compensates for the varying distances between speakers and the listening position, ensuring accurate phase coherence.

5. Equalizer (frequency response adjustment). Equalization allows precise control over specific frequency bands. It can be implemented as a graphic equalizer (with 3 to 31 bands) or a parametric equalizer, where the user defines the center frequency, bandwidth (Q), and level. Equalization helps correct cabin-induced acoustic issues (such as resonances, standing waves, and reflections) and fine-tune the sound to match listener preferences. However, improper equalization can complicate the overall sound balance and may even cause excessive strain on certain drivers.

2.2 Minimum setup requirements for safe initial power-up

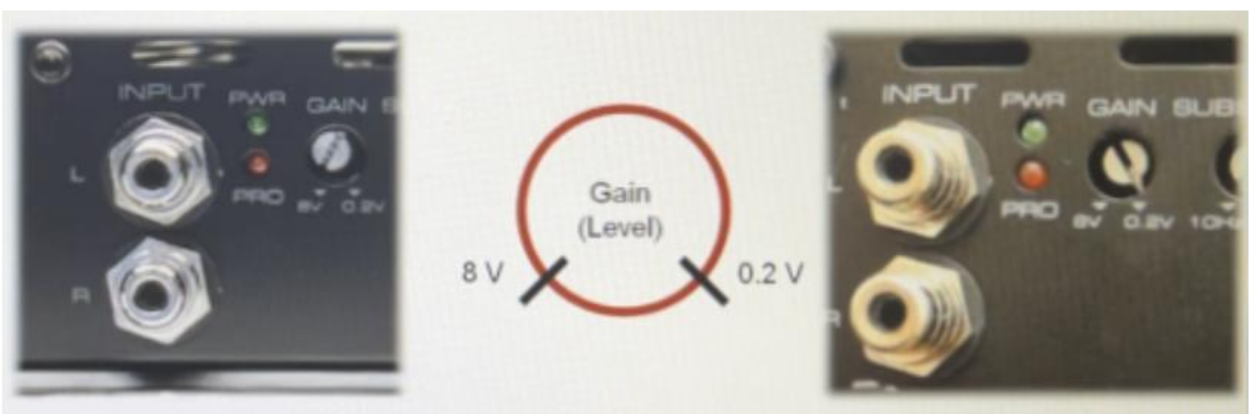
Before sending a signal to the speakers and increasing the volume, several critically important steps must be performed. Errors in this basic configuration can lead to severe consequences for the audio system components, ranging from coil overheating to mechanical damage to the speaker cones.

One of the most common causes of speaker failure in automotive audio systems is amplifier clipping. If the volume level (or processor output level) is set too high while the amplifier gain is overly sensitive (set to maximum), the signal quickly reaches amplitude limitations. In such cases:

- A portion of the waveform is transformed into a "clipped" square impulse;
- Excess energy is converted into heat, causing coil overheating;
- The mechanical limits of the speaker may be exceeded, leading to suspension deformation or severe damage.

Therefore, minimizing the risk of clipping during the initial power-up is crucial. This means:

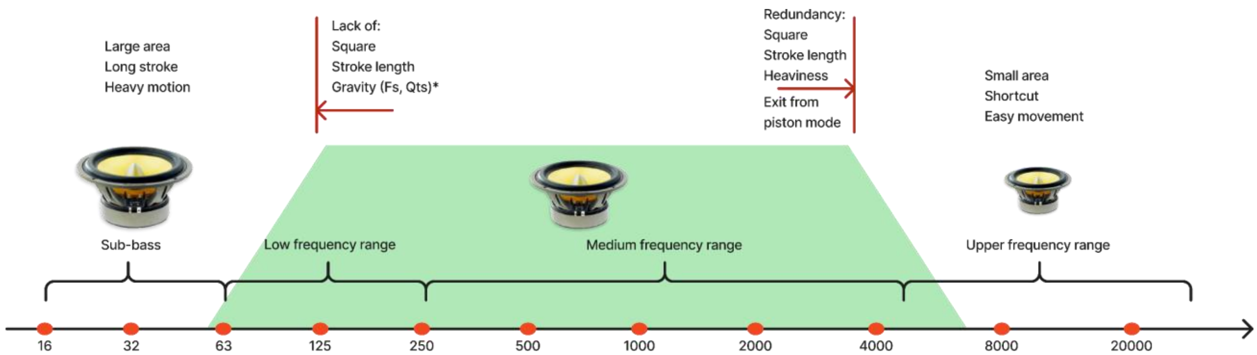
1. Setting the overall output level (Master Volume, SLA, etc.) to a neutral or slightly reduced position.
2. Adjusting the gain control on the amplifiers to the minimum setting (typically corresponding to an input signal level of 4–8 V) (Figure 4).
3. Ensuring that additional enhancement features (Loudness, Bass Boost) are not activated.



(Figure 4: Input Sensitivity – Left: Low; Right: High)

This approach provides a level reserve, allowing a gradual increase in volume and fine-tuning of gain until a stable and clean sound is achieved without distortion.

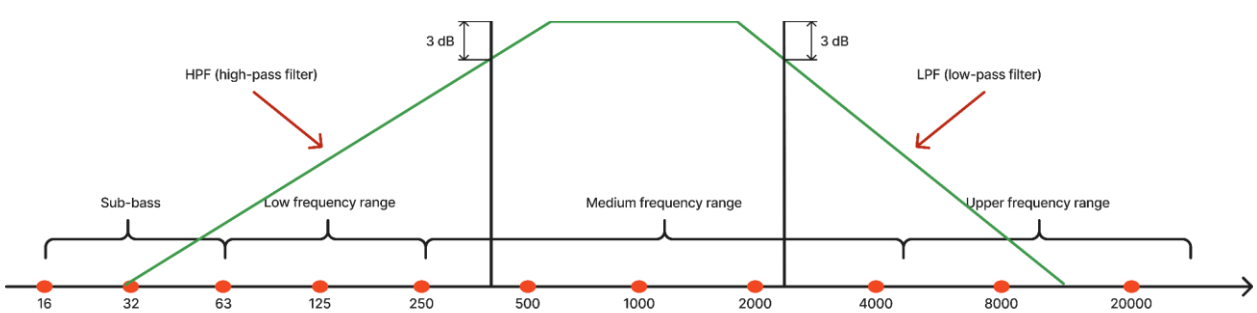
The second crucial aspect of safe setup is the use of filters (crossovers). If a midwoofer or midrange speaker receives a signal significantly below its resonance frequency (fs), it may reach uncontrolled excursion, posing a risk of mechanical failure. Similarly, a tweeter cannot reproduce frequencies significantly lower than 2–4 kHz (depending on the model). Attempting to force it to play lower midrange frequencies may cause severe damage to the dome or voice coil.



(Figure 5: Filters: Limitations of spectrum capabilities)

To ensure proper operation, each speaker should have an appropriately set filter:

- HPF (high-pass filter) – Used for midwoofers, midrange drivers, and tweeters to define their lower frequency limit.
- LPF (low-pass filter) – Applied to subwoofers or woofers to prevent excessive midrange output and avoid frequency overlap.
- Bandpass filter – Used when a speaker operates as a midrange driver in a three-way front stage.



(Figure 6: Filter Cutoff Frequency)

At the same time, the filter slope (crossover order: 6/12/18/24 dB per octave) should be set sufficiently steep (not lower than 12 dB per octave for HPF) to sharply limit unwanted low frequencies and protect the driver. For subwoofers in a ported enclosure, an HPF with a slope of 24–36 dB per octave is typically set slightly below the port tuning frequency.

Common mistakes before the first system power-up include:

- Setting the gain to maximum without considering actual power handling and input signal voltage.
- Failing to apply or incorrectly setting the HPF for mid and high-frequency drivers (especially dangerous for tweeters).
- Setting the subwoofer cutoff frequency too low or omitting a subsonic filter (in a ported enclosure, this can lead to uncontrolled cone excursion).
- Enabling enhancement features (Bass Boost, Loudness) at high volume levels, which results in partial clipping even at moderate volume settings.

Each of these mistakes risks damaging expensive speakers or amplifiers, making proper adherence to this "minimum setup" a mandatory step rather than an optional precaution.

2.3 Level configuration in the head unit and processor

Once the basic filters and gain settings have been adjusted to a safe level, more detailed level management within the head unit (HU) or external processor can be performed. Different manufacturers and models introduce their own terminology and functionality (SLA, Sub Level, Tone Control, etc.), but the adjustments generally fall into two categories: general levels (Master Volume, SLA) and specialized controls (subwoofer, front and rear channels, equalizer).

2.3.1 General level (SLA) and source comparison

Modern head units support multiple audio sources, including radio, USB/SD, Bluetooth, and AUX. Each source may have a different default output volume, which can cause discomfort when switching tracks, as the volume may suddenly increase or decrease.

To eliminate these inconsistencies, Source Level Adjustment (SLA) is used. For example, the radio may be set as the reference volume at 0 dB, while a USB drive may be adjusted to -3 dB if it plays too loudly. Conversely, if the USB signal is noticeably quieter, it can be increased to +2 dB. This method ensures consistent perceived volume across different sources.

Some systems offer Automatic Sound Leveling (ASL), which is intended to automatically balance volume levels when switching between sources. However, in practice, the results can be unpredictable. For complex systems focused on sound quality, manual fine-tuning is usually more effective.

2.3.2 Independent subwoofer and frequency band levels (in processor-based systems)

Most head units and DSP processors provide an option for independent subwoofer level control (Sub Level). This feature is particularly useful, as it allows quick adjustments to low-frequency intensity depending on the type of music being played. However, during initial setup, the following points should be considered:

- The subwoofer level should never be increased beyond the zero reference point, unless the HU/processor model explicitly supports specific threshold values (an exception exists in some older Pioneer models, where +6 dB on the subwoofer level corresponds to an actual 0 dB).

- If additional bass is needed, it is preferable to reduce the front speaker volume (or overall volume) rather than increasing the subwoofer level. This minimizes the risk of clipping in the low-frequency channel.

Processor-based systems allow for more refined control: in addition to the subwoofer, independent level adjustments for midbass, midrange, and tweeters are available. In some cases, even left and right speakers can be adjusted separately, helping to correct minor cabin asymmetries. While this approach offers greater flexibility, it also increases the risk of errors if the fundamental rule is not followed: "avoid boosting levels."

2.3.3 Tone compensation and "enhancement" features (LOUD, Bass Boost, etc.)

In addition to direct level control, head units may include additional sound enhancement features:

- Loudness (tone compensation). At very low listening volumes, the human ear is less sensitive to low and high frequencies, and Loudness compensates for these dips. However, at medium and high volumes, tone compensation often leads to exaggerated bass and treble, distorting the tonal balance.
- Bass Boost. This function acts as an equalization boost at a selected low-frequency range (usually 30–50 Hz). Excessive boosting significantly increases the risk of clipping.
- Compressed audio file enhancement (MP3 Enhancer, Sound Retriever, etc.). These functions attempt to improve the quality of compressed files but can introduce artifacts, often with questionable effectiveness. A better approach is to use high-quality (lossless) recordings or at least MP3 files with a high bitrate.

For proper system calibration, enhancement features and tone compensation are typically disabled. They may only be used under specific conditions (such as background listening at very low volumes), but even then, caution is required, as these settings can alter the overall balance and potentially overload certain frequency bands at signal peaks.

2.4 Managing amplifier input sensitivity (Gain)

The input sensitivity of an amplifier (commonly referred to as "Gain") is one of the most critical parameters that determines at what input signal level the amplifier will reach its maximum allowable output power. Incorrect gain settings can cause a range of issues, from partial signal clipping to excessive noise and interference if set too low.

Amplifier manufacturers often label the gain control with numerical values indicating the input voltage required for maximum power output. As illustrated in Figure 4:

- High sensitivity (Gain turned to the right, small numerical value, e.g., 0.2 V) means that the amplifier requires only a small input voltage to reach full power. This can be risky—if the output from the head unit or processor is too high, the amplifier will quickly enter clipping mode.
- Low sensitivity (Gain turned to the left, large numerical value, e.g., 8 V) means that the amplifier requires a higher input signal level. If the input voltage is too low, the sound may be too quiet, and the signal-to-noise ratio may degrade.

However, in practical installations, the absolute numerical value is less important than ensuring proper matching between the source (processor) output level and the speaker power rating. If the amplifier has a substantial power reserve, a minor gain setting error is unlikely to cause immediate speaker damage. However, if the amplifier operates close to the speaker's rated power limit, clipping or overload may occur much more easily.

Before the first full system power-up, it is advisable to set the gain control to the highest numerical value (e.g., 4–8 V). This prevents speaker overload while other settings (filters, levels) are being properly adjusted.

During test listening or measurement, the volume is gradually increased from the head unit while adjusting the gain to monitor for distortion. Clipping symptoms include a sudden loss of detail, characteristic "hissing," or tonal shifts at signal peaks. If any of these artifacts appear, the gain should be slightly reduced.

If the speakers have a lower nominal power rating (RMS) than the amplifier's output capacity, excessive gain must be avoided. Conversely, if the amplifier is weaker than the speakers, it is important to consider the potential for clipping due to insufficient clean power.

After initial gain adjustment, balancing vocal ranges, bass, and high frequencies should be performed comprehensively using the processor's level controls and the equalizer. However, gain establishes the foundation of the balance between frequency bands. For example, if a subwoofer remains too quiet even when set to 0 dB on the processor, increasing the gain on the subwoofer channel may be necessary—but only within the clean signal range.

The signal level delivered to each speaker affects not only volume but also the perceived tonal balance. The midrange may sound "thin" or "harsh" if the midwoofer is underpowered, while an excessively high gain setting on the tweeter can make the highs overly bright. In combination with equalization and crossover settings, precise gain adjustment ensures whether the music sounds "balanced" or "scattered" in tonal coloration.

For this reason, gain adjustment is performed iteratively—from an initial rough setting to final calibration in conjunction with all other tuning tools.

2.5 Fundamental principles of filtering

Filtering is one of the key tools that ensures each speaker operates within its designated frequency range, preventing it from attempting to reproduce frequencies beyond its physical capabilities. The cutoff frequency and filter slope directly affect both speaker protection and the final tonal integration of the system.

2.5.1 Types of filters and their function

- HPF (High-Pass Filter). Blocks low frequencies while allowing higher frequencies to pass through. Used for midwoofers (to prevent sub-bass reproduction), midrange drivers, and tweeters (particularly crucial for tweeters, which have high resonance frequencies).
- LPF (Low-Pass Filter). Attenuates high frequencies, allowing only low frequencies to pass. Primarily used to control the operational range of subwoofers or the lower band of woofers.
- Bandpass Filter. A combination of HPF and LPF that defines a specific frequency band for a driver. Commonly applied to midrange drivers in three-way configurations and some full-range setups.
- Filter Slope (6/12/18/24 dB per octave and higher). A steeper slope results in a sharper attenuation of unwanted frequencies. In practice, low-frequency drivers often require steeper slopes (up to 24–36 dB per octave) to protect them from excessive subsonic movement. Midwoofers and tweeters are sometimes set with gentler slopes (12–18 dB per octave) to provide a smoother transition between frequency bands.

2.5.2 Recommended initial cutoff frequencies and slopes

The table 3 below provides approximate initial filter settings for various speaker types, based on common specifications and resonance frequencies. These values are not final but serve as a safe starting point for system setup, allowing for further fine-tuning.

Table 3. Approximate initial filter settings for different speaker types

Speaker / Frequency Band	HPF (Lower Limit)	LPF (Upper Limit)	Recommended Slope
Subwoofer (ported enclosure)	~5 Hz below port tuning frequency (e.g., 30 Hz for Fb=35 Hz)	80–100 Hz (depending on system configuration)	HPF: 24–36 dB/oct, LPF: 12–24 dB/oct
Subwoofer (sealed enclosure)	20–25 Hz (if sufficient excursion headroom is available)	80–100 Hz	HPF: 12–24 dB/oct, LPF: 12–24 dB/oct
Woofer in a three-way front stage	60–80 Hz	200–300 Hz (depends on midrange selection)	12–18 dB/oct
Midwoofer (two-way front stage)	80–100 Hz	2.5–4 kHz (depending on tweeter selection)	12–18 dB/oct
Midrange driver (rigid cone)	~500 Hz	~4 kHz	12–18 dB/oct
Midrange driver (dome diaphragm)	~1–1.5 kHz	4–5 kHz (sometimes higher)	12–18 dB/oct
Mid-tweeter (rigid cone)	800–1000 Hz	20 kHz (or physical limit)	12–18 dB/oct
Tweeter (three-way front stage)	5–6 kHz	-	12–18 dB/oct
Tweeter (two-way front stage)	3–4 kHz	-	12–18 dB/oct

These cutoff frequencies and slopes ensure that a speaker does not operate significantly below or above its resonance range. Over time, through listening tests and measurements, these values may be refined (e.g., lowering the tweeter cutoff to 2.8 kHz if its design allows).

2.5.3 The role of filtering in speaker protection and frequency integration

Filters not only protect speakers from mechanical overload but also influence how the frequency bands of adjacent drivers overlap. Excessively steep cutoffs can create a gap in the crossover region (or, conversely, a peak if phase alignment is not maintained). Slopes that are too gentle can introduce additional unwanted energy, as drivers that are not optimized for certain frequencies may still reproduce them with distortion.

The optimal balance is usually determined experimentally. If a perceived "hole" in the crossover region is detected during listening tests, slightly reducing the filter slope or adjusting the crossover points to closer values may help achieve a more seamless transition between drivers.

2.6 Time delays: initial calibration

As previously noted, the distance from each speaker to the driver's ear is almost never identical in a vehicle interior, particularly when comparing the left and right channels. Time Alignment compensates for this asymmetry, helping to center the soundstage and expand its boundaries.

2.6.1 Calculation principle for time delays

The method is based on a formula that relates distance to the speed of sound ($v \approx 343$ m/s at 20°C):

$$t = d / v,$$

where t is the required delay time, and d is the difference in distances. For example, if the right midwoofer is 30 cm farther from the listener than the left one, the delay should be approximately 0.88 ms (0.30 m / 343 m/s ≈ 0.000875 s ≡ 0.88 ms).

Many modern processors allow delays to be entered directly in centimeters or milliseconds. The simplest approach is to measure the distance from each speaker to the listening position (typically the driver's head) using a tape measure and input these values. While this method provides only an approximate correction, it already significantly improves localization and tonal balance by compensating for the initial timing discrepancies of sound arrival.

2.6.2 Impact of errors on soundstage and tonal balance

Unlike gain and filters, incorrect time delays do not pose a direct risk to the integrity of the system components. However, improper settings can blur vocals, shift the soundstage to the left or right, or create the perception of a "hollow" midrange. Excessive delay on the left channel, for instance, may give the impression that most of the sound is coming from the right, shifting the center of gravity toward the passenger side.

Some users hesitate to apply time delays, fearing they might "ruin" the sound. However, the key principle here is experimentation—any incorrect setting can be easily reverted, and misconfigured time alignment does not physically damage speakers.

2.6.3 Practical guidelines for initial time alignment setup

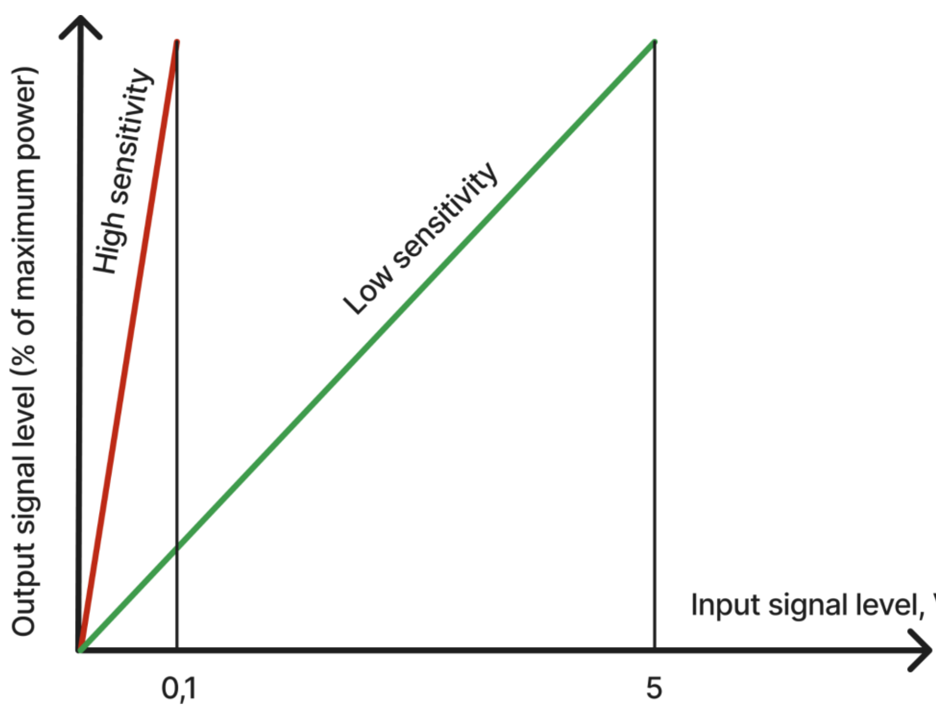
- Measure distances based on the actual position of the driver's ears. The reference point is often placed slightly behind the headrest to account for the body's posture while driving.

- Start by correcting major discrepancies (e.g., the difference between the left and right front channels). Then, fine-tune the alignment between the tweeter and midwoofer to achieve more precise synchronization.
- Use test tracks with a well-defined center (such as monophonic vocals). If the vocals appear to shift left or right, adjust the delay incrementally until the image locks into the center.
- Keep in mind that the subwoofer is usually located in the trunk or the rear section of the cabin, leading to a significant distance difference. To maintain coherent low-frequency reproduction, a delay is often applied to the front stage. If the subwoofer is severely out of phase, the bass may sound disconnected or seem to originate from the back of the vehicle.

CHAPTER 3: GAIN ADJUSTMENT

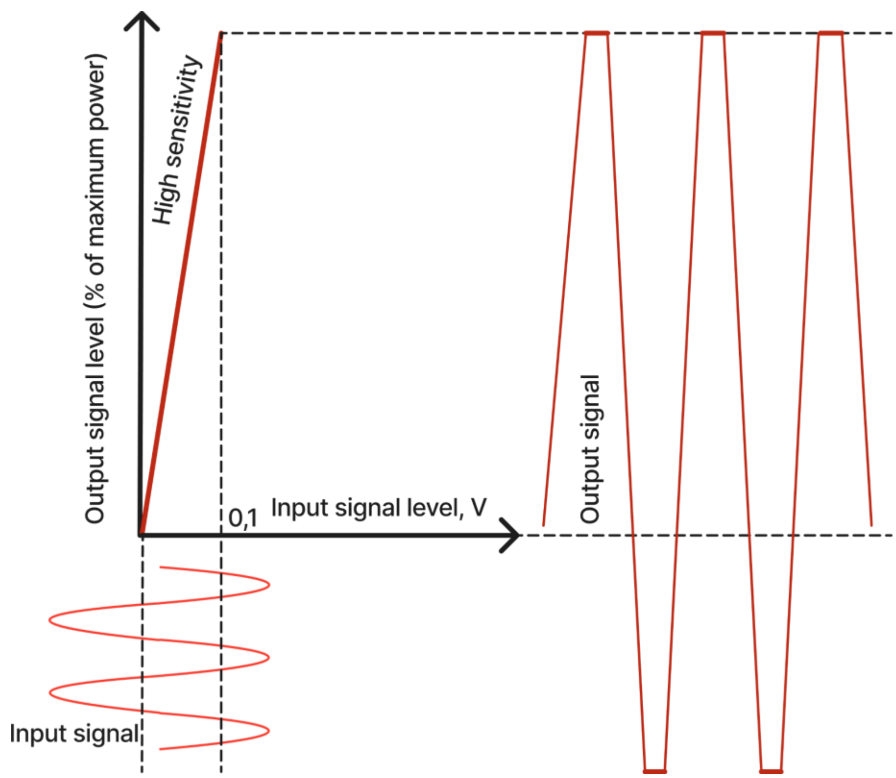
3.1 Amplifier input sensitivity

Input sensitivity (gain) is a parameter that determines the input signal level at which an amplifier reaches its rated output power. Understanding and correctly adjusting this parameter is critically important for high-performance audio systems, as improper sensitivity settings directly affect the signal-to-noise ratio, the risk of clipping, and overall sound quality.



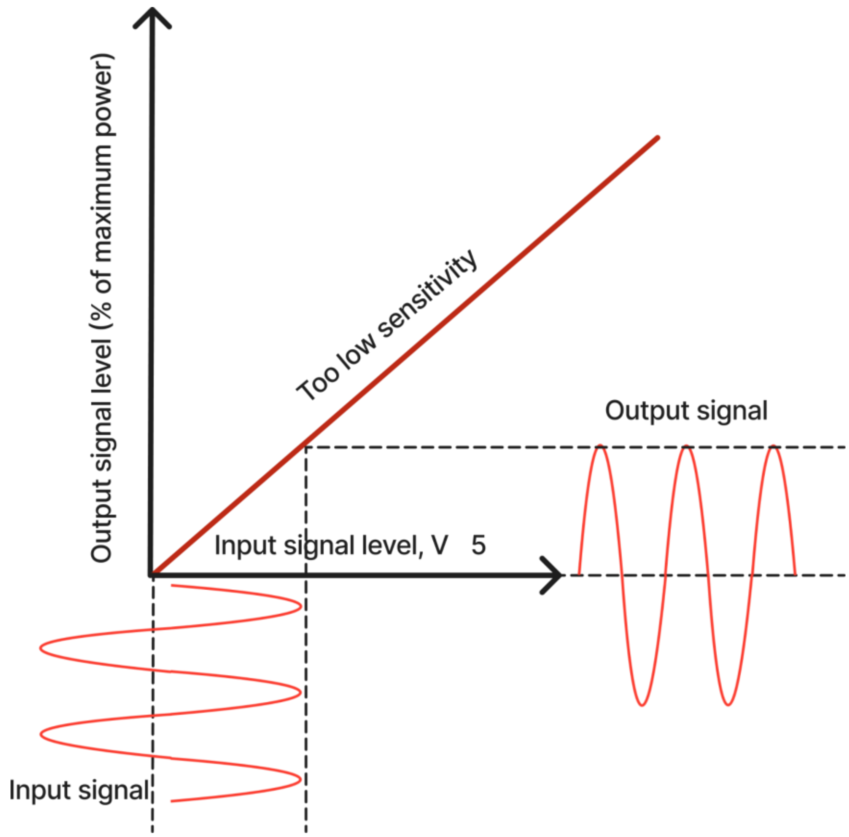
(Figure 7. Input Sensitivity)

A high input sensitivity means that even at relatively low input voltage, the amplifier reaches its maximum output power. From a technical standpoint, this results in a narrow volume adjustment range, as the amplifier requires only a minimal input level to operate at full capacity. Under these conditions, any minor electrical interference, digital-to-analog conversion noise, or analog signal chain distortions (e.g., induced noise in interconnect cables) will be significantly amplified. Moreover, high sensitivity limits the range of volume adjustments from the head unit or processor—a slightly higher input signal level may lead to immediate clipping *(see Figure 8)*.

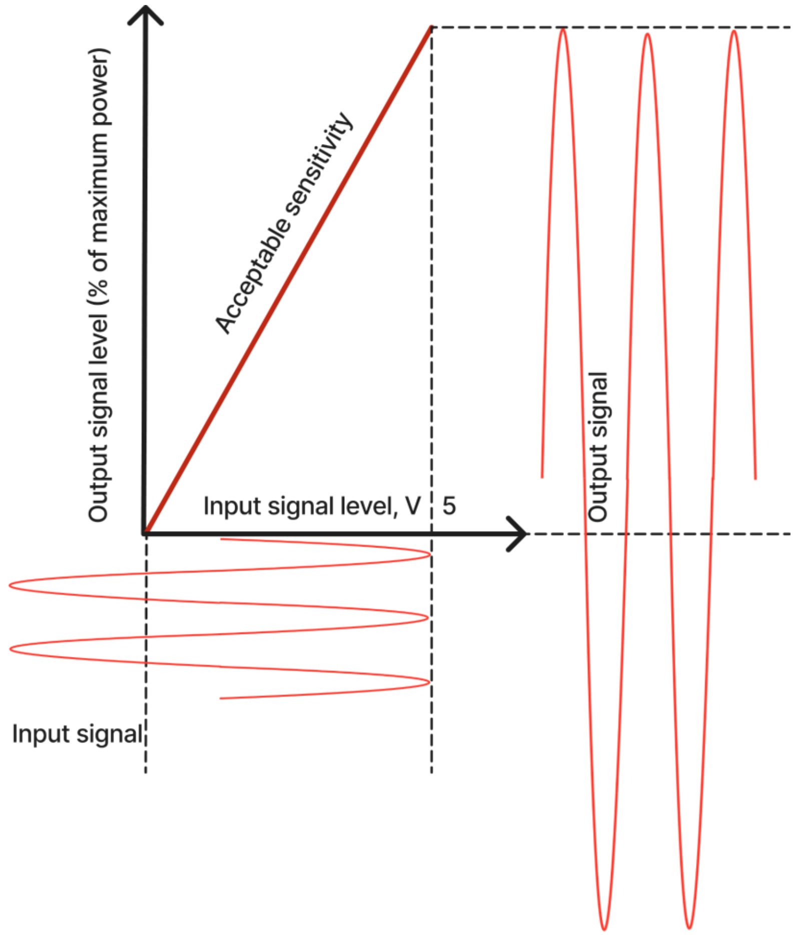


(Figure 8. Clipping at High Sensitivity)

On the other hand, excessively low input sensitivity means the amplifier requires a much higher input signal from the head unit or audio processor to reach its rated power. If the source device cannot provide sufficient voltage without introducing its own distortion, or if the speaker requires substantial power, the system may fail to reach its full potential in terms of maximum sound pressure level (SPL). However, in practical automotive audio setups, the opposite issue is more common—amplifiers often have excess power, and their sensitivity is set too high, causing the system to reach the gain limit with minimal adjustments on the head unit.



(Figure 9. Low Sensitivity)



(Figure 10. Optimal Sensitivity)

Thus, it is advisable to maintain an optimal (moderately low) sensitivity range, where:

- The amplifier delivers linear gain without significant distortion.
- Noise and interference levels remain minimal or insignificant relative to the useful signal.
- The volume control on the head unit has a comfortable adjustment range.
- The risk of clipping when slightly exceeding the input signal threshold is minimized.

Figures 7–10 (illustrative examples) graphically depict the difference between high, optimal, and low gain settings. The vertical axis represents output power (as a percentage of the maximum), while the horizontal axis represents input voltage. A steep curve indicates high sensitivity—even small changes in input voltage quickly reach the upper limit of output power. The optimal setting is represented by a more gradual curve, allowing full use of the dynamic potential of both the head unit and the amplifier.

To account for potential voltage fluctuations in the vehicle's electrical system (such as differences between operation with the engine on and off) and the unique characteristics of specific amplifiers, professionals recommend leaving a margin of safety rather than setting the gain at the absolute clipping threshold under maximum possible input levels. This ensures the system remains stable and predictable across various operating conditions.

3.2 Speaker sensitivity

Speaker sensitivity is defined as the sound pressure level (SPL), typically measured in decibels (dB SPL), produced when 1 watt of power is applied at a distance of 1 meter in a free field. With the development of automotive audio systems, sensitivity ratings are increasingly specified at 2.83V, which corresponds to 1 watt for 8-ohm speakers but results in higher power levels for 4-ohm or 2-ohm drivers. Consequently, specifications may list sensitivity at 1W/1m or sensitivity at 2.83V, which are not always directly comparable.

High sensitivity indicates that a speaker can generate a high sound pressure level with relatively low input power. The most sensitive drivers tend to be lightweight moving assemblies, such as tweeters and midrange speakers. These have a small diaphragm area, short voice coil, and typically high-power magnetic systems (often utilizing neodymium magnets), resulting in a high BL factor (magnetic force factor). As a result, some midrange and high-frequency driversachieve 92–95 dB sensitivity (1W/1m), which is significantly higher than subwoofers or long-excursion midwoofers, which usually fall within the 85–90 dB range.

To understand why subwoofers and midwoofers generally have lower sensitivity, Table 4 presents key parameters influencing speaker efficiency.

Table 4. Parameters affecting speaker efficiency

Parameter	Value / Impact
Moving Mass (MMS)	The heavier the diaphragm, the lower the sensitivity. A greater mass requires more energy for acceleration.
BL Factor (Magnetic Force Factor)	Higher magnetic induction and longer coil windings in the gap increase sensitivity. However, for long-excursion designs, part of the coil moves out of the gap, reducing efficiency.
Resonant Frequency (Fs)	A low resonance frequency is beneficial for subwoofers but often requires a heavier diaphragm, reducing sensitivity.
Magnetic Structure and Gap Design	Narrow gaps improve BL but make long excursion more difficult to achieve.
Use of Neodymium Magnets	Increases BL while reducing weight, but significantly raises costs, making it uncommon in subwoofers.

As shown in the table, engineers must balance low-frequency performance, allowable diaphragm excursion (for deep bass), and sensitivity. As a result, subwoofers and midwoofers typically exhibit lower sensitivity, requiring significantly higher power input compared to high-frequency or midrange speakers.

Psychoacoustic factors also play a role: human hearing is most sensitive to midrange frequencies (approximately 2–5 kHz), which means that tweeters and midrange drivers can sound perceptibly loud even at low power levels.

3.3 Identifying the "weakest link" in the system

In automotive audio systems, there is always a component that primarily limits the system's capabilities: this is where either overload, a sharp increase in distortion, or noticeable mechanical or thermal stress occurs. Identifying and controlling this "weakest link" is an essential part of gain adjustment and frequency band balancing.

1. Amplifier limitation. If the amplifier's power output is close to the nominal rating of the speakers (or even lower), it will be the first component to reach clipping as the volume increases. Clipping is the "hard" distortion of a signal when the amplifier reaches its maximum voltage or current limit. It is heard as crackling or distortion of the peaks of the musical signal. This situation is especially dangerous for high-frequency drivers, as a clipped signal is rich in high-order harmonics. Therefore, when the first signs of clipping appear, the gain responsible for the affected frequency range should be immediately reduced.

2. Speaker limitation. Conversely, if the amplifier has a power reserve (for example, 150–200 W per channel) while the speaker’s nominal power is significantly lower (such as 70–100 W), at a certain point, the speaker itself will become the weakest link. This may manifest as exceeding the linear excursion limit (the diaphragm starts "sloshing," losing control, and the voice coil may strike the backplate) or thermal overheating of the coil. Excessive excursion is audible and sometimes even visible—the diaphragm appears to move uncontrollably, bass loses focus and definition. In extreme cases, a sharp knocking sound from the coil indicates mechanical contact within the suspension.

The thermal limit is generally not visible directly: as the coil overheats, the speaker may degrade over time. In practice, this presents itself as a sudden drop in impedance (eventually leading to failure) or deformation (scraping) of the voice coil windings.

3. Subjective limitation (listening threshold). A "subjective" limit is reached when the sound pressure level is so high (often in the midrange frequencies) that the listener begins to experience physical fatigue, discomfort, or ringing in the ears. In high-quality audio systems with sufficient amplification and well-matched speakers, listener comfort is often the decisive factor: there is no practical benefit in increasing volume further if it causes discomfort or potential hearing damage.

Thus, when adjusting gains in real-world conditions, the weakest link—whether mechanical, electrical, or perceptual—is identified first. If it is the amplifier, the gain is set at the highest possible level before clipping occurs, considering power supply fluctuations, voltage drops, dynamic peaks, etc. If the limitation is the speaker, gain is adjusted based on its maximum allowable power and sound characteristics: subtle nonlinearities, loss of bass texture, or excessive sharpness in the midrange indicate that further gain increase is not advisable. If the limitation is auditory, the attenuation is determined more by psychoacoustic factors: selecting a comfortable volume that does not cause rapid fatigue or distortion.

In automotive audio systems, fluctuations in the vehicle's electrical system should also be taken into account. A voltage drop when the engine is off reduces the amplifier's actual power and can shift the clipping point to an earlier threshold, whereas with the engine running and a stable 13.8–14.4 V supply, the system can operate more freely. Therefore, a reliable gain adjustment process always includes a safety margin, avoiding operation at the absolute limit to prevent unforeseen distortions during real-world use.

3.4 Gain adjustment algorithm

One of the key stages in professional audio system tuning is finding the optimal balance between the input signal level supplied to the amplifier and the capabilities of the speakers. The goal is to utilize a sufficient gain margin without overloading the speakers or pushing the amplifier into clipping. Achieving this requires a step-by-step algorithm that combines theoretical analysis with practical testing in real-world conditions.

Before adjusting levels, it is essential to ensure that all audio system components are in proper working order and correctly installed. This includes verifying interconnect cable connections, checking for breaks or interruptions, ensuring speakers are securely mounted in doors or enclosures, and confirming correct polarity. Installation errors can directly affect the final outcome and may create a misleading perception of a "weak link" even before tuning begins.

Next, the system's filters and crossover points must be checked:

- All channels should have properly set low-pass (LPF), high-pass (HPF), or band-pass (BPF) filters according to their intended function (subwoofer, midwoofers, midrange drivers, tweeters).
- An incorrectly low HPF setting for a midwoofer will inevitably lead to overload, as the diaphragm will be forced to reproduce frequencies beyond its designed range. Similarly, the absence of an HPF for a tweeter can easily result in mechanical or thermal overload when exposed to excessive low frequencies.

For final testing, various musical tracks can be used, but for boundary limit checks, more controlled test signals are preferred:

- Sine wave tones (typically 20–50 Hz for subwoofers, 50–100 Hz for midbass, 500–2000 Hz for midrange, etc.).

- Pink noise for overall level control and channel balance assessment (especially for step 5).
- Musical fragments with well-defined low frequencies (double bass, kick drum, organ) and pronounced dynamic transitions.

In automotive audio systems, the low-frequency drivers (subwoofer and midwoofer) are most prone to mechanical overload. Therefore, gain adjustment for the lower frequency range should take these limitations into account.

For better focus on a specific frequency range and to clearly identify overload artifacts, it is recommended to mute or significantly reduce the levels of other channels. For example, when testing a subwoofer, midwoofers, midrange drivers, and tweeters should be turned off.

The gain of the relevant channel should initially be set to zero or the lowest possible level. Simultaneously, the volume on the head unit (or processor) should be set to a "maximum comfortable" level that is not expected to be exceeded during regular use.

Gradually increasing the amplifier's gain control (or using digital tuning in a processor), the level is adjusted until the first signs of unwanted artifacts become noticeable:

- Sloshing or droning in the low register – an indication that the diaphragm has lost control over its movement. The bass loses clarity and starts to "smear."
- Voice coil knocking (especially in subwoofers) – a critical overload stage where the voice coil physically strikes the bottom limiter (spider or magnetic assembly). This is extremely dangerous for the speaker and requires an immediate reduction in gain.

- Thermal symptoms (not always audible) – prolonged exposure to high power can cause the voice coil to overheat. In practice, this is often detected as a drop in speaker sensitivity or even the smell of burning insulation. This is an unacceptable condition that must be avoided.

When the threshold at which the speaker can still reproduce the assigned frequency range without noticeable distortion is identified, the gain position is recorded or memorized. This is considered the "upper operating limit" for that channel under the current filter settings and power supply conditions.

If the speaker produces excessive resonance due to an overly wide frequency range, it may be necessary to raise the HPF frequency (for a midwoofer) or adjust the LPF (for a subwoofer if it is encroaching into midrange frequencies). Sometimes, modifying the filter slope can help reduce the load at resonant frequencies. After such adjustments, it is advisable to repeat the procedure, as the new filters may allow a slightly different optimal gain setting.

However, even with properly configured filters, there remains a risk of the amplifier entering clipping mode if its nominal power does not have sufficient headroom. Clipping is not always easy to detect in its early stages; the initial signal cutoffs can be mistaken for a sharp transient attack, especially in heavy music genres. As clipping intensifies, it becomes more evident as "crackling," "harshness," and an unnatural "artifact" at peak levels.

Main methods for detecting clipping:

- Auditory monitoring. If a sharp, distorted sound occurs during peak sections (particularly with drum hits, bass guitar, or vocal fortissimo), the gain should be immediately reduced by 1–2 steps, or the system should be checked using measurement tools.

- Oscilloscope analysis. For professional tuning, an oscilloscope or a dedicated clipping monitor on the amplifier with an indicator is used. The oscilloscope displays real-time "clipped peaks" of the signal.
- Multimeters or specialized tools (such as DD-1). These help measure output voltage to determine the onset of clipping.

If the amplifier reaches clipping at the volume level considered necessary for comfortable listening, this indicates either insufficient power headroom in the amplifier or an excessively high gain setting. In most cases, it is preferable to reduce the gain to prevent excessive speaker load from high-order harmonics.

A key principle in high-quality tuning is to avoid pushing the system to its absolute limit. In practice:

1. The weakest link is identified, such as a midwoofer that begins to "resonate excessively" even at moderate volume or an amplifier that clips at high input voltage levels.
2. Its threshold is recorded—the level at which acceptable sound quality is maintained without nonlinearities or mechanical stress.
3. Other channels are adjusted so that they do not exceed the level of this weakest link. In many cases, tweeters and midrange drivers retain a significant volume reserve due to their high sensitivity, and their actual gain settings are often lower than their theoretical maximum. This ensures a clean signal and minimizes noise.

In real-world applications, the optimal gain levels are determined after listening to a variety of musical genres (a mix of electronic music, live instruments, and vocals). In controlled laboratory conditions, the full signal chain—head unit, interconnect cables, amplifier, and speakers—is tested through multiple cycles to eliminate potential distortions that may appear at specific frequencies or under prolonged high-power operation.

3.5 Band alignment and channel balancing

Even if each frequency range (subwoofer, midwoofer, midrange driver, tweeter) is properly filtered and has a "safe" gain level, this does not automatically ensure a balanced and cohesive system sound. It is necessary to align the levels between different frequency bands and compensate for differences between the left and right channels.

The most noticeable transition in the frequency range is between the subwoofer and midbass, as the human ear is particularly sensitive to inconsistencies in the 60–120 Hz range. This is where issues such as gaps or excessive boostsoften arise if levels are incorrectly set.

For this reason, it is recommended to use tracks featuring double bass, deep bass guitar, or drums, where some notes fall into the sub-low range (20–60 Hz) while others overlap with the midwoofer range (80–100 Hz and above).

The tuning process involves sequential listening:

- First, the subwoofer is played alone to evaluate the impact and depth of the low-frequency register.
- Then, the subwoofer is turned off, and only the midwoofers are played to assess where the body of the bass ends.
- Finally, both ranges are played together. If gaps appear (a lack of "body" in certain notes) or, conversely, there is excessive resonance and a boomy effect, the gain levels of the subwoofer and/or midwoofer are adjusted.

Typically, optimal balancing is achieved after several iterations, with reference points based on specific frequency ranges:

- 40–50 Hz (subwoofer zone)

- 80–100 Hz (transition zone)
- 120 Hz (midwoofer range)

Subjectively, the notes should sound approximately equal in volume within the overall musical fragment, without sudden drops or spikes when transitioning between octaves.

While the subwoofer-midwoofer transition directly affects the foundation of the bass, the midwoofer-to-midrange and midrange-to-tweeter transitions define the body of musical instruments and the tonality of vocals.

To check the alignment between midwoofers and midrange drivers, tracks featuring acoustic guitar or piano are particularly useful. These instruments cover a range from approximately 80–100 Hz (bass notes) to 1–2 kHz (upper harmonics). Any unevenness in levels here can create a hollow midrange or, conversely, a muddy and overpoweringeffect.

The transition between mid and high frequencies significantly influences vocal clarity and the brightness of cymbals, brass instruments, and strings. An overly loud tweeter introduces harshness in sibilants and excessive sharpness, while an underpowered tweeter masks details, making the sound muffled and dull.

Again, well-known tracks are used for precise balancing. The listener should focus on the smoothness of the transition—there should be no perception that some notes are missing or, conversely, unnaturally emphasized. By adjusting the gain of the midrange or tweeter amplifier by fractions of a decibel, a seamless tonal balance is achieved.

In an automotive environment, the left speaker (in left-hand drive configurations) is usually positioned closer to the listener than the right speaker. This results in higher sound pressure in the closer channel, particularly in the mid and high-frequency ranges, where wavelengths are shorter than cabin distances. Even if amplifier gain settings are identical, the left channel may subjectively sound louder, requiring additional compensation.

Practical method for leveling:

1. Pink noise (stereo). Serves as a convenient test signal, providing a broadband energy component.

2. Alternating frequency band activation. Used to assess differences between the tweeter, midrange driver, midwoofer, and subwoofer. Ideally, each driver's level is adjusted separately (if the processor allows).

3. Balance adjustment. In simpler systems without individual level controls for each channel, the standard balance/fader adjustment of the head unit is used. While less precise, it provides an approximate alignment.

4. Accounting for maximum deviations. If speakers are properly installed, the level difference between the left and right channels typically does not exceed 2–4 dB. A greater reduction in one channel may indicate installation errors (incorrect angles, geometric or acoustic obstacles).

It is important to note that reducing the level in one channel can slightly alter the overall frequency balance previously established. For example, if the left midwoofer level is reduced by 1 dB, and the left midrange driver by 2 dB, the left side may sound slightly different in tonal balance compared to the right. For this reason, after final channel balancing, it is recommended to re-evaluate the entire system and, if necessary, make final micro-adjustments within minimal limits (±0.5 dB).

After fine-tuning the transitions between frequency bands and compensating for interchannel differences, multiple listening tests with a variety of musical recordings are recommended:

1. Acoustic instruments (guitar, violin, piano): Used to check natural timbre and tonal balance uniformity.
2. Live orchestral recordings: A wide dynamic range helps assess whether any frequency band overwhelms others during peaks.
3. Modern electronic music: Verifies subwoofer integration and consistency with other frequency bands under intense bass pulsations.
4. Vocals across different registers (male, female): Helps refine the balance between the midrange and high frequencies, detecting excessive sibilance or dullness.

At this stage, factors related to real-world vehicle use should also be considered:

- Cabin noise while driving. A slight boost in the midrange may be necessary to counteract road noise.
- Voltage drop when the engine is off, which can affect amplifier peak power output.
- The likelihood of different listeners (friends, passengers) adjusting the volume. The gain margin established in earlier stages prevents clipping or speaker overload if the volume is accidentally increased.

Final tuning typically occurs over multiple days of system use, as the listener identifies nuances in familiar recordings and makes subtle level adjustments in small fractions of a decibel to achieve the subjectively perfect balance. For this reason, professional installers often advise that "instant tuning" is merely a starting point, and further fine calibration requires time and attentive listening across various musical styles.

CHAPTER 4: FILTER TUNING

4.1 Fundamentals and classification of filters

Filtering in audio systems is the process of selectively allowing or attenuating specific frequency components of a signal. Filters make it possible to divide the original broadband musical content into multiple bands (e.g., low, mid, and high frequencies) and direct each band to the appropriate speaker: subwoofer, midwoofer, midrange driver, or tweeter. In automotive audio, precise and flexible filtering is especially important due to spatial limitations and the unique acoustics of the vehicle cabin, requiring fine control over frequency distribution.

Previously, various types of filters and their functions (HPF, LPF, BPF, notch filters) were discussed. In audio systems, these filters can be used individually or in flexible combinations. Their primary role is to adjust the frequency response (FR) so that the load on each speaker matches its design capabilities. Therefore, the next step is to proceed with configuring the filtering parameters.

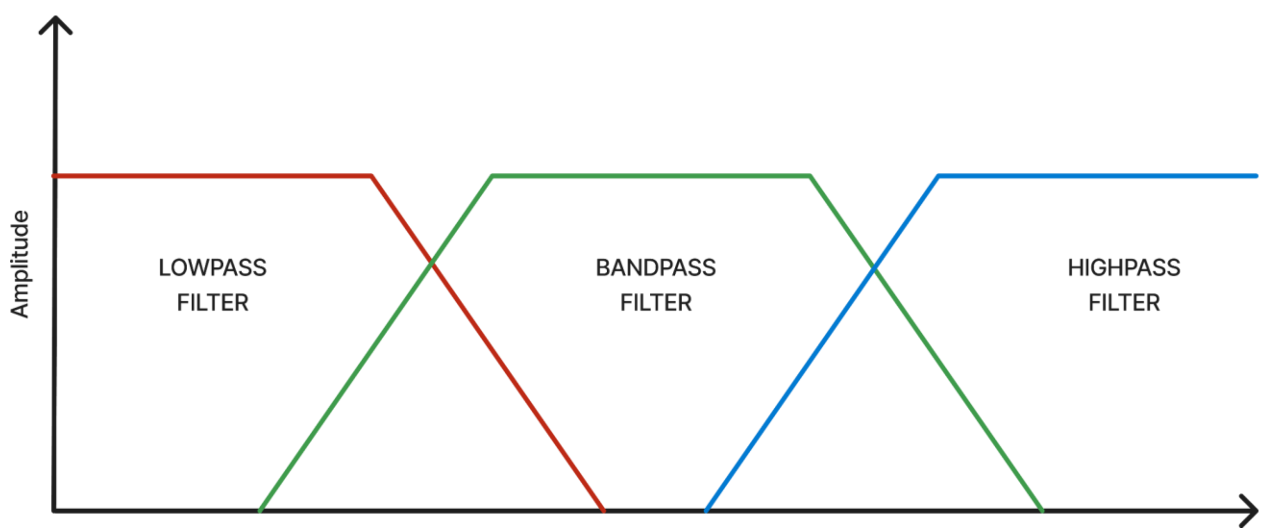


Figure 11. Filters by operating mode

The cutoff frequency is typically defined as the point where the filter's output signal amplitude drops by 3 dB relative to the flat portion of the frequency response curve. For example, in a simple first-order filter (RC circuit), the cutoff frequency can be calculated using the formula:

$$fc = 1 / (2\pi RC),$$

where R is the resistance of the resistor and C is the capacitance of the capacitor. Similarly, for an LC circuit (inductor and capacitor), specific equations apply, but the principle remains the same: by adjusting the reactive components, the filter's cutoff frequency can be shifted either upward or downward along the frequency spectrum.

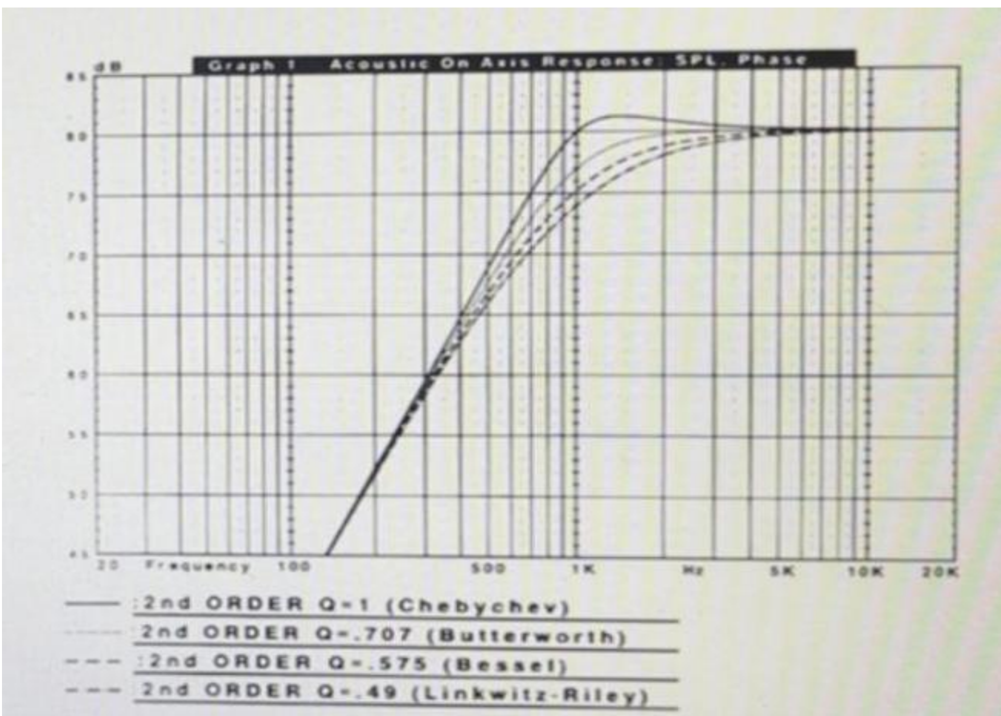


Figure 12. Cutoff frequency and slope

The filter order determines how steeply the frequency response drops beyond the passband. As the order increases, the roll-off slope becomes steeper (6 dB/octave for 1st order, 12 dB/octave for 2nd order, 18 dB/octave for 3rd order, 24 dB/octave for 4th order, etc.). High-end audio systems often employ 2nd- and 4th-order filters, with 3rd-order or higher being less common. However, increasing the order leads to greater phase distortion and increased circuit complexity.

The quality factor (Q) influences the filter's behavior near the cutoff frequency:

- At $Q \approx 0.707$ (Butterworth filter), the most uniform frequency response is achieved without a resonance peak.
- At $Q = 1$ (Chebyshev filter), ripple occurs in the passband, creating an additional boost while ensuring a sharper roll-off outside the passband.
- At lower Q values, such as 0.49 (Linkwitz–Riley filter), phase characteristics and overall crossover behavior are more predictable, but careful tuning is required to avoid dips or phase cancellations.

In *Figure 13*, multiple curves illustrate the response of a 2nd-order filter with the same cutoff frequency but different Q values. The graph shows that as Q increases, the peak near the cutoff frequency becomes more pronounced, resulting in a sharper transition to the attenuation zone.

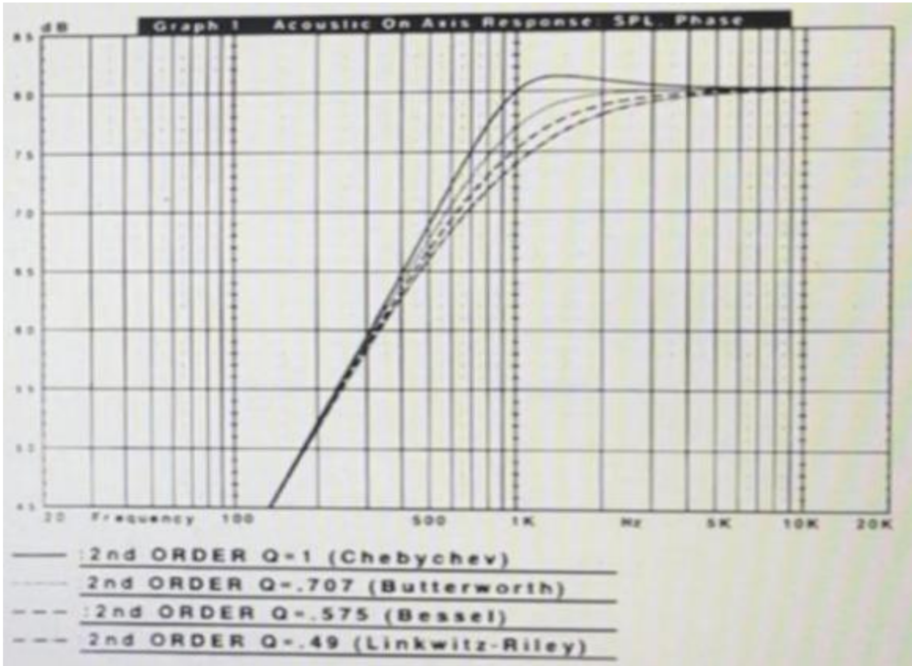


Figure 13. Filter quality factor (Q) characteristics

Filters can be classified into three major categories:

1. Passive filters. These are built using passive components such as resistors, inductors, and capacitors. They do not require external power but inherently cause some signal loss. Additionally, high-order passive filters become bulky and costly. In automotive audio, 1st- and 2nd-order passive crossovers are the most commonly used, often integrated directly into speaker assemblies.

2. Active analog filters. These utilize operational amplifiers (Op-Amps), which compensate for the signal loss inherent in passive components while allowing flexible adjustments of cutoff frequency, Q factor, and filter order. Such filters are frequently found in amplifiers with built-in crossovers.

3. Digital filters (DSP filters). Digital filters fall into two primary types:

○ IIR Filters (Infinite Impulse Response): These replicate traditional analog filter designs (e.g., Butterworth, Chebyshev, Bessel, Linkwitz–Riley) in a digital format. They require minimal computational resources but introduce phase shifts, similar to active analog filters.

○ FIR Filters (Finite Impulse Response): These maintain linear phase characteristics and do not distort the original signal's phase. However, they demand greater computational power and introduce signal latency. In automotive audio, FIR filters are less common and are typically used in high-end DSP processors.

The choice of filter type depends on the balance between sound quality requirements, implementation complexity, available computational resources (in digital systems), and overall project budget. The following sections will examine specific circuit designs and their application in automotive audio systems.

4.2 Passive filtering

Passive filters remain popular due to their simplicity and reliability. They are widely used in stock or manufacturer-supplied crossovers that accompany speaker systems. The absence of active components simplifies the design but also limits flexibility and results in power losses. However, when used with a well-matched set of drivers, a properly designed passive crossover can deliver high-quality sound without requiring an external processor.

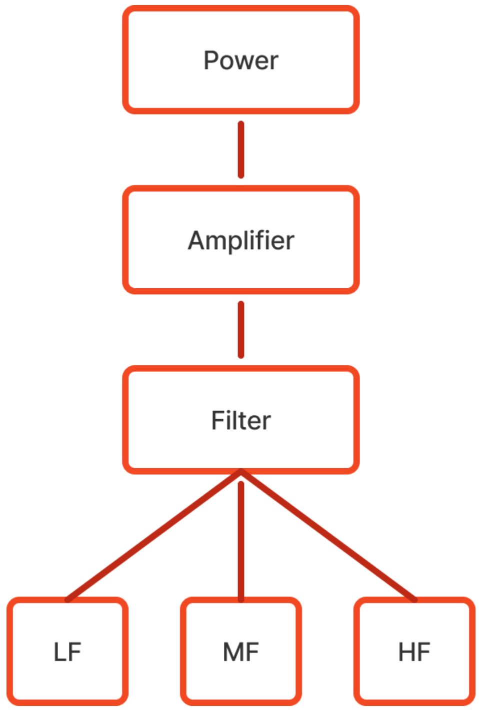


Figure 14. Passive filter circuit

The simplest configuration is a first-order filter (6 dB/octave). For a high-frequency driver, this can be achieved with a capacitor connected in series with the tweeter. The smaller the capacitance, the higher the cutoff frequency. Similarly, for a low-frequency driver (midbass), an inductor connected in series determines the cutoff frequency—the larger the inductance, the lower the cutoff.

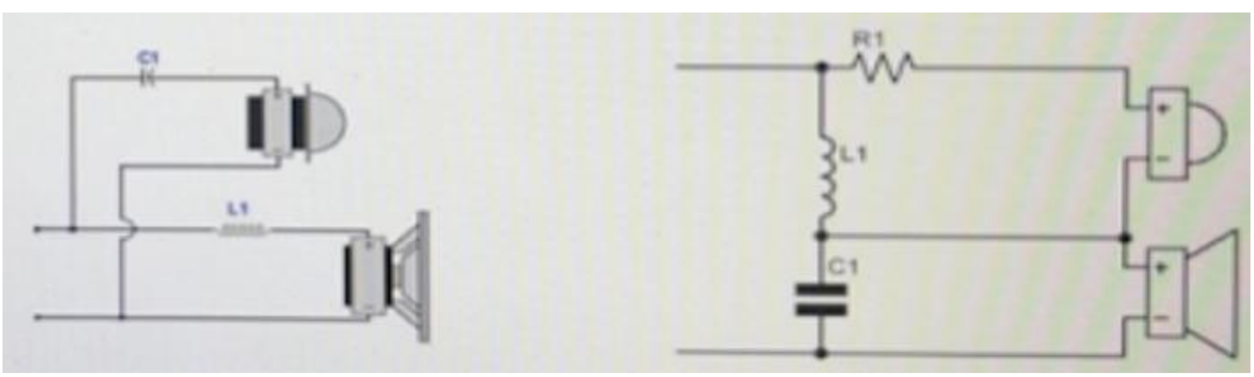


Figure 15. First-order passive filter

Higher-order filters—second-order (12 dB/octave), third-order (18 dB/octave), and fourth-order (24 dB/octave)—are constructed using cascaded RC or LC networks. For example, a second-order filter may include two inductors, two capacitors, and several resistors for level matching. Increasing the number of components complicates the circuit and often increases the physical size, especially when large inductors are used.

Below is a simplified version of Table 5, showing basic formulas for calculating the cutoff frequency of passive filter stages.

Table 5. Basic cutoff frequency formulas for passive filter networks

Filter Type	**Circuit Configuration**	**Cutoff Frequency (fc)**
First-order HPF	Series capacitor (C)	fc = 1 / (2π R_tweeter C)
First-order LPF	Series inductor (L)	fc = R_midbass / (2πL)
Second-order HPF	C + L (series/parallel)	Formula depends on configuration (series/parallel)

Here, R_tweeter and R_midbass represent the equivalent impedance of the respective speakers, which varies with frequency. More precise calculations use impedance graphs of the drivers to account for resonance peaks and actual coil characteristics.

In automotive crossovers, the following elements are often included:

- Additional resistors for tweeter level adjustment (reducing high frequencies by 1–3 dB). This helps equalize the sound pressure of the tweeter with the midbass and prevents excessive brightness or harshness in the high-frequency range.

- Notch filters for attenuating specific frequency peaks. These circuits typically consist of an LC network (sometimes with a resistor) connected in parallel, which creates a dip in the frequency response at a specific frequency, eliminating unwanted resonance in the driver.
- Zobel networks to compensate for the inductance of the speaker coil. These help stabilize impedance in the mid and high-frequency range, simplifying the design of passive filtering.

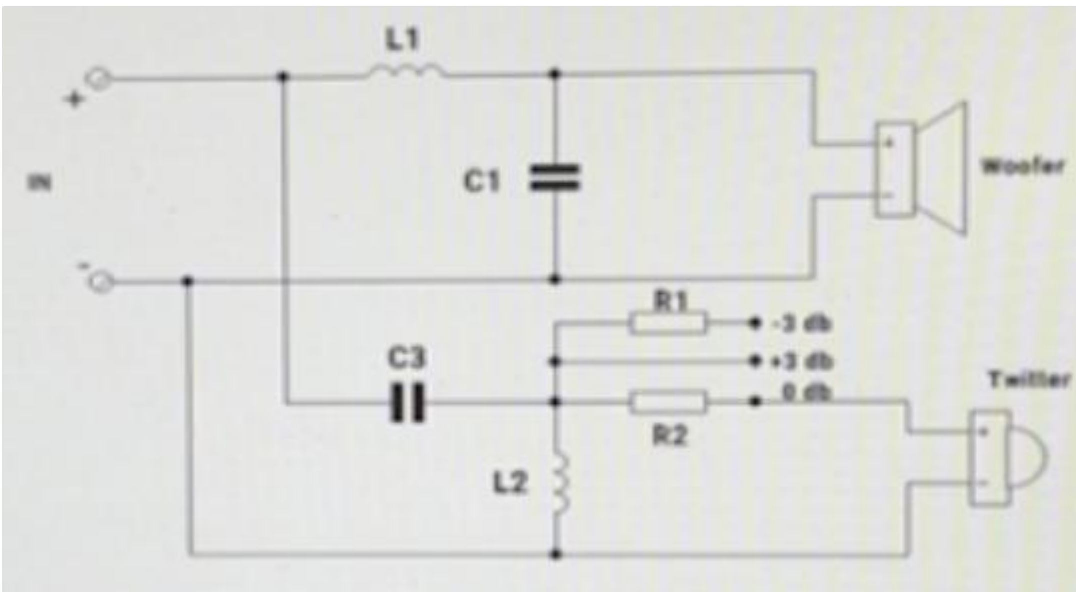


Figure 16. Schematic of a typical automotive crossover

A practical feature in some proprietary crossovers is the presence of a switch (or a set of terminals) to adjust the tweeter level (-3 dB, 0 dB, +3 dB). This allows for fine-tuning of high frequencies based on listener preferences or the acoustic characteristics of the vehicle cabin.

In many cases, a speaker installed in a car may exhibit dips or peaks in specific frequency bands due to resonances from the door panels or dashboard. A band-stop filter can be used for local correction. This filter can be placed between the amplifier and the speaker as part of the passive crossover network.

Another approach is bi-amping. In a standard crossover, the tweeter and midbass are typically connected in parallel. However, it is possible to separate the signal path and route individual amplifier channels to the tweeter and midbass while still using the same passive filters. This offers several advantages:

1. Improved control over the load, as high and mid/low frequencies receive power from separate amplifiers.
2. Easier balancing of volume levels between frequency bands without completely abandoning passive filtering.

However, true flexibility in control is achieved through active filters, which will be examined in the next section.

4.3 Active filters

The transition from passive LC crossovers to active circuits offers significantly greater flexibility in fine-tuning frequency boundaries, slope steepness, resonance characteristics, and band alignment. Additionally, the costs associated with large inductance coils and high-capacity capacitors are minimized—bulky components are replaced with low-power resistors and capacitors paired with operational amplifiers (for analog solutions) or are digitally processed within a DSP.

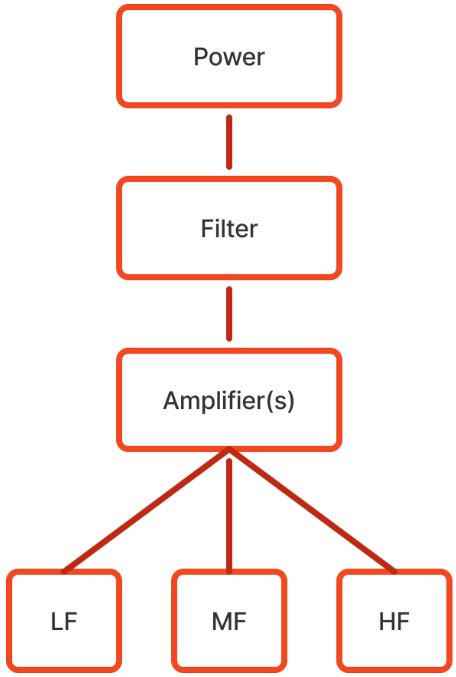


Figure 17. Active filter circuit

In an analog active filter, the signal passes through one or more stages using operational amplifiers. Each stage provides a 6 or 12 dB per octave roll-off, depending on the feedback configuration and RC circuit arrangement. By cascading multiple stages, filters of any required order can be achieved.

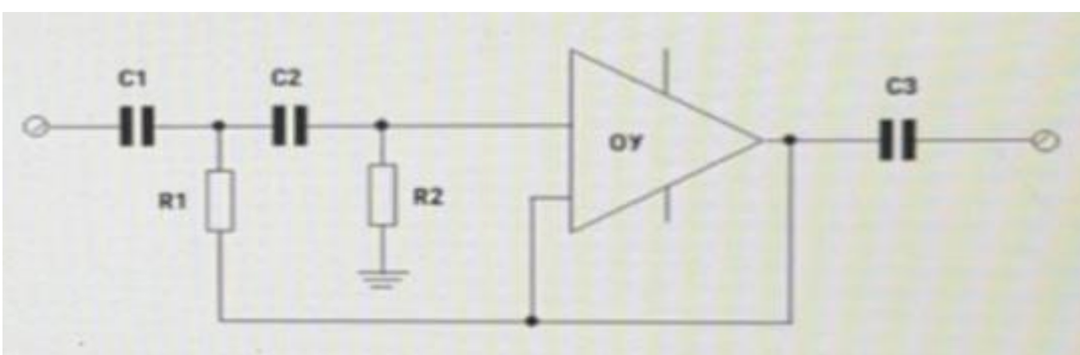


Figure 18. Active filter circuit

Key characteristics:

1. Compensation for signal attenuation: Operational amplifiers can amplify the desired signal, counteracting the losses inherent in passive RC circuits.

2. Flexible tuning: By adjusting resistors or capacitors (or using variable resistors), the cutoff frequency and resonance characteristics can be modified easily.

3. Stability and overload capability: Sound quality is heavily dependent on the operational amplifier itself, requiring low noise levels and minimal nonlinear distortions. High-quality amplifiers with built-in filters often use specialized audio-grade operational amplifiers.

In automotive applications, such filters are integrated into amplifiers with built-in crossovers. Control panels typically feature HPF/LPF adjustments, sometimes a subsonic filter (High-Pass around 20–30 Hz), and selectable slope steepness (12 dB/octave, 24 dB/octave). Some designs also include Q-factor adjustments (Bass Boost) or phase switches.

Digital processing (DSP) involves digitizing the signal before applying mathematical filtering. An IIR filter (Infinite Impulse Response) follows a recursive formula that mimics the behavior of classic analog designs:

$$y[n] = \sum(bk * x[n-k]) - \sum(am * y[n-m])$$

- Advantages of IIR: A low polynomial order provides steep roll-offs with minimal computational load.
- Disadvantages of IIR: Phase distortions, particularly with steep slopes, are similar to those found in analog filters.

FIR filters (Finite Impulse Response) rely solely on weighted sums of past input samples:

$$y[n] = \sum(hk * x[n-k])$$

where hk represents the finite-length impulse response. FIR filters can be designed with linear phase, preventing unwanted phase shifts.

- Advantages of FIR: Reduced phase distortion and fine control over frequency response.
- Disadvantages of FIR: Longer filter lengths (high K values) increase computational demands and signal latency, limiting their use in cost-effective DSPs mainly to high-frequency bands or narrow-band corrections.

Modern automotive head units and external DSP processors used in sound quality competitions often combine IIR and FIR algorithms for different bands. Midbass and subwoofer processing commonly utilize IIR filters (due to power demands), while midrange and high-frequency bands are corrected with FIR filters to maintain phase accuracy.

Ultimately, implementations in amplifiers, processors, and head units fall into three categories:

1. Amplifiers with built-in active filters: The most common type, particularly for monoblocks driving subwoofers, with adjustable LPF (typically 30–250 Hz), subsonic filters (HPF around 15–30 Hz), and optional Bass Boost (Q-factor adjustment around 40–50 Hz). Multi-channel amplifiers may include switchable HPF/LPF per channel (2/4/5-channel configurations, etc.).
2. External active analog crossovers: These standalone units provide multi-band filtering, allowing individual frequency and level adjustments per channel. While less common in the digital era, they remain relevant in mid-range systems.
3. DSP processors or head units with built-in digital crossovers: Offering fine-tuned IIR and/or FIR filtering across all channels, along with level controls, delay adjustments, and equalization. These are the primary tools for achieving precise filtering and tonal balance in high-end competition-level systems.

Active solutions—whether analog or digital—provide a level of precision and flexibility unattainable with traditional passive designs. However, the choice between passive and active filtering ultimately depends on budget constraints, the availability of high-quality amplifiers and processors, and specific engineering goals, whether for a quick-install OEM solution or a fully customized competition-grade system.

4.4 Selection of crossover frequencies and orders

When designing a multi-band audio system, a key step is determining the frequency boundaries for each band (subwoofer, midwoofer, midrange driver, tweeter) and selecting the optimal filter orders. Incorrect evaluation of a speaker's capabilities or an improper slope setting can lead to either overloading (forcing the speaker to reproduce frequencies it is not designed for) or the appearance of "gaps" or "peaks" in the frequency response. The following method outlines a step-by-step approach to achieving correct initial settings.

The process begins with determining the working range boundaries for each component:

1. Subwoofer

- Lower limit (HPF). In a sealed enclosure (SE), a high-pass filter is often unnecessary or set very low (20–25 Hz) to protect the speaker from extreme infrasound. In a ported enclosure (PE), the lower boundary should be slightly below the tuning frequency of the port (usually 20–30 Hz).
- Upper limit (LPF). Selected based on the subwoofer's ability to "cleanly" operate at the crossover point with the midwoofer. The typical range is 60–80 Hz, occasionally up to 100 Hz if the midwoofer cannot play sufficiently low.

2. (Mid)woofer

- Lower limit (HPF). For "lightweight" paper midwoofers in door installations, the HPF is usually set at 80–100 Hz; for more rigid 8-inch drivers, it may be lowered to 60–70 Hz. If the midwoofer is in a small enclosure, higher cutoffs (100–120 Hz) may be necessary.
- Upper limit (LPF). Depends on whether the woofer transitions directly to the midrange or to the tweeter in a two-way setup. In a two-way system, the crossover is often set at 3–4 kHz, which is relatively high. In a three-way system, the transition typically occurs between 400 Hz and 2–3 kHz.

3. Midrange driver

- Lower limit (HPF). The choice depends on the resonance frequency (Fs). A common recommendation is to set the cutoff no lower than twice the Fs. For example, if Fs ~ 500 Hz, the HPF is set at 1–1.2 kHz. In a three-way system, this range can start from 300–500 Hz (rigid diaphragm) or 800–1000 Hz (dome midrange).
- Upper limit (LPF). The midrange driver’s upper boundary is usually between 3–6 kHz, where the tweeter takes over.

4. Tweeter

- Lower limit (HPF). In a standard two-way setup, this is typically set around 3–4 kHz, provided the speaker can safely handle these frequencies at high volume levels. In three-way systems with a dedicated midrange, the tweeter can be introduced at a higher frequency (5–6 kHz).
- Upper limit (LPF). Generally not needed, but in rare cases, a high-frequency cutoff (e.g., 18–20 kHz) is used to eliminate ultrasonic noise.

The primary rule is that the working range should not exceed the capabilities of the specific driver. Otherwise, compromises must be made in either quality or reliability.

Next, practical tuning techniques are considered (accounting for speaker Fs, enclosure type SE/PE, and mid/high limitations).

1. Referencing resonance frequency (Fs). A speaker should not be forced to play significantly below its Fs, especially at high volumes. In a ported subwoofer system, the HPF frequency should be set slightly below the port tuning frequency.

2. Considering enclosure type. A sealed enclosure provides a gentler low-frequency roll-off, so an HPF may not be necessary for a subwoofer in SE (if playing at a reasonable volume). However, for PE, failing to set a subsonic filter (e.g., 20–25 Hz) can cause the speaker to lose control below the resonance point and sustain mechanical damage.

3. Midwoofer/midrange limitations. In a real vehicle interior, doors are often not airtight, so an excessively low HPF may cause the speaker to "bottom out" on bass peaks, losing definition. The midrange (500–1,500 Hz) requires careful handling to avoid "boomy" or "harsh" peaks, making filter settings—especially orders—important for listening comfort.

Once approximate frequency boundaries for each component are determined, the next stage involves checking how smoothly adjacent bands blend.

For the subwoofer/midwoofer crossover, an optimal range of ~60–80 Hz is sought to ensure bass instruments, such as double bass or bass guitar, do not "disappear." For midwoofer/tweeter in a two-way setup, this critical crossover zone is 2–4 kHz, where vocals and most instruments are concentrated.

Phase shifts caused by high-order filters can lead to "gaps" at the crossover point. A common solution is a 180° phase inversion of the subwoofer or midwoofer (using the "Phase Invert" button on the amplifier/processor) to align the summed frequency response.

In practice, effective combinations include:

- Two second-order filters (12 dB/oct) with a full octave difference.
- Two fourth-order filters (24 dB/oct) with a half-octave or even one-third-octave difference.
- A "third-order + third-order" combination requires finer adjustments and sometimes phase inversion on only one component.

In Figure 19 (a conceptual schematic graph), an example of midwoofer and tweeter blending is illustrated: the green curve represents the LPF (midwoofer), the blue represents the HPF (tweeter). A well-matched crossover is achieved when the summed frequency response (yellow line) remains smooth, without sharp peaks or dips.

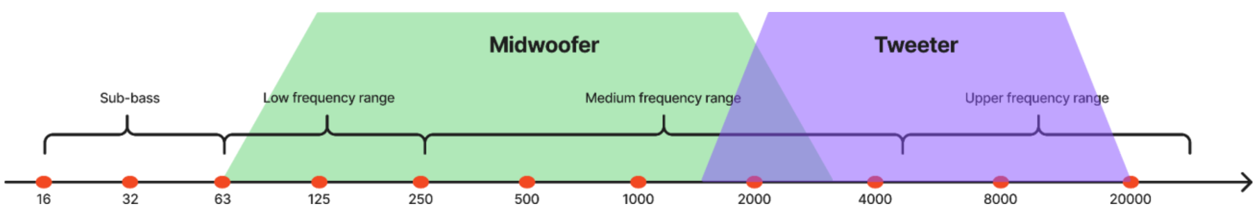


Figure 19. Midwoofer and tweeter mating

4.5 Step-by-step filtering method in a multi-band system

Once the general principles have been established (band boundaries, filter order), it is advisable to proceed from "isolating" each band to their final integration. The following is a basic algorithm applicable to both active digital and passive systems (although the latter has limited adjustment flexibility).

For the subwoofer, an HPF (if needed) should be set at 20–25 Hz (for a ported enclosure) or disabled/set to a minimum (for a sealed enclosure). It is important to check for any "chuffing" at the lowest notes. Then, the LPF (60–80 Hz) should be adjusted gradually, ensuring bass clarity and the absence of excessive "booming." Test tracks with double bass and kick drum should be used to evaluate articulation and tonal "body."

For the (mid)woofer, an HPF should be set at 80–100 Hz, or 60–70 Hz for larger drivers. The LPF should allow the midwoofer to play up to the upper midrange (2–3 kHz) or higher in a two-way system.

The optimal filter order (12, 18, or 24 dB/oct) must be determined: too shallow (6 dB/oct) may cause overloading and distortion, while too steep (24 dB/oct) can introduce phase shifts.

For a midrange driver (in three-way systems), the HPF should be no lower than twice the resonance frequency. It is essential to ensure no overloading occurs at high volume levels. The LPF should be in the range of 3–5 kHz (for a rigid midrange) or 4–6 kHz (for dome midranges). Clarity in vocals, violins, and flutes should be carefully assessed.

For the tweeter, the HPF should be at least twice the tweeter's Fs. In a two-way system, this is usually 3–4 kHz. In a three-way system, it is often 5–6 kHz. The LPF is generally unnecessary or set symbolically (20 kHz). Harshness or excessive brightness at high levels should be checked.

Each driver should be adjusted "in isolation" (muting other bands or reducing their level to a minimum). This ensures that specific artifacts (booming, chuffing, harshness) are clearly identified.

Once individual adjustments are complete, the crossover transitions must be refined.

For the subwoofer + (mid)woofer transition, only these two bands should be active. The sub LPF and woofer/midwoofer HPF should be adjusted for a smooth crossover. The double bass should be clear across both its lowest notes (20–40 Hz) and higher notes (80–100 Hz). If a "gap" or an excessive peak is detected, the frequency or filter order should be corrected.

For the (mid)woofer + midrange transition (in a three-way system), the overlap region of 300–800 Hz (or 400–1000 Hz) should be checked using male vocals, guitar, and cello. Ideally, there should be no perceptible transition between drivers.

For the midrange + tweeter transition, female vocals, violins, and brass instruments should be used. The crossover typically occurs at 3–5 kHz. A transition that is too early may overload the tweeter, while one that is too late may introduce midrange distortion. A phase inversion test on one of the drivers may be helpful.

For the midwoofer + tweeter transition (in a two-way system), the crossover zone is broader (2–4 kHz). Vocals and brass instruments should be used to balance brightness and midrange detail. In some cases, adjusting the filter order (2nd, 3rd, or 4th) helps optimize the summed frequency response.

Once the frequency alignment is complete, the left and right channel levels should be balanced:

- Stereo pink noise should be used to measure (or subjectively compare) loudness.
- Gains or balance adjustments should be made. If a processor is available, each pair of drivers (left/right) should be adjusted separately. In simpler systems, overall balance controls are used.
- Cabin asymmetry (with the left speaker closer to the driver) should be accounted for. The closer channel is usually attenuated by 1–3 dB. However, this may require additional tonal balance corrections to maintain a consistent sound image on both sides.

At this stage, the system should be tested not only with "noise" but also with real music tracks. Sometimes, the left midwoofer may be slightly louder, while the right tweeter is quieter, requiring fine adjustments. The depth of final tuning depends on the listener's preferences and the acoustic characteristics of the vehicle.

4.6 Final adjustments and the influence of tonal balance

Even with meticulous adherence to the previously described algorithm, the final result may require fine-tuning. Each specialist has their own perception of "ideal sound." Additional adjustments help make the system sound more natural and better suited to real musical material.

It is necessary to evaluate the sound using various musical genres:

1. Acoustic music (string quartets, jazz with double bass) helps assess the evenness and naturalness of the midrange and low frequencies.
2. Rock and electronic music (emphasizing kick drums and synthesized bass) reveal whether the subwoofer "locks up" or if the midbass experiences clipping during sharp attacks.
3. Vocals (male and female) expose excesses or deficiencies in the 1–4 kHz range.
4. Orchestral recordings with a wide dynamic range are a good test to determine whether quiet details get lost or if harshness appears in peak moments.

During playback, attention should be given to the subjective depth of bass, the "transparency" of the midrange, and the natural quality of high frequencies. Minor frequency crossover adjustments (±5–10 Hz), filter order changes (switching between 12/18/24 dB per octave), or phase inversion may be necessary.

Even if the amplitude-frequency response appears smooth, phase distortions caused by filters can disrupt the attack and decay of instruments. For instance, a combination of a second-order HPF and a second-order LPF results in a 180° phase shift at the crossover point, sometimes necessitating polarity inversion of one driver. With third- and fourth-order filters, the issue becomes even more complex, as each stage introduces additional phase shifts. This is why specialists often experiment with different "phase invert" options (for the midrange or tweeter) during the crossover tuning process.

Another consideration is FIR filters with linear phase. If the processor supports them and proper calibration experience is available, phase distortions can be minimized. However, this approach requires significant computational resources and precise measurement equipment.

In most cases, a "perfect" tuning is not achieved in a single session. Over time, the user may notice artifacts in familiar tracks or feel that one frequency band is too pronounced or recessed. In such cases, minor adjustments (±1 dB in tweeter level or ±5 Hz in subwoofer crossover frequency) can help balance the sound. Sometimes, it may be necessary to revisit filter order settings—for example, reducing the subwoofer LPF slope from 24 dB per octave to 18 dB per octave to create a smoother transition to the midbass.

Maintaining tonal balance is an ongoing process that accounts for new musical material, shifts in listener preferences, and even seasonal factors (temperature, humidity, suspension condition of the drivers). For this reason, engineers continuously fine-tune the system, striving for optimal sound within a specific vehicle.

Thus, the steps outlined in sections 4.4–4.6 complete the overall filtering process in a multi-band audio system: from defining the initial crossover frequencies and filter orders to the final comprehensive tuning of all frequency bands, including left-right balance and overall tonal balance. The result is a detailed, rich, and well-balanced sound that maximizes the potential of the installed components.

CHAPTER 5. TIME DELAY ADJUSTMENT

5.1 The essence of time correction and its configuration objectives

Inside a vehicle cabin, speakers are typically positioned at varying distances from the listener. These differences can range from a few centimeters (such as the distance between the left and right midbass drivers) to several dozen centimeters (if the subwoofer is installed in the trunk). Since sound propagates through the air at approximately 344 m/s, a difference of 20–30 cm might seem insignificant. However, the human ear possesses an exceptionally high temporal resolution, on the order of tens of microseconds, making even such small discrepancies perceptible.

The purpose of time correction is to virtually equalize the distances from all speakers to the listener by introducing small signal delays to the closer speakers. This effectively shifts their output backward in time, creating the perception that all sound sources are equidistant. The result is a smooth, well-balanced stereo image, where central elements (vocals, lead instruments) are perceived as originating from the center of the dashboard rather than being skewed toward the closest speaker.

The perceived sound source (PSS) is a key concept in this process. The human ear determines the spatial location of each sound primarily through the differences in arrival time of sound waves and their relative levels. When delay settings are precisely configured, each PSS is localized at a specific point and does not shift within the cabin depending on frequency changes or volume adjustments.

The Time Correction function is typically available in digital processors, either built-in or external, as well as in some advanced head units. It is crucial to understand that properly applied delay adjustments transform a set of spatially separated speakers into a cohesive acoustic system with clearly defined imaging and a natural tonal balance.

5.2 Preparatory steps and basic measurements

Before proceeding with the direct adjustment of time delays, several critical steps must be completed.

First, all equalizers and enhancement features should be disabled. Any additional processing (equalization, compression, "vocal enhancement," pseudo-3D effects) can distort the perception of temporal localization. It is recommended to set the equalizer to the FLAT mode and disable all "Bass Boost" or similar functions. This ensures that the effect of time delays can be observed in its pure form without interference from external processing.

Next, the polarity of the speaker connections must be verified. A specialized "phase-inverted" test is used for this purpose (such as a monotone voice with a 180° phase shift). If, when the test is in "phase," there is no clearly focused perceived sound source (PSS) in front of the listener, but it suddenly appears in "out-of-phase" mode, this indicates incorrect wiring of the "+" and "-" terminals. In such cases, one or more speakers are wired in reverse.

Additionally, some amplifiers invert the signal on certain output channels due to their circuit design. In such cases, it may be necessary to manually reverse the wiring at the speaker terminals to maintain correct polarity.

The next step involves measuring the distance to each speaker. The measurement should be taken from the "listening position," typically at the driver's ear level (sometimes from the center of the headrest). If the processor allows for per-ear delay settings, distances should be recorded separately for the left and right ears.

For each speaker, the exact distance (in centimeters or millimeters) to this reference point is noted. Precision is important, as every 10 cm of difference corresponds to approximately 0.29 ms, which significantly affects spatial localization.

Initial delay values are then entered:

- Some processors (Pioneer, Clarion) allow distances to be entered directly, automatically converting centimeters into milliseconds.
- Others (Alpine, Kenwood, and certain external DSP units) require delay values to be input manually in milliseconds. A common recommendation is to initially "raise" all channels by a certain amount (e.g., +5 ms) and then delay the closer channels (reducing milliseconds) to achieve the desired balance.

A simplified formula for conversion (if measuring distances in centimeters) is:

$$t[ms] = d[cm] / (344 \times 10),$$

where 344 m/s is the approximate speed of sound, and the multiplication by 10 converts seconds into milliseconds while accounting for the cm → m conversion.

Table 6. Approximate delay values for typical distance differences

Distance Difference (cm)	**Approximate Delay (ms)**
10 cm	0.29 ms
20 cm	0.58 ms
30 cm	0.87 ms
40 cm	1.16 ms
50 cm	1.45 ms

Variations in temperature and humidity (which affect the speed of sound) generally have an insignificant impact within the scale of a vehicle cabin.

Following the steps above, a "baseline" delay configuration is established, aligning the system to the "farthest" speaker (typically the subwoofer). Further fine-tuning will be the primary focus in the subsequent stages.

5.3 Methods for central image alignment

The primary goal of proper time correction is to achieve a stable central image (often referred to as CI) and a balanced distribution of other sound sources along the horizontal plane (left-center, right-center) and vertical alignment (ensuring that high-frequency and mid-frequency components merge into a single perceived sound source). Several methods can be used to precisely position the vocal in the center and create the widest possible soundstage.

5.3.1 First method (using monophonic recordings and single-channel delay)

First, all frequency bands should be disabled except for the midrange. In a two-way system, the midbass drivers serve as the "midrange," while in a three-way system, the dedicated midrange speakers (500 Hz–3 kHz) are used. The mid-frequency range is the most critical for the localization of vocal elements, making it the easiest reference point for centering the perceived image.

A monophonic track or a human voice recording should be played. Since mono material does not contain stereo differences, any deviation in perceived placement occurs solely due to differences in loudness or arrival time between the left and right channels.

The delay in the left channel should then be gradually increased (for a left-hand drive vehicle):

- Initially, the vocal will "stick" to the left speaker if the distance difference is significant.
- As the delay increases, the vocal image starts to shift from the left side toward the center.
- The adjustment should stop when the voice is perceived precisely in the center of the dashboard or slightly closer to the right speaker.

A final check should be performed to confirm the actual center position. Typically, the ideal acoustic center (CI) is slightly offset from the geometric center of the vehicle—subjective listening from the driver's seat should be the priority. A deviation of 5–10 cm within the cabin width is normal.

If large delay adjustments do not shift the vocal away from the left side, the left speaker's volume should be checked, as it may be playing significantly louder.

Once the central image has been established in the midrange, a stereo recording should be played to confirm that the left and right edges of the soundstage are correctly positioned relative to the center. If properly adjusted, the vocal remains in the center while instruments are evenly distributed to the left and right.

5.3.2 Second method (using a "phase-inverted" test signal)

This method is suitable for more accurate center detection, especially if there are special recordings where the voice or tone generator switches between phase and counter-phase.

1. Leave only the midrange (or midbass in a two-way speaker) as in the previous method.
2. We turn on the phase-counter-phase signal. It can be a voice saying, "Phase... Counter-phase..." with periodic 180° turns.

- With "phase" (summation of signals), the image should be formed in front.
- With "counter-phase" the signal is partially extinguished in the center and may "spread" to the sides if the channels are not evenly adjusted.

3. Introduce a delay in the left channel, observing how the voice moves from the left edge to the center when "phased". At the optimal position, "phase" gives a clear focused image in the center, "counter-phase" gives a diffuse sound without a pronounced KIZ.

4. Check the result: if in “phase” the CI is already in the center, but in “counter-phase” you can still feel that the voice “shifts” more to the right or left, it may be necessary to adjust the volume level of one of the channels.

After completing this procedure for the midrange, the same process should be repeated for the high and low frequencies. The goal is to ensure that, across all frequency bands, the central image remains aligned when a mono component is present in the music. When switching to full-range playback, the "phase-inverted" test should create a sharp image in "phase" mode and a diffuse, centerless soundfield in "inverted phase."

5.3.3 Consideration of volume levels: the relationship between image localization and amplitude

An essential factor in time delay adjustment is the volume level of individual frequency bands. If, for example, the left midrange speaker is 2–3 dB louder than the right one, the image will be pulled toward the left side even if the time alignment is correct. Therefore:

- When adjusting the delay, it should be ensured that a slight reduction (1–2 dB) in the louder speaker helps refine the center alignment without altering the timing parameters.
- If a significant level adjustment (more than 2–3 dB) is required, the delay settings may need to be reviewed, as the auditory system perceives balance differently after such changes.

Once the central mono image is consistently held in place, and further delay adjustments of ±0.1 ms no longer cause noticeable shifts in localization, the optimal setting has been achieved. The next step is to refine vertical alignment across frequency bands and integrate the subwoofer. However, before finalizing adjustments, it is crucial to ensure that the central image remains stable across different music tracks and volume levels, without "drifting" or "jumping." Only after this verification can the time alignment process for the mid and high frequencies be considered complete.

5.4 Adjustment of the vertical phase and band alignment

After performing the basic equalization correction for the central image in the mid-frequency range, a relatively even "horizontal" soundstage (left-center-right) can be achieved. However, in multi-band systems (especially three- or four-way systems), the question arises: where are the other frequency bands (high-frequency, mid-bass) positioned vertically? Ideally, the listener should perceive all frequency bands corresponding to a single instrument as a cohesive sound source, located along the same "horizontal line" in the soundstage. If the tweeter sounds as if it is coming from "above" while the mid-bass is coming from "below," the result is an incoherent soundstage with "double" images.

The process begins with vertical calibration of the mid-bass, midrange, and tweeter. This is typically done starting with the left-side channel (left mid-bass, left midrange, left tweeter) while all other channels are deactivated.

Next, a musical track is played—often a mono recording with a wide frequency range or test tracks featuring female vocals, guitar, and percussion.

The goal is to unify all frequency registers of an instrument into a single sound source. For example, when listening to a female vocal track, part of the spectrum falls into the mid-bass (lower midrange), part into the midrange driver, and the highest overtones into the tweeter. If the "voice" seems to stretch upwards, it indicates that the tweeter's timing is too early or that it is playing louder than necessary.

To correct this, the delays of the tweeter and mid-bass are adjusted so that different parts of the vocal spectrum merge into a single image without upward or downward displacement.

A similar approach is applied when analyzing percussion instruments: the bass drum (which falls within the mid-bass spectrum) and the snap of the snare drum (midrange and high frequencies) should blend into a unified sound.

The same procedure is then carried out for the right-side channels. The principles remain the same, though the exact delay values and levels may vary due to installation differences and the vehicle's cabin geometry. As a result, the left and right channels independently produce vertically aligned sound images.

The final step involves verifying both channels simultaneously. A stereo recording is played to assess how evenly instruments are distributed in terms of height. Often, minor adjustments to the tweeter or mid-bass delays are needed to ensure that the overall soundstage (left-center-right) remains level in height.

In practice, achieving proper vertical alignment requires multiple listening sessions and may involve slight equalization adjustments—especially if, for example, the tweeter sounds too "bright" at the transition to the midrange. The more accurately the vertical phase is adjusted, the more precise the localization of instruments, and the higher (closer to eye level) the perceived soundstage becomes.

When each pair (or triple) of bands within the left and right channels is “internally” matched, the stage of matching all bands simultaneously comes:

1. Turn on the whole front (left + right) and listen to the test music in stereo format, paying attention to whether there are “double” images or phase dips at the junctions between the bands and channels.

Sometimes it may appear that when the left and right tweeters sound simultaneously, the overall HF image “fails” or “blurs”. This may indicate that we forgot a bit about the interchannel phase when converging the vertical phase within each side.

2. Correcting for “fractions of milliseconds.” In contrast to the initial rough settings (0.5-1 ms), now changes of the order of 0.1-0.2 ms can already noticeably affect localization.

At the same time, you should carefully check whether the problems have not “come out” in the midrange band: it is the most critical for the perception of vocals and most instruments.

3. Stability of the scene at different types of music. For this purpose several tracks are used - from live orchestral recordings (where depth and echelonization are important) to “narrow-range” electronic compositions (with accents on different parts of the spectrum).

And the scene should remain whole both when listening quietly and loudly; the “center” should not “jump” even when some frequencies are amplified.

The result of a competent vertical phase is the feeling that all sound comes from an imaginary line at the level of the upper part of the torpedo or a little higher. Midbass and midrange frequencies do not “fall” downwards, and tweeters do not “shout” from above the cabin.

However, even a technically perfect calibration based on measurement instruments may not translate perfectly in a real car interior due to reflections from the windshield, pillars, and door panels. If the system is designed with a strong speaker angle toward the windshield, additional reflections may disrupt the formation of a precise sound source.

Influence of Reflections and Installation Factors

- Windshield Reflections: In some cases, a greater delay is needed for tweeters than their geometric distance suggests, since reflected sound reaches the ear later (or earlier) than the direct signal.
- Speaker Installation: If the mid-bass speakers are mounted in the doors with large gaps, some sound may be lost inside the door, while some is projected into the cabin, leading to unpredictable phase shifts.
- Interior Materials: Cabin materials such as carpets, seats, and headrests can significantly affect high- and mid-frequency propagation. Final delay adjustments should be made while the listener is seated in their usual position, as even clothing can reflect or absorb sound.

In cases where noticeable reflection issues arise, installation modifications are recommended. These may include additional acoustic damping of doors and pillars, more precise tweeter positioning, and the use of absorbing materials in critical areas. With these refinements, time correction settings can fully realize their potential.

5.5 Subwoofer connection and delay adjustment

After completing the front stage tuning (in terms of levels, phase, and vertical alignment), the subwoofer is activated. The goal is to "hide" the subwoofer so that it is not perceived as a separate bass source in the trunk but rather as an integrated extension of the low-frequency register.

To achieve this, methods for determining the optimal phase and time delay of the subwoofer are applied:

1. Case with a full-featured processor and separate subwoofer channels. Most DSP or multimedia head units allow for individual delay adjustments for the subwoofer. In some cases, each subwoofer channel can be adjusted separately (if the sub is in stereo mode).

The process starts with a base delay value that corresponds to the actual difference in distance between the subwoofer and the front speakers. Since the subwoofer is usually positioned farther away, it typically requires a lower or even zero delay.

If the subwoofer "pulls" the sound towards the trunk, its delay is increased (or decreased if the processor is designed such that increasing the distance "removes" the delay). At the same time, polarity switching ("phase inversion") is tested.

The correct result: when playing a double bass or bass guitar, the bass should seem to originate from the same plane as the rest of the music. With eyes closed, it should be impossible to pinpoint the subwoofer's location in the trunk.

2. Case with a mono amplifier (monoblock) featuring a "Phase" control. If the processor does not allow for subwoofer delay adjustment, but the monoblock has a "Phase Control" knob (typically ranging from 0–180° or 0–360°), phase adjustment is used to achieve the best combined frequency response at the subwoofer-midbass crossover point.

Testing method: Bass guitar notes (40–80 Hz) are played while rotating the "Phase" knob until the subwoofer no longer sounds detached from the front stage and instead produces a well-defined, energetic bass response. It is important to avoid overloading the midrange or causing a "dip" at the crossover frequency.

3. Combined approach. Sometimes, a combination of minor delay adjustment in the DSP and manual phase correction on the monoblock is used. This is particularly relevant when the subwoofer is positioned in an unconventional location (e.g., in a side-mounted enclosure in the trunk), causing complex reflections.

If, despite delay adjustments, the bass still appears to come from the rear, the subwoofer's polarity (positive/negative) should be checked:

- Physical wire inversion (switching + and – manually) or software inversion (using the "Invert Phase" button in the DSP).
- In some cases, an inverted phase results in better integration, particularly with certain filter orders (e.g., a 24 dB/octave slope on both the subwoofer and midbass).

A key indicator of successful tuning: the bass seamlessly blends into the overall sound and cannot be localized as a separate source.

An ideally tuned in-car subwoofer does not stand out as an individual sound source:

- The double bass, bass guitar, and kick drum sound naturally positioned "in front" alongside other instruments, while the low-frequency foundation remains rich and deep.
- When the subwoofer is turned off, the absence of bass is immediately noticeable, yet when it is on, its physical location remains undetectable.

If, despite delay adjustments and phase inversion, the subwoofer still "lingers" in the rear, the crossover frequency (LPF for the subwoofer, HPF for the midbass), subwoofer channel levels, and acoustic characteristics of the enclosure and cabin should be reassessed. Adjustments such as repositioning the subwoofer to face the cabin or installing a sealed partition in the trunk may be necessary.

5.6 Evaluation of the final result and final checks

Once all frequency bands (high frequencies, midrange, mid-bass) and the subwoofer are aligned in both phase and timing, a crucial step follows: the final assessment of the entire soundstage under real listening conditions.

The first aspect to evaluate is the soundstage’s distance, width, height, and depth (layering).

1. Distance to the soundstage. In an ideal setup, the vocals (central image) should appear at the level of the windshield or even farther, sometimes giving the impression that the performer is positioned near the hood of the car. If the stage appears too "close," the sound may seem to originate from under the dashboard, whereas if it is too "distant," the sense of immersion may be lost.

2. Soundstage width. Recordings with well-defined panoramic effects are used for evaluation (such as string orchestras or live stereo-mixed recordings). The right edge of the stage often extends beyond the side pillars; the left side is more challenging to extend past the door, but with proper tuning, it can also be expanded. If one side of the stage is noticeably narrower than the other, it is necessary to check the balance of levels (ensuring that no left-side channel is overly dominant) and account for reflection nuances.

3. Soundstage height. The optimal height is approximately at eye level or slightly above, ensuring that vocals sound natural. Some system designers aim for a stage that appears above the dashboard, creating the impression that the performers are "floating" in front of the windshield.

4. Soundstage depth (layering). The listener should be able to mentally position different rows of instruments in depth: percussion is often perceived "behind" the vocalist, while certain reverb effects create an illusion of a vast space. In well-optimized systems, the stage extends far beyond the windshield rather than being confined to a flat "panel."

The next step involves localization and focus of sound images across various test tracks.

Vocals should have a clearly defined focal point, while instruments such as the guitar and piano should occupy their appropriate positions to the left or right. At the same time, percussion and bass should not appear duplicated.

Assessment of depth layers (foreground–midground–background). With proper tuning, different "horizons" of instruments become distinct, maintaining detail and clarity.

Some recordings may create complex panoramic effects. It is essential that stereo effects remain controlled and do not shift uncontrollably to the left or right while bass remains seamlessly integrated with the rest of the frequency spectrum.

A professional indicator of a well-phase-aligned system is that when the driver slightly tilts their head left, right, forward, or backward, the central image does not drastically shift. Minor variations may occur, but the soundstage should not collapse. If, however, a slight movement causes the vocals to relocate to the nearest pillar, it suggests either insufficiently accurate delay settings or an excessive level in one of the channels.

Additional factors to consider:

- A passenger in the adjacent seat affects reflection characteristics (since a person partially absorbs and reflects sound). If significant differences are observed, tuning should be performed without passengers and while wearing clothing that minimally influences the frequency response.
- Seasonal factors (changes in humidity and temperature) may affect sound velocity and the condition of seals, but these variations are generally not significant enough to disrupt the entire tuning.
- On-location measurements. Some specialists use real-time analyzers (RTA) at different head positions to ensure the soundstage remains as stable as possible.

Thus, the full tuning process of time alignment and frequency band integration, including the subwoofer, concludes with a final soundstage assessment. If all parameters (distance, width, height, depth, precise localization, and focus of sound images) meet auditory criteria, Time Correction can be considered successfully executed at the highest level. Naturally, minor adjustments (±0.1–0.2 ms in delay or ±1 dB in level) may be made later if new music reveals shifts in sound images. However, the fundamental structure of the soundstage will already be stable and of high quality, making the system fully prepared for regular use or even participation in sound quality competitions.

CHAPTER 6. TONAL BALANCE

6.1 The importance and assessment of tonal balance

Tonal balance refers to the relative levels of different frequency ranges (low, mid, and high frequencies). A properly adjusted tonal balance ensures that instruments sound natural and recognizable, while any disproportion (such as excessive bass or an overly sharp midrange) significantly reduces the authenticity of the musical image.

The perceived tonal balance is derived from the amplitude-frequency response (AFR) of the entire audio system at the listening position. Subjectively, an ideal tonal balance exhibits a slight decrease in levels as frequency increases. In automotive audio, this often translates to an elevated (but not "boomy") bass and slightly "softened" high frequencies.

The assessment of instrument "recognizability" is conducted as follows:

- A violin should not sound like a squeaky, undefined noise.
- Vocals should not become nasal or muffled.
- A bass guitar or double bass should have a full-bodied foundation without excessive resonance.

Various evaluation methods are applied:

- Auditory assessment: An experienced listener relies on familiar musical passages (vocals, acoustic instruments, recordings of live drum performances).
- AFR analysis: The use of measurement microphones and real-time analyzers (RTA) to identify major peaks or dips and align the response curve to the desired "descending" contour.

The balance between low, mid, and high frequencies is what creates the perception of "proper" music, forming a smooth and pleasant sound palette.

6.2 Correction tools: gain, crossover, and equalizer

Several key tools are used to manage tonal balance in audio systems. Their application allows for a step-by-step optimization of AFR without immediately resorting to complex equalization adjustments.

The simplest way to influence tonal balance is to adjust the gain of the amplifier (or processor) for a specific frequency band:

1. Reducing gain. If high frequencies seem excessively "harsh," lowering the tweeter level by 1–2 dB can help. This not only eliminates excessive brightness but also makes the timbre of vocals and string instruments sound more natural.
2. Increasing gain. Similarly, if mid frequencies seem lacking in presence, the instinctive response might be to "boost" them. However, it is often safer to reduce the levels of the tweeter and mid-bass while keeping the midrange unchanged. This approach minimizes the risk of clipping and unwanted distortion.

A recommended strategy is to reduce the volume of excessively loud bands rather than amplifying quieter ones. This minimizes the likelihood of amplifier overload and preserves the system's overall dynamic range.

The second crucial tool is the crossover. By adjusting the crossover frequency (e.g., shifting the mid-bass from 80 Hz to 100 Hz) or modifying the slope (e.g., changing from a 2nd-order to a 3rd-order filter), the transition between frequency bands is altered.

- Peak: If two bands (e.g., mid-bass and subwoofer, or midrange and high frequencies) overlap excessively at the crossover point, an excessive level in that region may occur. For example, if both drivers reproduce the same frequencies around 2 kHz, a 3 dB peak can form.
- Dip: Conversely, a large gap between crossover points can lead to a lack of energy (a dip) near the crossover frequency.

Correcting these issues can enhance tonal balance without relying on an equalizer by slightly shifting the crossover points or selecting an optimal filter slope (12, 18, or 24 dB/octave).

An equalizer provides a more precise and targeted approach to adjusting specific frequency ranges when other methods fail to yield satisfactory results. However, equalization always introduces potential drawbacks, such as increased signal distortion and frequency-specific clipping. Therefore, it is recommended to exhaust all possible adjustments using gain and crossovers before resorting to equalization.

6.3 Types of equalizers and their features

Equalizers allow for increasing or decreasing the amplitude of a signal in specific frequency bands. They are classified based on the number and characteristics of bands, such as fixed or adjustable frequencies and bandwidth (Q factor).

A tone control is the simplest type (typically a two-band control for bass and treble). It allows for minor adjustments to the bass and high frequencies. While useful in factory-installed head units, it is practically ineffective for precise audio tuning in a modern system.

A graphic equalizer has fixed band frequencies (e.g., a 5-band or 31-band EQ). The greater the number of bands, the narrower each band's range, allowing for more precise adjustments in problematic areas. However, the tuning process can become complex, and an inexperienced user may struggle with managing multiple controls.

For example, in a 5-band equalizer, the "2.5 kHz" slider may affect frequencies from approximately 1 to 4 kHz (due to a wide Q factor). In a 31-band equalizer, each step is significantly narrower (1/3 octave), enabling precise attenuation of peaks without affecting adjacent frequencies.

A parametric and a semi-parametric equalizer should also be considered. A parametric equalizer allows for adjusting not only the gain or cut (in dB) but also the center frequency (in Hz) and the Q factor. If a noticeable peak exists at 120 Hz, the user can set that exact frequency, apply a narrow Q (~1.0), and attenuate the range by 3–4 dB. This provides highly precise adjustments.

A semi-parametric equalizer, on the other hand, allows for adjusting the center frequency while keeping the Q factor fixed. While this offers more control than a graphic equalizer, it does not provide full flexibility.

Regardless of the type of equalizer, it is generally safer to reduce rather than boost frequencies. Increasing gain in a specific band can lead to localized clipping and additional harmonic distortion, particularly in the high-frequency range. Cutting 2–3 dB is a safer approach than boosting by the same amount, especially when the amplifier's overall power is limited.

6.4 The influence of the car interior and resonances

In automotive installations, the acoustic properties of the cabin cannot be ignored. Even if the speaker's frequency response is well-balanced in controlled conditions, additional peaks and dips can occur inside the vehicle due to reflections and standing waves.

Key resonances based on the cabin's dimensions (length, width, height) and their impact on frequency response:

1. Length resonance (typically in the low-frequency range, 40–80 Hz) leads to standing waves between the trunk and the windshield.
2. Width resonance (affecting the mid-frequency range) can cause a peak around 120–200 Hz, depending on the cabin's width.
3. Height resonance sometimes appears in the 200–300 Hz range, influenced by the vertical distance between the floor and the ceiling.

In smaller vehicles such as hatchbacks, some resonances may shift and appear closer to 50–60 Hz. In sedans, a peak is often observed around 60–70 Hz.

Methods for managing these effects include acoustic treatment and optimal speaker placement:

- Soundproofing and vibration damping in doors and the trunk reduce unwanted resonances and minimize bass leakage.
- Installing speakers at the correct angles (especially tweeters and midrange drivers) helps to reduce reflections from the windshield and pillars.
- Using materials with a high absorption coefficient in problematic areas can slightly mitigate reflections.

However, even with proper acoustic treatment, not all resonances can be eliminated, especially in the low-frequency range (20–80 Hz). In such cases, an equalizer or a dedicated DSP module with room correction functions remains the primary tool for fine-tuning tonal balance by attenuating excessive amplitude in specific frequency ranges.

6.5 Practical approach to tonal balance adjustment

Tonal balance tuning is an iterative process: first, gain levels and filters are adjusted, and only then is equalization applied.

In a multi-channel system (two- or three-way front stage + subwoofer), the left and right channels are first adjusted separately to achieve a flat tonal balance. If individual equalizers are available per channel, peaks and dips are corrected independently. The goal is to ensure that the left channel plays evenly across the entire spectrum and that the right channel does not exhibit peaks or drops at the same frequencies.

Table 7. Approach to equalization

Frequency (Hz)	Measured Level (dB)	EQ Correction (dB)	Final Level (dB)
125	+6	−3	+3
250	+3	−3	0
2500	−2	0	−2
4000	+4	−2	+2

Once this step is completed, switching to stereo mode (left + right) ensures that the frequency balance does not shift toward either side.

After each channel is individually balanced:

1. Both channels are activated, and the overall spectrum is analyzed. In some cases, the overlap of the left and right channels may result in unexpected peaks (in the midrange) or dips (in the high frequencies).
2. A slight downward tilt (3–6 dB roll-off towards higher frequencies) is maintained. The sub-bass may be slightly elevated (e.g., around 50–70 Hz), while high frequencies should not exceed the overall level.

Excessive equalization boosts should be avoided. Any increase of +5–6 dB can quickly lead to amplifier clipping. For this reason, professionals tend to attenuate frequencies rather than boost them.

Final tests include: Various music genres (classical, jazz, rock, electronic) to ensure the tonal balance remains realistic; Vocal realism using familiar male or female voices (a cappella recordings). Any excessive distortion immediately reveals unnatural coloration; Indicator monitoring on amplifiers and processors to check for overloading in specific frequency bands during loud passages.

6.6 System-wide control and final listening

After completing all previous steps, a final evaluation of the system's overall sound (front stage + subwoofer, left + right channels) is conducted using full-range musical material.

1. Assessment of coherence and alignment with a descending frequency response. If the bass is overpowering, mid-bass or subwoofer gain may have been overcompensated; if the low end lacks depth, excessive attenuation may have been applied.

High frequencies are evaluated for excessive sharpness. Cymbals and high-frequency noise instruments should be detailed but not overly harsh.

2. Checking stereo image stability. Tonal balance is closely related to spatial localization, as louder frequency bands can appear more forward. It should not happen that, for example, high-frequency overtones shift the image significantly to the left or right.

3. Consideration of real-world cabin conditions and listening preferences:

- Driving conditions: Road noise while driving may mask parts of the midrange or high frequencies, sometimes requiring minor real-time tonal balance adjustments using equalizer settings (Fade/Bass/Treble).
- Personal preferences: Some listeners prefer a "brighter" sound, while others favor a "softer" presentation. The key is to maintain a technically precise baseline balance, allowing for minor subjective adjustments without compromising overall quality.

A properly tuned tonal balance serves as the foundation for an enjoyable in-car listening experience. All tools—gain adjustment, filtering, and equalization—must be used in combination without overemphasizing one parameter while neglecting others. The optimal result is achieved through careful, logical steps, regular auditory checks, and, when necessary, instrumental measurements in various conditions.

CHAPTER 7. EVALUATION OF AUDIO SYSTEM SOUND

7.1 Approaches to developing an "auditory reference"

The final assessment of sound is always based on the listener's experience and "auditory memory." To understand where the "norm" should be in perceiving musical instruments and vocals, it is necessary to develop an internal auditory reference—a personal standard for sound reproduction.

A good starting point is listening to live concerts or visiting a philharmonic hall. The most accurate representation of natural sound comes from acoustic instruments heard in a real venue without electronic amplification.

Observing how string instruments (violin, double bass), a grand piano, or a wind trio sound allows for memorizing the timbral characteristics of each instrument, as well as the spatial scale and volume of the room.

If live concerts are not accessible, visiting owners of high-quality stationary audio systems (Hi-End) in a home environment can be beneficial. Well-configured home speakers in an acoustically treated room allow for hearing a wide range of sonic nuances, which can later be sought in an automotive setup.

Listening to vehicles at sound quality competitions is also valuable. Such events provide an opportunity to compare personal impressions with "professional" or "expert-level" systems.

It is advisable to focus on categories where there are fewer restrictions on acoustic components and where participants aim for the most natural and balanced sound reproduction.

7.2 Differences in frequency ranges and their role

To properly analyze sound, it is essential to clearly understand where the fundamental tone and its harmonics (overtones) of musical instruments are located in the frequency spectrum. This approach helps diagnose problem areas (dips, peaks, or harshness) with greater accuracy.

1. Octaves and frequency subdivisions:

- Sub-bass (16–63 Hz), including the sub-contra octave (16–32 Hz) and contra octave (32–63 Hz). Influences depth and the foundational aspect of sound.
- Low frequencies (63–250 Hz), divided into lower and upper mid-bass, which provide rhythmic foundation and density for bass instruments.
- Mid frequencies (250–4000 Hz), spanning the 1st to 4th octaves, where the primary formants of vocals and most instruments are concentrated.
- High frequencies (4000–20000 Hz), responsible for brightness, clarity, and reverberation (5th octave and above).

2. Fundamental tones and harmonics. Most real musical instruments are characterized by a rich set of overtones that shape their timbral characteristics. If a specific portion of upper harmonics is "cut off" or excessively amplified, the instrument loses its natural tonal quality.

3. Common effects of "deficiency" or "excess":

- A lack of bass in the 63–100 Hz range deprives the music of rhythmic foundation.
- An excess in the 200–400 Hz range results in a boomy, muddled lower midrange.
- A strong dip at 2–4 kHz makes the sound dull, while an excess in the same range makes it overly sharp and harsh.

7.3 Subjective evaluation of frequency response by ear

Even without specialized equipment, it is possible to diagnose frequency range issues. This can be done by sequentially listening to specialized tracks (sine wave signals, pink noise, test samples) and noting where the volume seems excessive or, conversely, too low.

Identifying peaks and dips:

- Below 32 Hz: A deficiency in this range results in a loss of "depth and spaciousness," while an excess introduces a boomy heaviness.
- 63–125 Hz: Resonances in this range often create a "boomy" or "overly thick" bass. A deficiency leads to a weak kick drum and an uninspiring bass response.
- 200–400 Hz: An accumulation of sound energy in this range can cause "muddiness," while a deficiency makes the sound "dry."
- 1–4 kHz: This is the critical range for vocal intelligibility. An excess results in harshness, while a deficiency causes a "dull" and unclear presentation.

A practical approach is to print blank graphs in advance (with "frequency" on the horizontal axis and "level" on the vertical axis). While listening to sine sweeps or short noise bursts, the listener marks where the volume appears higher or lower than adjacent frequencies. The resulting subjective curve visually represents which areas are overemphasized or attenuated.

It is also useful to compare the left and right channels. If, under identical conditions at the same frequency, the left channel sounds noticeably louder than the right (or vice versa), this indicates a frequency response imbalance. Such discrepancies affect the localization of sound images, potentially shifting the perceived position of the audio source.

7.4 Objective frequency response measurements

To obtain a more accurate picture, specialized measurement equipment is used. This allows for assessing the actual frequency response of the system at the listening position.

Measurement microphones and analyzers, such as RTA microphones (SPL LAB, Phonic PAA) and software-based measurement tools (ARTA, REW), help identify narrow peaks or dips caused by cabin resonances, which are difficult to detect by ear alone.

The characteristics of the vehicle interior play a crucial role. The dimensions of the cabin, glass surfaces, and dashboard significantly affect reflections. Even if a speaker's theoretical frequency response is flat, distortions inevitably occur inside the vehicle. Therefore, the microphone is placed at the actual listening position, typically at the driver's head level.

Combining instrumental data with auditory evaluation:

- Measurement tools indicate where and to what extent deviations occur.
- Listening tests confirm whether these deviations negatively impact perception. Sometimes, a slight boost in the 100–125 Hz range subjectively adds "density" to the sound, even if it appears as a 3–4 dB peak on the measurement graph.

7.5 Analysis and verification of sound image stability

Beyond general frequency response levels, it is crucial to evaluate how different frequencies are spatially localized. The perceived sound source (PSI) represents the apparent origin of any instrument or vocal element, and its stability—particularly in the central image region—is often compromised by significant level imbalances or phase misalignment.

Tests for PSI stability across frequencies typically involve sine wave sweeps covering the full range (20 Hz → 20 kHz). The listener notes whether the PSI remains centered or shifts left or right. If at 500 Hz the PSI moves left and at 700 Hz it returns to the center, this may indicate either channel level imbalance or phase anomalies in a specific range.

Interpreting PSI shifts left or right: a leftward shift may suggest a peak in the left channel or a dip in the right. Accurate diagnosis requires cross-referencing with frequency response graphs (either from auditory tests or measurement equipment) for each channel individually.

Correction upon detecting issues: if a significant level mismatch exists between the left and right channels, adjustments are made through gain control or equalization. If levels are balanced but the PSI still shifts, additional phase tuning may be necessary, including verification of time alignment.

7.6 Final criteria for an "ideal" tuning

After all adjustments (levels, filters, equalization, and time delays), the audio system should be close to its optimal state. Several indicators determine whether the tuning process is complete.

1. Harmonized frequency response between the left and right channels. When the left and right channels are tested separately, their frequency responses (both measured and perceived) should not show drastic discrepancies. The overall system response should follow a smooth descending contour without sharp peaks or dips.

2. Natural timbre and accurate instrument localization:

○ Each instrument (guitar, violin, piano) retains its distinctive tonal characteristics and overtones.

○ Vocals remain clear in the mix without being overly recessed or acquiring a nasal or harsh quality.

○ Every sound source maintains a stable position within the stereo image, without "floating" when volume changes or between different tracks.

3. Maximum optimization achieved. The system is considered fully refined if further adjustments (such as additional equalization, crossover

modifications, or subwoofer level changes) result in a deterioration of sound quality rather than an improvement. However, in practical scenarios, refinement is an ongoing process, as the vehicle cabin remains a complex acoustic environment, and listener preferences may evolve.

A combination of auditory training (listening to live concerts and reference systems), precise measurements (using RTA microphones), PSI evaluation, and incremental adjustments to all parameters enables the achievement of a truly high-level audio performance—where each instrument is precisely positioned, and the music attains both naturalness and expressiveness.

CONCLUSION

The study and implementation of the methodologies presented in this monograph demonstrate that the modern automotive audio system is no longer a simple assembly of speakers and amplifiers but rather a complex synthesis of electroacoustic technologies, mathematical models, and psychological observations. A detailed system configuration at the design stage, a well-reasoned selection of digital filtering and delay correction technologies, precise level and equalization adjustments, and a final evaluation of sound quality based on both auditory and instrumental factors make it possible to achieve performance comparable to the most prestigious stationary audio systems.

Despite its inherent limitations, the vehicle cabin is not an obstacle but rather a unique acoustic environment where carefully and methodically adjusted sound reproduction can reveal the full depth and nuance of music. The meticulous application of the techniques outlined in this text ensures that each instrument within the soundstage acquires its own character, is positioned exactly as intended by the system designer, and delivers a presence that closely resembles a live performance.

The combination of applied knowledge in vibration isolation, speaker positioning, cabin acoustics, and digital signal processing elevates the tuning process to a highly precise engineering discipline, where both technical efficiency and the aesthetic perception of the final result are equally important. Overcoming technological barriers and bridging academic research in sound propagation with practical applications in automotive environments mark a new stage in the industry's evolution, where achieving high-fidelity in-car audio is no longer a compromise but a natural outcome of scientific methodology and dedicated effort.

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