Bernadette Schreyer, B.Sc.

End-of-line calibration for multi-channel sound systems in automotive audio

MASTER'S THESIS

to achieve the university degree of

Master of Science

Master's degree program: Electrical Engineering and Audio Engineering

submitted to

Graz University of Technology

Supervisor

Univ.-Doz. Dipl.-Ing. Dr.techn. Daniel Watzenig

Institute of Electrical Measurement and Measurement Signal Processing

Graz, July 2016

STATUTORY DECLARATION

I declare that I have authored this thesis independently, that I have not used other than the declared sources / resources and that I have explicitly marked all material which has been quoted either literally or by content from the used sources.

June 26, 2016

Date Signature

Abstract

With audiophile as the keyword, car companies focus more and more attention on the design, development and improvement of their automotive sound systems. Especially luxury car brands are spending a huge amount of time and resources on the development of highly advanced sound systems and want to ensure the reliable delivery of fully functioning audio systems to every single customer as the latter is willing to spend a decent amount of money for their luxury product.

In practice, the final performance of the sound system is influenced by component tolerances and hardware characteristics which will always result in a spread in performance across vehicles in mass production. The current production procedures require product testing and hardware tolerances on the supplier side but there is no test procedure at the end of the production line for the final product.

This thesis introduces an end-of-line (EOL) test tool which establishes a fast way to evaluate sound system characteristics in the vehicle using a Matlab program with simultaneous audio input and output. With the help of this tool, system properties are assessed. First, only the amplifier is tested. The second test includes the complete signal path from the amplifier to the actual listening area – the car cabin. By quantifying occurring tolerances, a significant variation in performance between a golden reference car and a range of test vehicles can be shown. Whereas differences between amplifiers have been stated to be negligibly small, the spread of results for the complete signal chain, including the electrical, mechanical and acoustical domain, is significantly high. This confirms the necessity of an EOL calibration which should not only contain the measurement on its own, but also a smooth compensation of component tolerances.

For future considerations, a discussion follows about how to realize an EOL calibration in practice. This includes topics such as challenges and requirements in order to establish a reliable test concept in the near future. This is an important step to empower not only car manufacturers but also audio suppliers to deliver sound systems which perfectly match the golden target performance.

Contents

List of figures

List of abbreviations

ANC	Active Noise Cancellation
CCF	Cross-correlation function
DAC	Digital Analog Conversion
DSP	Digital Signal Processing
EOL	End Of Line
EQ	Equalizer
GADK	Graduated Audio Level Adjustment
GALA	Velocity dependent level adjustment (German)
HMI	Human Machine Interface
IR	Impulse Response
MOST	Media Oriented Systems Transport
PHD	Peak Harmonic Distortion
PSD	Power Spectrum Density
RMS	Root Mean Square
SNR	Signal to Noise Ratio
SOP	Start Of Production
SPL	Sound Pressure Level
SUV	Sport Utility Vehicle
THD	Total Harmonic Distortion

Channel names

RLW	Rear Left Woofer
RRW	Rear Right Woofer

FLW	Front Left Woofer
FRW	Front Right Woofer
SUB	Subwoofer
RLM	Rear Left Mid
RRM	Rear Right Mid
FLM	Front Left Mid
FRM	Front Right Mid
RLT	Rear Left Tweeter
RRT	Rear Right Tweeter
FLT	Front Left Tweeter
FRT	Front Right Tweeter
C	Center
RLM SR	Rear Left Mid Surround
RRM SR	Rear Right Mid Surround
FLM SR	Front Left Mid Surround
FRM SR	Front Right Mid Surround

1 Introduction

We are surrounded by sound on a daily basis. And most of us sit in a car every day. The increasing importance of mobility and high-quality audio at the same time results in growth of innovative sound concepts in the automotive sector. Especially car brands of the upper price class have the goal to excel with extraordinary and high-quality sound systems on the market and aim to deliver a fully functional system in every car.

Although high-quality components are used for aforementioned systems, some audio installations show deviations to the target performance of the sign-off vehicle due to component tolerances or other hardware problems and even slight differences can have a significant impact on auditory perception.

It is common practice to test components during the research and development process before the vehicle goes into production, but there is no final check at the end of the line before delivery. This thesis focuses on the possibilities and requirements how to evaluate the functionality of a sound system in the finished car and how to quantify and compensate problems and deficiencies.

After a short overview about automotive sound systems, an end-of-line (EOL) test concept is presented in order to evaluate the characteristics of the sound system in the electrical and acoustical domain and to have a closer look at tolerances. After discussing the test results, requirements for the integration of this tool into the production line will complete the picture, including upcoming challenges and influential factors on the way towards a working solution.

2 Overview of automotive sound systems

At the beginning, a short overview about automotive sound systems is given, including general requirements and involved challenges. Additionally, some configuration concepts are presented and components are discussed in more detail with common measurement procedures in practice for amplifiers and loudspeakers.

2.1 Requirements

2.1.1 Sound quality

Of course it appears obvious to list sound quality as the first main point to focus on. We have to ensure a balanced sound quality in the audible frequency range from 20 Hz to 20 kHz with a flat frequency response for all components.

2.1.2 Speech intelligibility

The other essential part of an automotive audio system is informational speech and applications such as broadcasted news, traffic information, navigation and hands-free devices for telephony. The system characteristics have to establish a sufficient speech intelligibility to ensure a stable information flow.

2.1.3 Connectivity

The automotive sound system or often called infotainment system contains a decent amount of different components and audio sources as you can see in figure 2.1. Infotainment is an artificial word derived from the terms information and entertainment. In clockwise direction, it contains the following connections:

- Navigation
- Radio
  - ➔ Analog frequency modulation (FM) and amplitude modulation (AM)
  - ➔ Digital Audio Broadcast (DAB)
- Phone applications (online apps, communication, maps etc.)
- CD, SD, USB, Aux
- Tools/Settings
- Traffic information
- Bluetooth (headset, telephony)

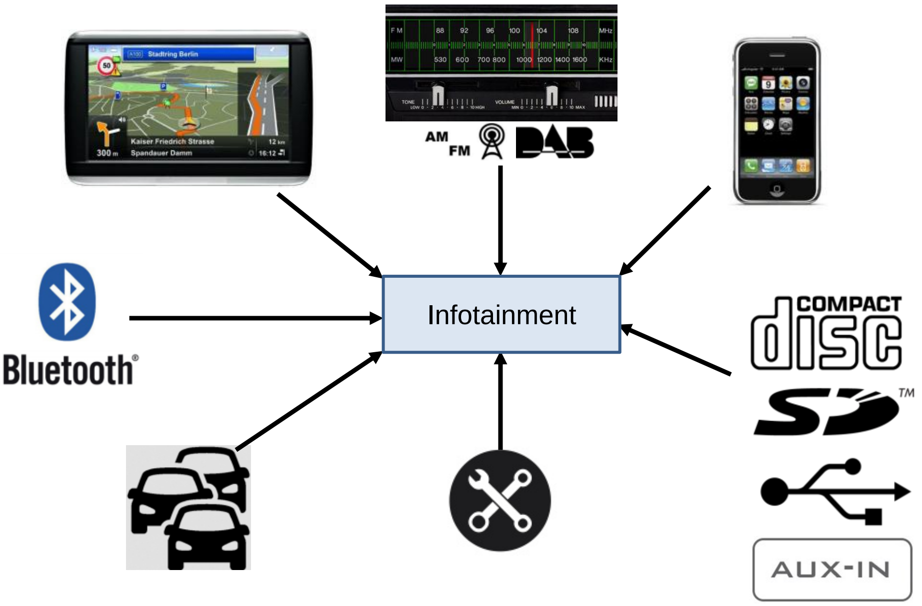


Figure 2.1: Connected components and audio sources [OLI15]

According to the figure, the infotainment head unit is the hub for all incoming sound sources. Handling many different sources in an acoustical environment as we have in the car involves several challenges which are described in the next chapter.

2.2 Challenges

2.2.1 Audio source level alignment

The infotainment system and its user interface combine all audio components. When switching between audio sources, an unpleasant level difference can occur since the different sources do not have the same volume level depending on the audio processing. For example, levels for incoming sources via radio transmission like FM, AM and DAB usually have a different level than the navigation device or the CD input. Thus, the levels for all inputs have to be measured and aligned internally in the amplifier. This is an important step in the development process.

2.2.2 Car interior and reflections

Now we want to focus on room acoustics. While it is relatively easy to design studios or listening rooms, the car interior is a bit more sophisticated due to its small size and irregular shape. Moreover, sound systems are implemented in different types or cars which have also different room impulse responses. As a consequence, a perfectly equalized sound system for one particular car type normally cannot be used for other models. There are even considerable differences between left- and right-hand drive cars of the same model.

Furthermore, when excited by an acoustic source like a loudspeaker, a high degree of short ranged reflections and strong modes can get difficult to handle. In some cases, fast reflections can have more impact on auditory perception than the direct sound according to the law of the first wave front and localization of sound sources can be distorted [SHI13].

In order to counteract dominant modes, resonances have to be determined for each car model. This can be done by a series of measurements of impulse responses (IR in the following) in the car interior. Once detected a strong mode, equalizer presets can dampen the particular frequency with a notch filter implemented in the signal chain (see also chapter 2.3.6 Equalization).

2.2.3 Head position

The IR measurements have to be performed with a microphone on the estimated position of the head of the passengers. Of course, head positions can be different from estimated positions due to different body heights, seating habits and seat positions in the car. Thus, this measurement only provides an average value of the car IR. For a more precise IR identification, the number of measurement points has to be increased. For more exact statements with numerical examples, more investigations have to be conducted to provide a clear relation between measurement points and IR accuracy.

2.2.4 Noise impact

Another significant difference between car interior and listening rooms is the level of present noise, composed of noise components from the engine, the exhaust system, wind and tires. While at low speeds the engine noise and the exhaust system contribute more to the total sound pressure level, the noise from wind and friction between tires and asphalt have a higher impact at higher speeds [ZEL12]. Of course it is obvious that a higher noise level decreases the signal to noise ratio (SNR in the following) which leads to reduced speech intelligibility and influences the perception of sound to a large extent. Therefore, dynamic filtering tools have been introduced such as GALA, GADK and loudness adjustment. These units will be explained in more detail in chapter 2.3 [KAI13].

2.2.5 Component tolerances

This point is one of the key elements in this thesis. Even if all components have to undergo several steps of testing and evaluation during the development process, it cannot be assured that all vehicles leaving the production line have identical sound systems. The internal structure of the amplifiers, the wiring, connections and components of the loudspeakers can modify the signal unintentionally and this results inevitably in perceptible defects. But before going into the matter any further in chapter 3, a few basic concepts of automotive audio systems are explained in the following chapter.

2.3 Configurations

2.3.1 Simple implementation examples

The main parameter for possible configurations is the number of channels and the number of loudspeakers. Figure 2.3 and 2.4 show examples for sound system implementations with two, four and six channels. The number of loudspeakers ranges from four to ten. The installed types of loudspeakers are explained in more detail in figure 2.2.

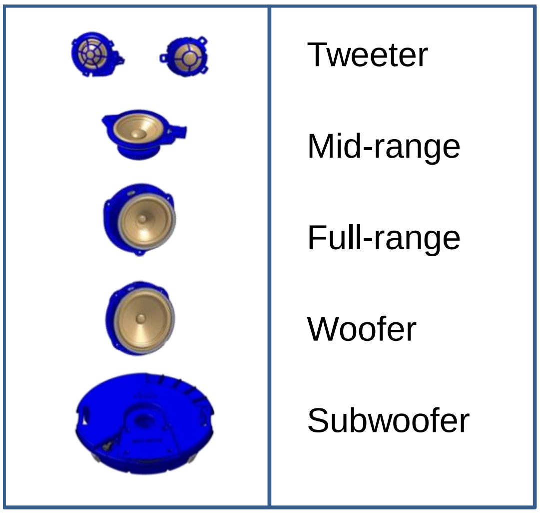


Figure 2.2: Types of loudspeakers (see figure 2.3 and 2.4) [cf. OLI15]

In figure 2.3, two- and four-channel systems are shown. Version A and C have twice the number of speakers as channels. Tweeters and woofers are in parallel and a capacitor functions as frequency crossover.

Figure 2.4 describes six-channel systems with eight or ten speakers. In version E, a mid-range center and a subwoofer are added.

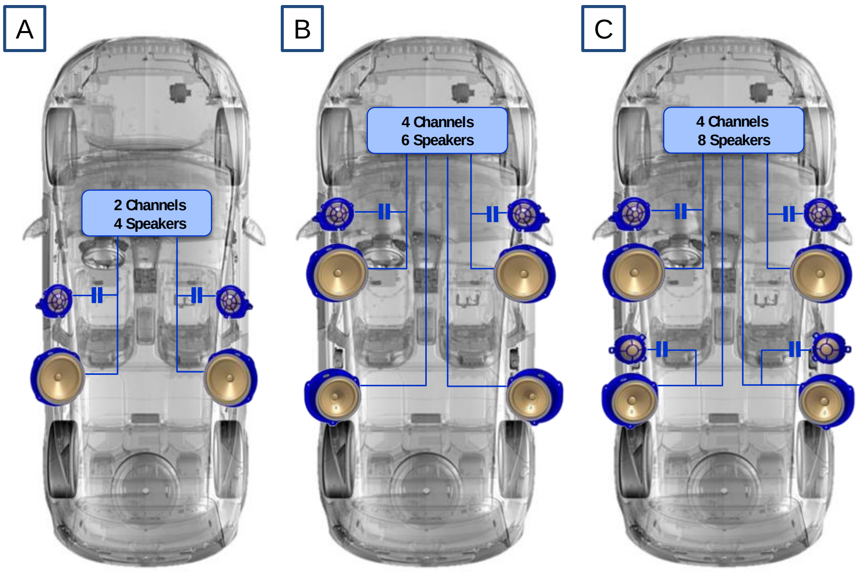


Figure 2.3: Sound systems; A: 2 channels and 4 speakers; B: 4 channels and 4 speakers; C: 4 channels and 8 speakers [cf. OLI15]

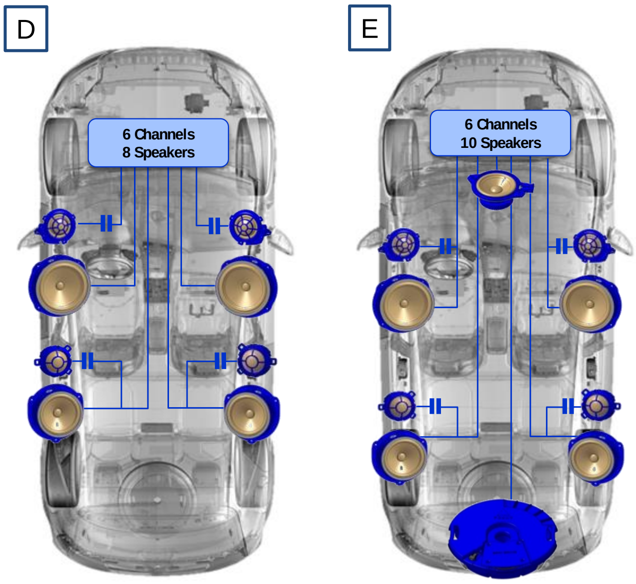


Figure 2.4: Sound systems; D: 6 channels and 8 speakers; E: 6 channels and 10 speakers (additional center and subwoofer) [cf. OLI15]

Woofers are usually installed into the doors, tweeters into the A- or B-pillars as it is shown in figure 2.5.

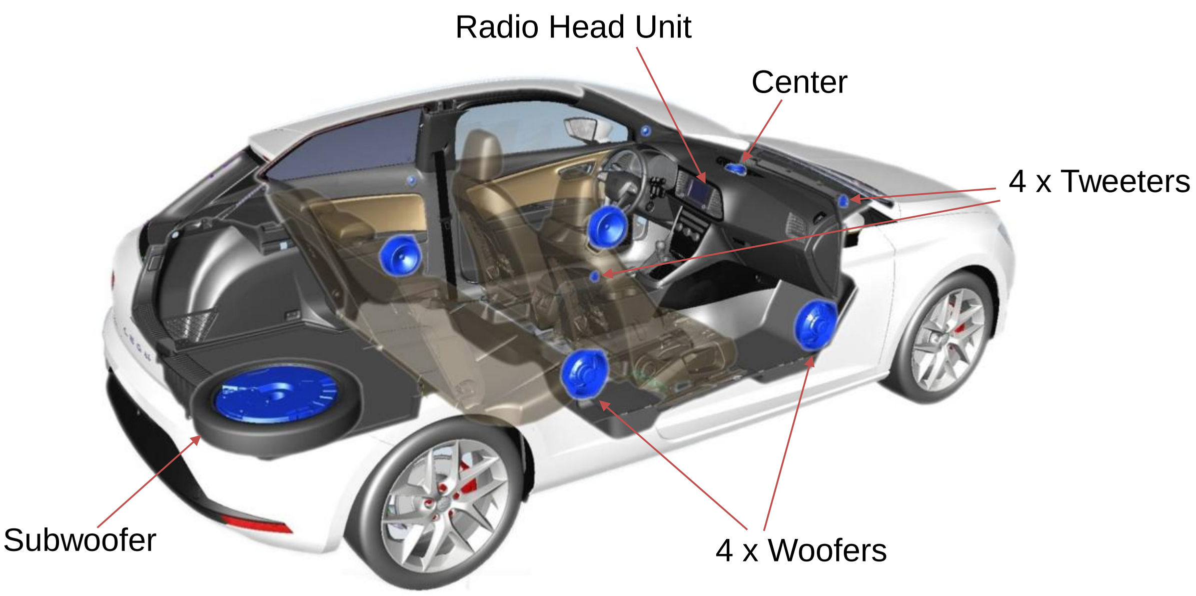


Figure 2.5: Sound system implemented according to version E: 6 channels and 10 speakers (4 tweeters, 4 woofers, center and subwoofer) [cf. OLI15]

2.3.2 Signal chain

The signal chain from sound source to speaker output contains a series of components. Figure 2.6 shows an example with six channels according to version E (see figure 2.4 on the right) with a center and a subwoofer. A short overview of the sequential modules is given in the list below.

- Audio source input (stereo L/R)
- Loudness (see chapter 2.3.4)
- 5-band equalizer (EQ)
- GALA and GADK (see chapter 2.3.5)

Then, the signal is split into six channels and passes through multiple filtering steps.

- Filtering (several filters in parallel)
- Gain
- Delay
- Mute on/off
- Limiter

The channels are labelled as follows:

- RR: Rear Right
- FR: Front Right
- FL. Front Left
- RL: Rear Left
- C: Center
- SUB: Subwoofer

All available channels are then distributed to the number of speakers in the car (in this case a 10-speaker system).

Audio input from telephone and navigation is treated differently. Due to distinct audio content and compression schemes, the filtering methods differ from the entertainment input source.
The user can adjust control parameters such as volume, 5-band-EQ, gains between channels (balance and fader), loudness, GALA/GADK and sound focus on the head unit.

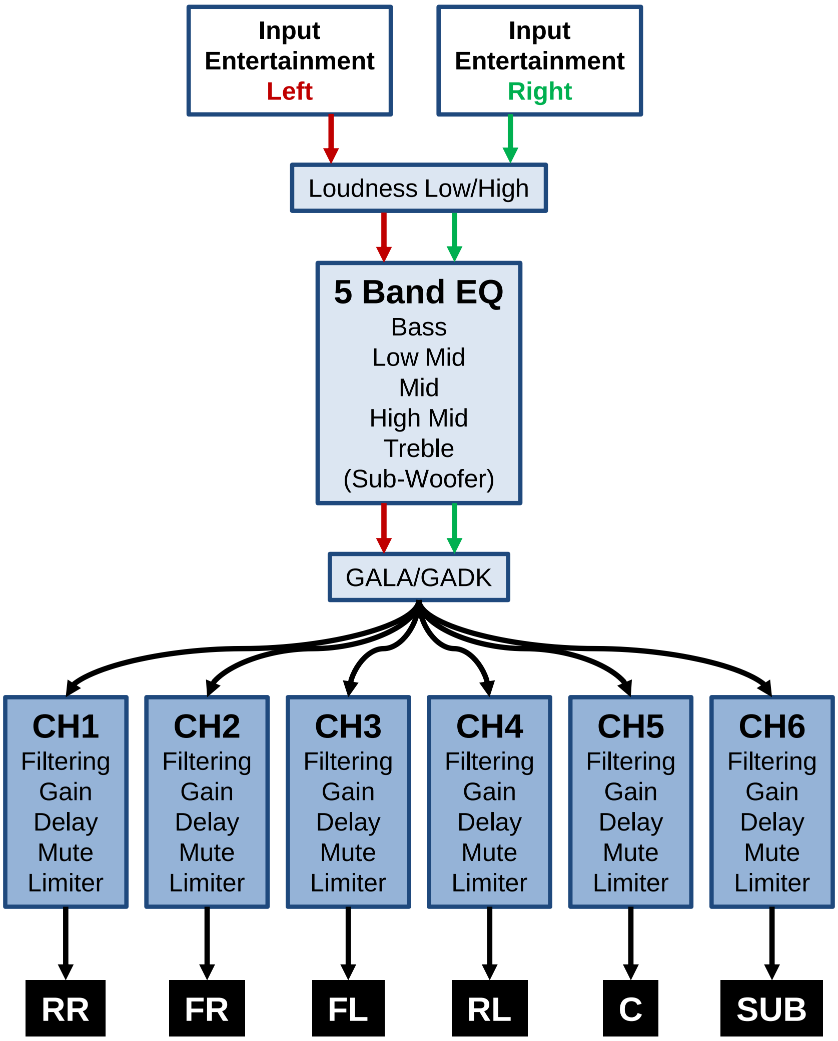


Figure 2.6: Signal chain entertainment input (stereo)

2.3.3 Spatialization

The driver has the possibility to change the sound focus between “All” and “Driver”. The latter concentrates the center of the stereo image in front of the driver´s seat. This functional enhancement is intended for a scenario without any other passengers in the car. In case of more persons than just the driver, the sound focus should be changed back to “All” in order to guarantee a balanced stereo perception for all passengers.

In general, stereo perception can be modified by changing the delays between single channels. Both configurations “All” and “Driver” have presets with adapted delays for each channel.

The subwoofer is a special case. Normally, the subwoofer channel is seen as the reference channel without any delay. Implemented in the trunk, the sound perception can be moved to the front by inverting its phase.

2.3.4 Loudness adjustment

This section describes the loudness behavior of the sound system. This refers to the characteristics of the human auditory perception and the so called phon curves (see figure 2.7). Particularly in low and high frequency ranges, the human ear is less sensitive and needs higher sound pressure levels to perceive the sound signal [TRU99].

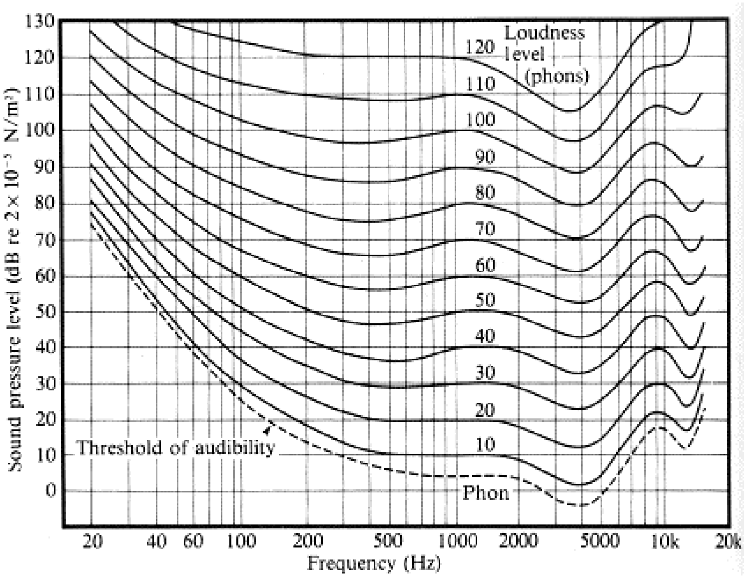


Figure 2.7: Curves of equal loudness (phon);
x-axis: frequency in Hz, y-axis: sound pressure level in dB [TRU99]

Depending on the chosen volume of the sound system, the frequency response can be changed at low and high frequencies in order to compensate the psychoacoustic effects of the frequency-dependent sensitivity of the ear. This means in particular that low and high frequencies are enhanced when the volume falls below a certain threshold to establish a balanced perception of the whole frequency range. The enhancement is realized by peak filters with center frequencies around 100 Hz and 8000 Hz. The frequency responses in terms of volume steps are shown in more detail in chapter 2.4.1 where basic amplifier measurement is discussed.

2.3.5 Velocity dependent parameters

Two modules are introduced which can modify parameters according to the current vehicle velocity: GALA and GADK.

GALA stands for Graduated Audio Level Adjustment. Depending on the current speed, the sound application can increase or decrease the audio output level of the sound system. The gain boost is predefined by the sound engineers during development and sound tuning. The user can set a specific level of GALA directly on the interface. According to the level, a certain gain boost is introduced when the speed exceeds a predefined threshold in order to keep a stable signal to noise ratio when the noise level is increased. Figure 2.8 shows a simple signal flow of the GALA system. Speed values can be handled up to 240 km/h. Depending on the GALA level set by the user, a maximum gain boost up to 14 dB can be applied [KAI13].

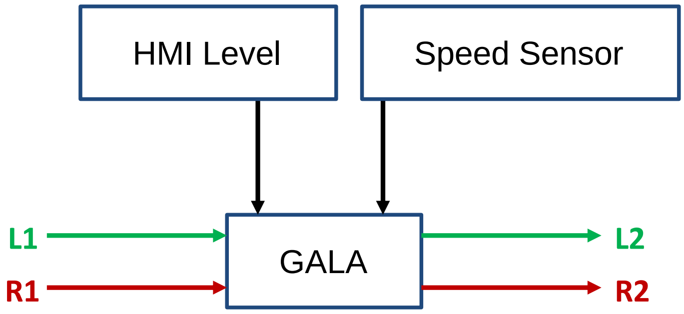


Figure 2.8: GALA system with speed sensor and HMI (human-machine interface); L1/R1: stereo input level, R2/L2: stereo output level [cf. KAI13]

Another way of improving sound quality with increasing speed is introduced as GADK, which is a German abbreviation for velocity dependent dynamic compression. This adjustment tool works in addition to GALA. According to speed, it modifies the dynamic behavior of the output level. Increasing speed leads to a stronger dynamic compression. According to the principle of dynamic compression, the system also needs the current RMS (root mean square) value of the input signal and speed-dependent parameters like thresholds, attack and release time [KAI13].

2.3.6 Equalization and tuning tool

Although a car cabin is a quite challenging listening environment to be designed, it provides the information where the listener is likely to be located in the car. This means, that the sound field can be optimized at those particular spots where the position of the head can be assumed for every passenger.

In order to determine a frequency response for each channel, a set of microphones is placed on the seat at the estimated ear height. Two different positions are determined in order to include both ears. The test signal can be a sweep or random noise which is played by each single channel of the audio system consecutively. The frequency response is measured for both ear positions for each channel with one or more microphones. The number of measurement results for one seat is defined as the product of the number of microphones times the number of channels. The average of the datasets is then compared to the desired frequency response for each channel. This can be, for example, a notch filter with the center frequency at a strong resonance point. The interface to connect to every channel of the sound system and to create filters is a specific tuning tool software. More details about filter design and frequency response adjustment is found in the next chapter about digital signal processing.
One interesting thought is to tune rear speakers to lower frequencies especially in family cars since the passengers on the back seats are likely to be children who have more sensitive hearing in the high frequency range.

During the development process, it is of course not only important to study frequency responses but also to include listening sessions on the respective car seat with reference signals. These can be, for example, well-known recordings of music containing a broad range of different genres with various instruments and vocals. All channels with the set of created filters build the acoustic sum. Additional delays between channels can be used to modify the overall frequency response on the listening position, also referred to as reference point.

For listening purposes, real-time applications in the tuning tool convolve the music signal with the designed filters and delays. The following section provides more information about the possibilities with digital signal processing in the field of automotive audio.

2.3.7 The role of digital signal processing

An advanced way to adapt frequency responses of channels is to include digital signal processing (DSP) tools. In general, loudspeaker characteristics and the listening room itself introduce modifications and colorations on the reproduced sound. These inevitable colorations can be very difficult to handle with conventional hardware design or modification of the room by installing absorbing or diffusing materials. DSP software analyzes the performance of speakers and the characteristics of the room and corrects the colorations in order to improve the acoustic experience or to remove resonances. This is commonly called room correction. With appropriate test signals and one or multiple microphones, impulse and frequency responses are detected and the tool detects deficiencies to be compensated. This allows to calibrate rooms in a very cost efficient way based on detailed acoustic measurements in the listening area.

Sound optimization technology enjoys continuously growing popularity in the sound labs of car companies in order to improve the performance of their high-quality audio systems. Each channel is digitally tuned to reach not only the highest level of sound quality but also a clear, detailed stereo image [DIRa, DIRb].

Impulse response correction

Localization cues are critically dependent on time-domain properties or more precisely on differences and similarities between the incident sound on the left and the right ear. Thus, the impulse response is the critical element for stereo imaging, positioning and clarity. Important parameters to be modified in this field are characteristics of the direct wave, early reflections and decay time.

Frequency response correction

The characteristics of sound are generally improved by modifying the frequency response. Since temporal aspects also affect the sound to a large extent, frequency correction is usually done after time-domain modifications. Figure 2.9 shows a general example for frequency response correction. The yellow line represents the target curve, the blue line demonstrates the average spectrum before filtering and the green line is the result after applying the digital filter to the original spectrum in order to adjust it to the target curve.

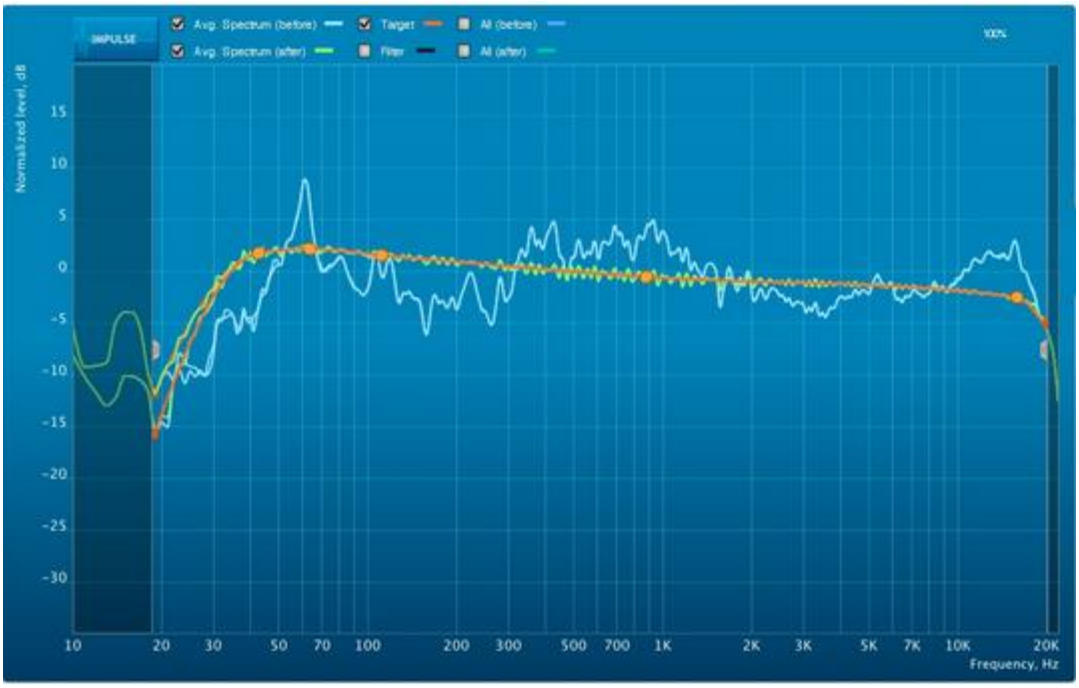


Figure 2.9: Software-based frequency response correction [DIRb]

By including sound field technologies, room correction achieves the next level of possibilities, for example 3D technology, placement of virtual sound sources and reproduction of predefined sound fields. In the first steps, the principle is similar to the procedure before. A microphone array gathers information of the already existing sound field. The sound field synthesis system uses multiple loudspeakers to generate a 3D sound field whose impulse responses have been measured or simulated before as a reference. In luxury car audio design, it provides the possibility to give all passengers an impression to listen to sound sources outside the car, including automotive surround sound and much more. By expanding the acoustic space, variations over different seat positions can be reduced.

Even more, also active acoustic treatment gains more and more popularity. One important keyword in this matter is active noise control as it is known from active noise cancellation headphones and active noise reduction in machines, for example air conditioning ducts. To reduce the primary unwanted noise, small microphones in the car pick up the surrounding noise and a secondary sound source adds a specifically designed “anti-noise” signal which cancels out the primary noise source. In practice, it leads to a reduction of the noise source and achieves quite good results, especially in case of stationary characteristics of the treated noise [HAG15].

In conclusion, DSP software provides many possibilities for carmakers to deliver a highly advanced listening experience to their customers and engineers are putting a high amount of effort in the development of such software and systems.

However, even the best software is not able to compensate insufficiencies of the hardware parts. In order to ensure a high quality for audio system components, a series of measurements and evaluations is required during development processes. In the next chapter, methods for testing amplifiers and loudspeakers are shown, including measurement examples in practice.

2.4 Basic evaluation of sound system components during the research and development phase

This chapter addresses the measurements of components for sound reproduction systems. Before the system goes into production, all components provided by suppliers have to be checked for functionality in order to be approved for the production line. First, a short overview is given about the complete signal chain in the electrical, mechanical and acoustical domain. Then, we will focus on two main components – amplifiers and loudspeakers and how to evaluate their characteristics. Finally, the link is built to the main part of this thesis: the assessment of the approved sign-off sound system at the end of the production line.

Figure 2.10 shows a detailed overview about the signal chain with the audio signal passing through the electrical, mechanical and acoustical domain. Undesired linear and nonlinear distortion modifies the signal throughout the way. As a digital sequence of ones and zeros, the signal is converted into an analog voltage and is fed to an amplifier. In case of two or more ways, a frequency crossover separates the broadband signal into bandpass signals according to the particular frequency range of woofers, mid-range speakers or tweeters. The moving coil of the speaker system acts as an electro-mechanical transducer converting voltage values into movements of the coil due to the principle of electromagnetic induction. In the next step, the signal travels from the mechanical into the acoustical domain. The displacement of the cone and the vibration of the diaphragm are translated into sound pressure in air. This mechanical wave resulting from the back and forth vibration of particles propagates through the medium and is modified according to the laws of wave propagation, diffraction and reflection, depending on the surroundings such as free field conditions or a quiet room. The end of the signal chain is represented by psychoacoustics – the human auditory system and its characteristics of perception and evaluation [KLI14].

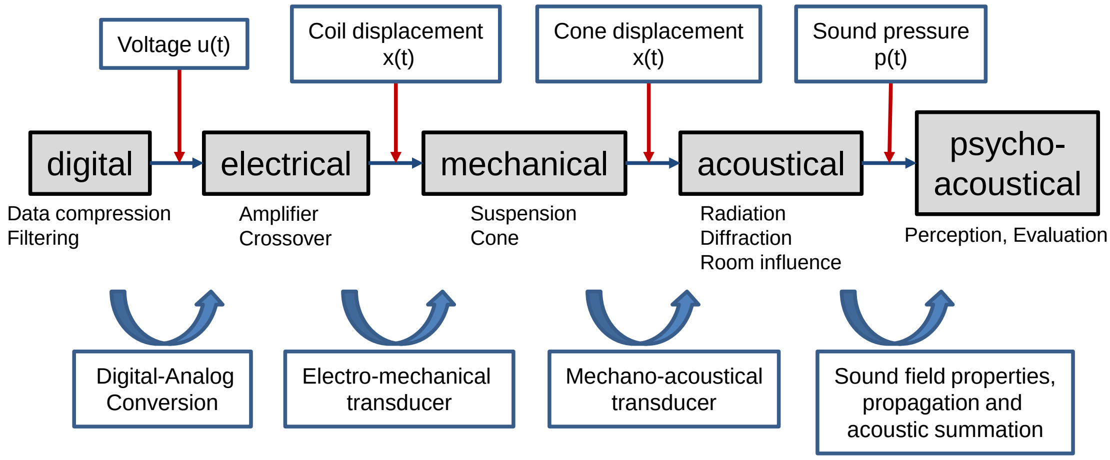


Figure 2.10: Signal domains overview [cf. KLI14, p.10]

In the following, two main components of audio systems are about to be explained in more detail: loudspeakers and amplifiers.

2.4.1 Amplifier testing

2.4.1.1 Test setup with Audio Precision

When testing audio equipment, Audio Precision is the recognized standard in audio electronics, providing a wide range of high-quality tests for multiple kinds of devices. The Audio Precision multi-channel analyzer comes with an internal signal generator but also allows external input sources. Common test signals are slow and fast sweeps from 20 Hz to 20 kHz and 1 kHz sinusoidal waves from -60 to 0 dBFS in steps of 10 dB.

The audio amplifier is connected to the Audio Precision analyzer in a setup which simulates a car-like situation, the ignition system, power supply and the resistors of the loudspeakers. All amplifier settings can be controlled via the infotainment system screen, the human-machine interface (HMI). The Audio Precision analyzer is accessible via software. Figure 2.11 demonstrates the complete signal chain.

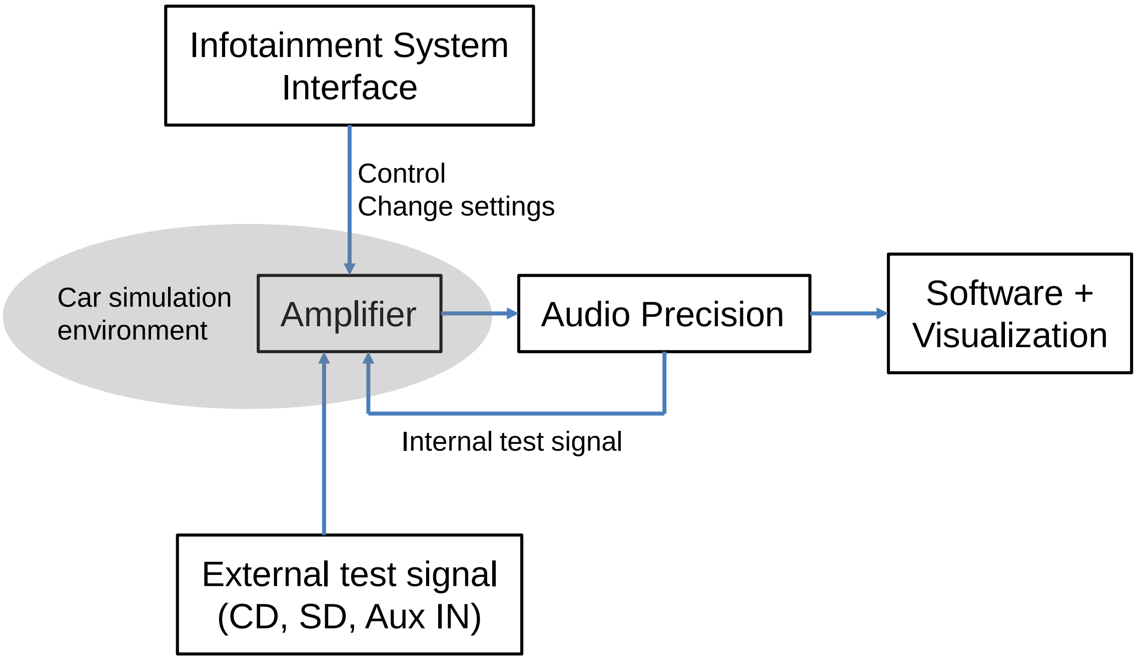


Figure 2.11: Signal chain of the measurement setup for internal amplifier testing

The tool offers measurements of a variety of device properties such as system gain, total harmonic distortion, frequency response and the delay between channels [AP].

2.4.1.2 Measurement example

Figure 2.12 provides insight into one of the measurements on a 6-channel amplifier implemented in the sound system which is shown in chapter 2.3.1, figure 2.4, version E. It shows the loudness adjustment as it was mentioned in chapter 2.3.4. For twelve volume steps between maximum and minimum, the frequency response is measured. As it can be seen in the figure, the adjustment filter is only implemented for the low frequency range. According to the specifications in [KAI13], a peak filter with a center frequency of 65 Hz increases the amplitude of low frequencies in order to establish a more balanced sound impression over the complete audible frequency range for low volumes. The frequency responses are only shown for the rear right channel (RR) as a reference channel.

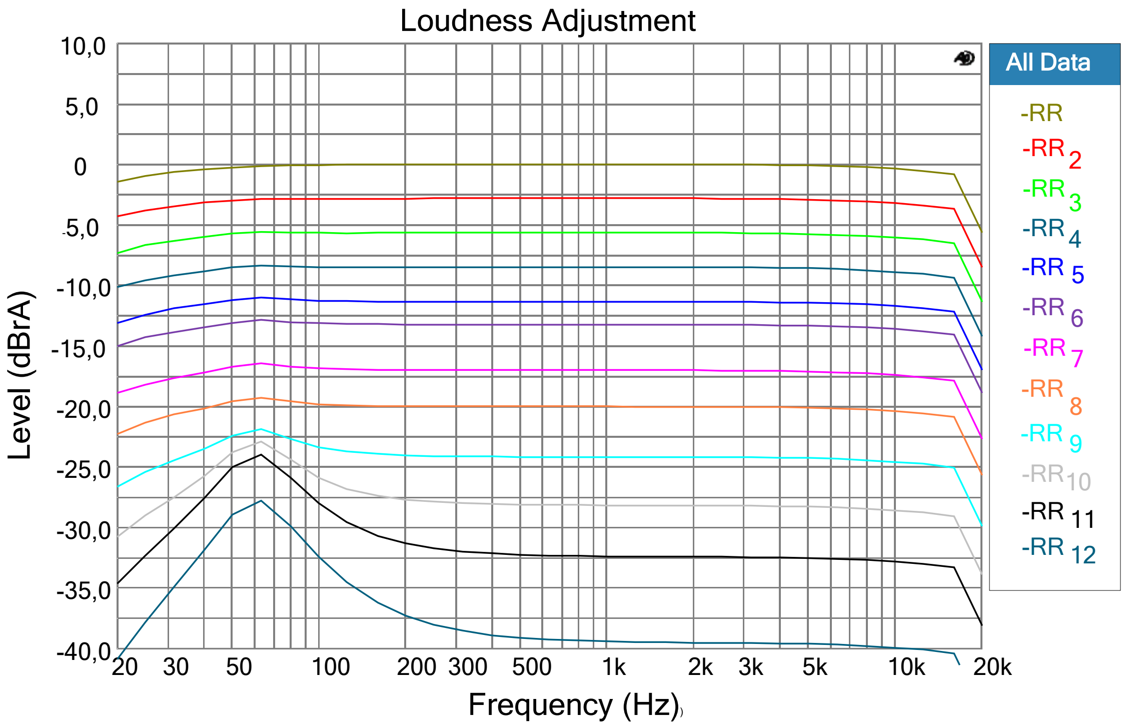


Figure 2.12: Loudness adjustment for twelve volume steps on the rear right channel (RR)

Before running the measurements, it is essential to set all input gain offsets, filters, fader, balance and equalizer settings to zero.

2.4.2 Loudspeaker testing

2.4.2.1 Overview

As mentioned in the introduction of chapter 2.4, the signal undergoes significant changes from its origin as an electric signal until being propagated through air. The speaker acts as an electro-mechanical and mechano-acoustical transducer. Each domain has its characteristic variables and requires different measurement techniques and sensor types:

- Electrical domain: current and voltage sensors
- Mechanical domain: optical laser sensors
- Acoustical domain: microphones

The advantage of describing the model in the electrical domain lies in the robust and reliable measurement of electric quantities and relatively easy ways of calculating the network, but it reflects the mechanical and acoustical system only indirectly.

Figure 2.13 demonstrates a simplified model of the loudspeaker with its properties in the electrical, mechanical and acoustical domain. In this context, two different kinds of measured variables have to be examined in more detail – lumped and distributed parameters. The elements in a lumped system are thought of being concentrated at singular points in space. The classical example is an electrical circuit with components such as resistors, capacitors and inductances. Physical quantities such as current and voltage are time-dependent and two-dimensional.
In contrast, distributed parameters are multi-dimensional and are distributed in space. Variables of distributed models are functions of both time and space. In our case, cone vibration and sound pressure waves in air provide suitable examples.

The mechanical transfer function H(f) describes the relation between voltage u(t) and the resulting displacement x(t) of the coil due to inductive effects.

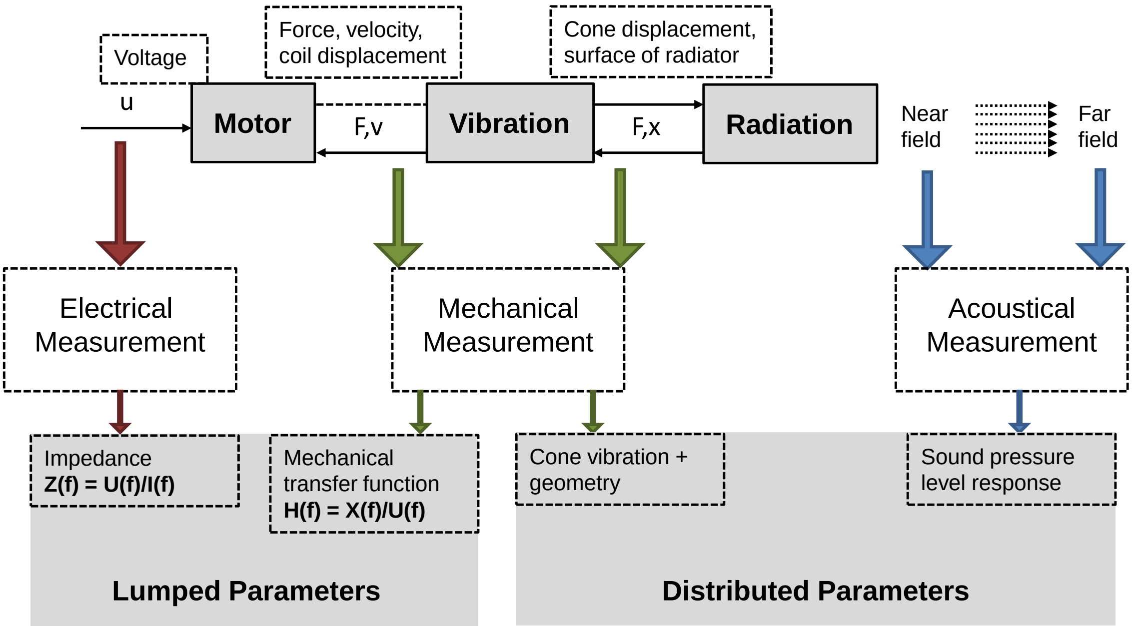


Figure 2.13: Speaker parameters in electrical, mechanical and acoustical domain [cf. KLI14, p.61]

While accessing the sound pressure level responses with microphones is a quite common task, vibration patterns of the cone are less easily accessible. As mentioned on the list on page 18, optical laser sensors are applied to determine mechanical quantities such as cone displacement, vibration patterns and other geometric parameters [KLI06, KLI14, SCH].

2.4.2.2 Test setup with Klippel

Klippel is an innovative German company on the cutting-edge of technology focused on advanced measurement technology for loudspeakers. The loudspeaker generates a high amount of perceptible nonlinearities, especially at high amplitudes. One main focus of Klippel is to develop measurement techniques such as distortion analyzers to facilitate the direct quantification of distortion and its contribution to the overall signal [KLIb].

In automotive audio, Klippel measurement systems are widely used to assess the functionality of loudspeakers applying triangulation laser technology. They are seen as the standard measurement system in the automotive sector providing a reliable reference.

The R&D System (Research and Development) provides tools for research and development of loudspeakers with the goal to improve and accelerate the continuous improvement of hardware and software components during the development phase of a product. By analyzing prototypes, the measurement tool provides insight into characteristics and complex behavior of loudspeakers, including the assessment of lumped parameters, distortion, cone vibration and radiation. A simplified measurement configuration of such a system is described in figure 2.14. The triangulation laser device and a sensitive microphone receive signals from the speaker in the mechanical and acoustical domain simultaneously. The I/O device manages input and output signal and acts as the connecting element to the computer providing the software for analyzing and visualizing data.

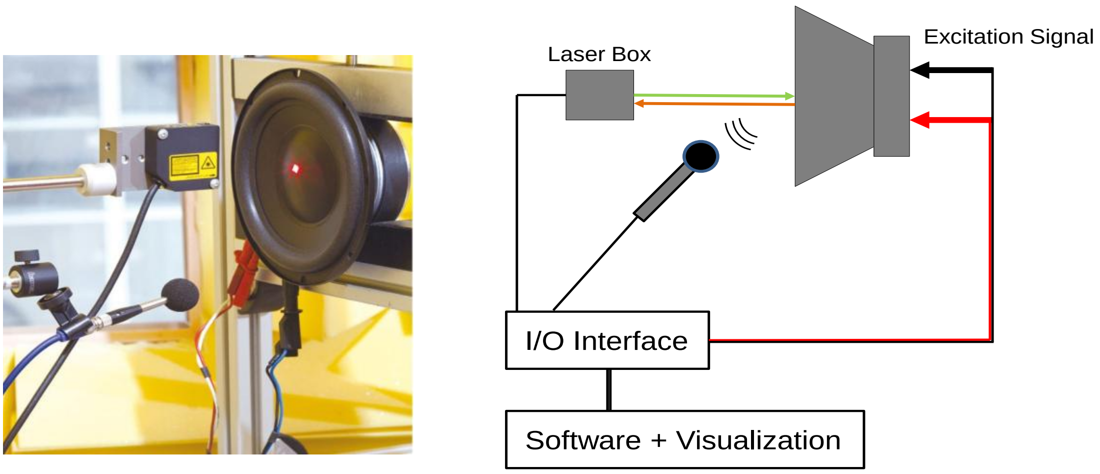


Figure 2.14: Klippel R&D System [LAV, cf. KLIc]

The R&D system is a powerful tool to get access to a wide range of parameters such as resonance frequency, impedance, distortion, frequency response or nonlinear and thermal behavior of the speaker under extreme conditions pushing the amplitude to its limits. In the following section, one example measurement is shown, performed on a full-range speaker [KLI06, KLIb, SCH].

2.4.2.3 Measurement example

Figure 2.15 shows the magnitude of the fundamental and the harmonic distortion components in dB as a function of frequency, plotted into the same figure. The pink curve indicates the total harmonic distortion (THD) which represents the power of all harmonics of the signal in relation to the fundamental frequency. Additionally, the graph also illustrates the level of the second and third harmonics of the signal. As an option, more harmonics can be selected to be measured and displayed. The peak harmonic distortion (PHD) is demonstrated by the red curve. The PHD limit lies 40 dB below the fundamental mean by definition of Klippel. Overall, the plot provides information about the frequency response of the fundamental frequency (dark green) and distortion components in the signal [KLIa].

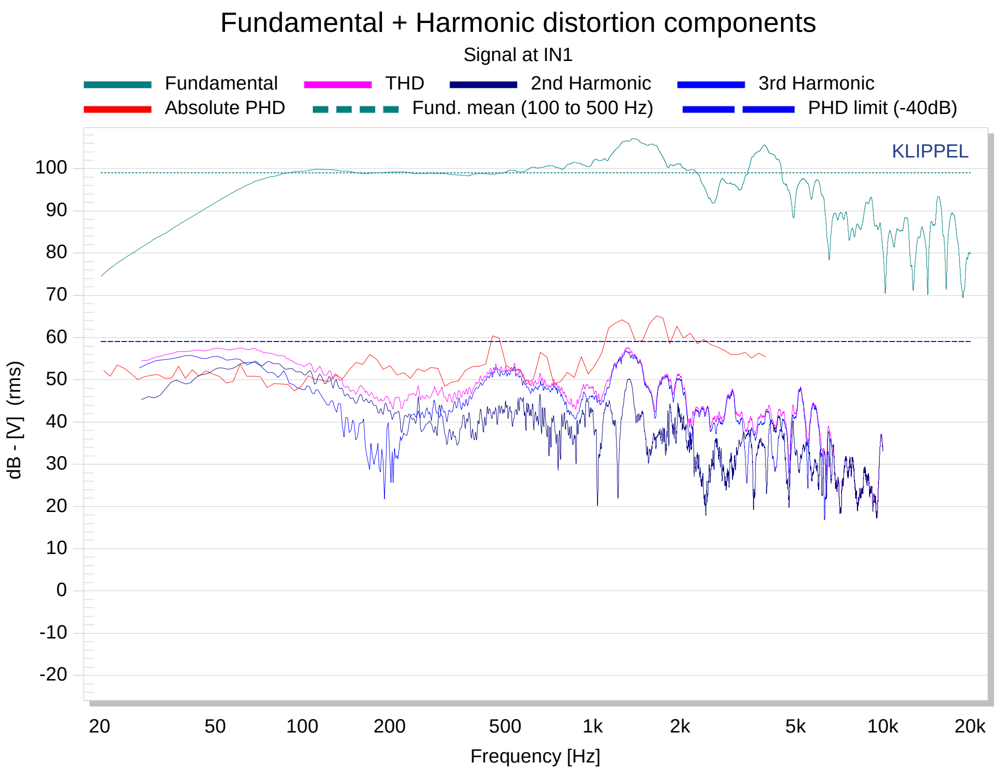


Figure 2.15: Fundamental + Harmonic distortion components in dB as a function of frequency

2.4.2.4 Reliability testing

In order to develop fully functional products, the components have to undergo various test procedures for evaluating the compliance with predefined requirements. At the end of every test, the speaker has to be in a certain predefined functional condition in order to pass the test. A range of signal parameters are measured again and compared to the results of initial testing. The following table shows an overview about several test procedures, including some examples [BAY11].

Testing	Examples
Basic testing	Mechanical and electrical properties such as impedance, resonance frequency, voice coil displacement
Electrical testing	Insulation resistance, behavior in DC voltage operation
Mechanical testing	Drop test, shock test, checking plug connections and tensile strength of cables
Climatic testing	Long-term operation in extreme cold, heat or moist heat, waterproof test
Chemical testing	Speaker structure confronted with petrol, detergents, acids, ethyl alcohol
Durability testing	Endurance and lifetime tests under normal and extreme circumstances

Figure 2.16: Overview about testing procedures and examples

As we can see, there is a variety of test procedures for audio system components before they are approved for the production line. After this step, there are no quality checks anymore for the final product before it goes on sale. In the next chapter, a first attempt is made to build a test tool for a 20-channel sound system which is already approved for production and implemented into the final car.

3 Simulation of an end-of-line test concept for a 20-channel sound system

This chapter describes a test tool implemented in Matlab to get access to sound system properties in the electrical and acoustical domain. At first, a short discussion about the reasons for the necessity of an end-of-line test concept is presented. Next, the 20-channel sound system is explained in more detail, including the Matlab program and the test setup for electrical and acoustical testing. Finally, the results will be shown and analyzed.

3.1 Why end-of-line testing? A comment on tolerances

Why end-of-line testing for automotive sound systems? In the luxury car brand sector, about 10,000 cars are sold a year on average. The majority of these cars are equipped with highly advanced audio systems to provide a high-quality sound experience to the customer. The differences to the common standard audio system are the higher number of channels, more expensive components and especially a more powerful amplification stage. In our particular case, the amplifier runs on 2000 W.

As already mentioned in chapter 2.2.5 on page 5, one of the challenges in the field of automotive audio are the tolerances of component properties which can result in distortion of the desired well-balanced and elaborated sound concept. Not only tolerances of loudspeakers and amplifiers, but also the wiring and connection problems can lead to deficiencies from the perfect system – often called the "golden reference". Even if the products come from the assembly line, they are never exactly the same. Thus, the upcoming question in this matter is the quantification of tolerances and how to understand their influence and importance in the signal chain of a sound system.

The first problem arises with the manufacturer's specifications for a certain product and their methods to ensure to deliver a product complying with the requirements. Common problems are widely varying quality performance and non-standardized measurement methods and equipment across industries and countries. Aforementioned equipment also shows measurement tolerances itself, for example in sensitivity, which can distort the assessment of the true part characteristics. Such deficiencies and problems lead to a certain proportion of parts which are sorted out by mistake even if they work perfectly fine, also called false failures. In the other case however, faulty parts are accepted, so called missed faults. Furthermore, measurements are sensitive to environmental conditions and influential factors such as temperature, humidity, noise or dirt. This can influence the accuracy of the measurement process additionally in a negative way. Overall, considering a non-reliable quality assurance gate, defective parts pass the check and are accidentally introduced into the production line and implemented into the vehicle [CHA14].

But even if all system parts lie within the specified tolerance limits of product variation, the tolerances of each part are added up throughout the whole signal chain in the electrical, mechanical and acoustical domain. This involves the amplifier and the digital-analog conversion (DAC), connections, wiring and the loudspeaker itself, including all its components ranging from voice coil to diaphragm suspension, just to name a few. The tolerance issues in the signal chain can be summarized as follows in figure 3.1 [CHA16]:

Pre-loudspeaker:

Part	Variations
Amplifier and DAC	+/- 0.5 dB typical, +/- 1.5 dB worst case
Cable harness and connectors	+/- 0.2 dB typical, +/- 0.5 dB worst case

Loudspeaker:

Part	Variations
DC resistance	+/- 10%
Resonance frequency (compliance and moving mass variations)	+/- 10%
Sensitivity (moving mass, force factor)	+/- 1.5 dB
Frequency response	+/- 2.5 dB

Figure 3.1: Quantified tolerances in the signal chain [cf. CHA16]

The tables point out that the tolerances of all components are accumulated and can have a negative impact on the overall performance of the system.

In the following section, tolerances for sound systems should be quantified by simulating an end-of-line test process with Matlab.

3.2 System and speaker configuration

Now let us have a closer look at the system to be tested. The advanced sound system for the SUV of a luxury car brand has 20 channels. The amplifier provides 24 channels, but only 20 channels are used. The overall amplification power lies in the order or 2000 W. Figure 3.2 shows a list of the implemented loudspeaker types in the SUV illustrated in figure 3.3. A channel list in figure 3.4 completes the overview.

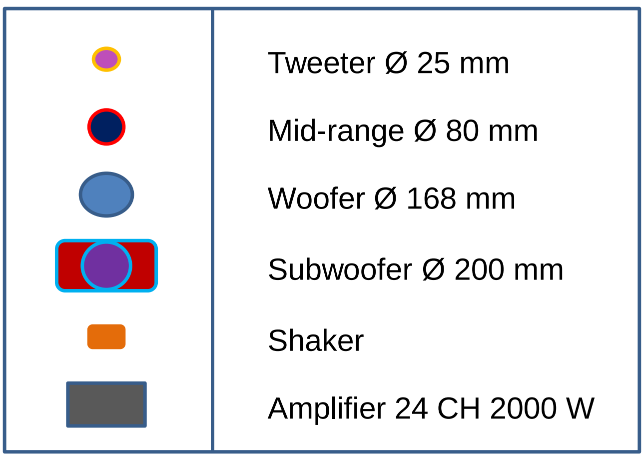


Figure 3.2: Types of loudspeakers

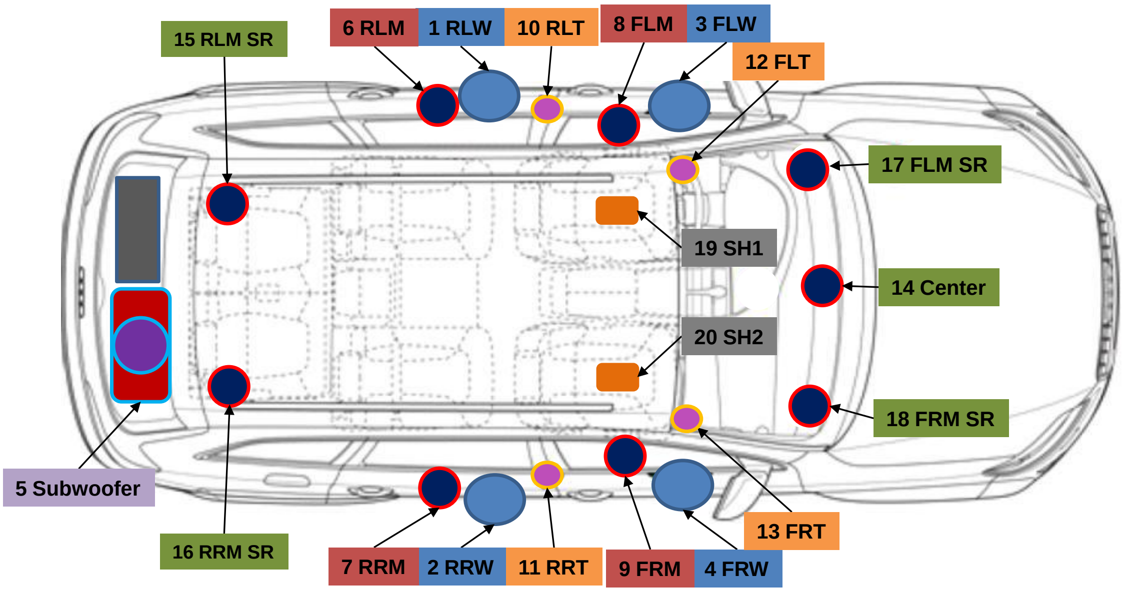


Figure 3.3: SUV sound system with 20 loudspeakers [cf. INT16]

Ch.-No.	Channel Name
1	Rear Left Woofer (RLW)
2	Rear Right Woofer (RRW)
3	Front Left Woofer (FLW)
4	Front Right Woofer (FRW)
5	Subwoofer
6	Rear Left Mid (RLM)
7	Rear Right Mid (RRM)
8	Front Left Mid (FLM)
9	Front Right Mid (FRM)
10	Rear Left Tweeter (RLT)
11	Rear Right Tweeter (RRT)
12	Front Left Tweeter (FLT)
13	Front Right Tweeter (FRT)
14	Center
15	Rear Left Mid Surround (RLM SR)
16	Rear Right Mid Surround (RRM SR)
17	Front Left Mid Surround (FLM SR)
18	Front Right Mid Surround (FRM SR)
19	Shaker 1 (SH1)
20	Shaker 2 (SH2)

Figure 3.4: Channel list

Channel 19 und 20 are reserved for bass shakers in order to support the bass experience by enhancing vibration in the non-audible frequency range below 20 Hz. As standard, they are implemented underneath the front seats.

3.3 Test setup

After a short description of the sound system and speaker configuration in the SUV, the test setup and its components are presented in figure 3.5. The core of the test tool is the Matlab program running on a laptop which is connected to the sound card via Firewire or USB. The output signal of the sound card goes into the Aux-IN connector of the head unit as input source. The link between the head unit and the 20-channel amplifier is built by MOST (media oriented systems transport), an optical protocol for high-speed data transmission which is commonly used in the automotive industry. In the electrical domain, the amplifier outputs are directly connected to the sound card inputs in order to get access to frequency response curves and the system gain without including the speakers. For measurements in the acoustical domain, a microphone in the car cabin is required and the signal is fed back through the microphone input to the laptop.

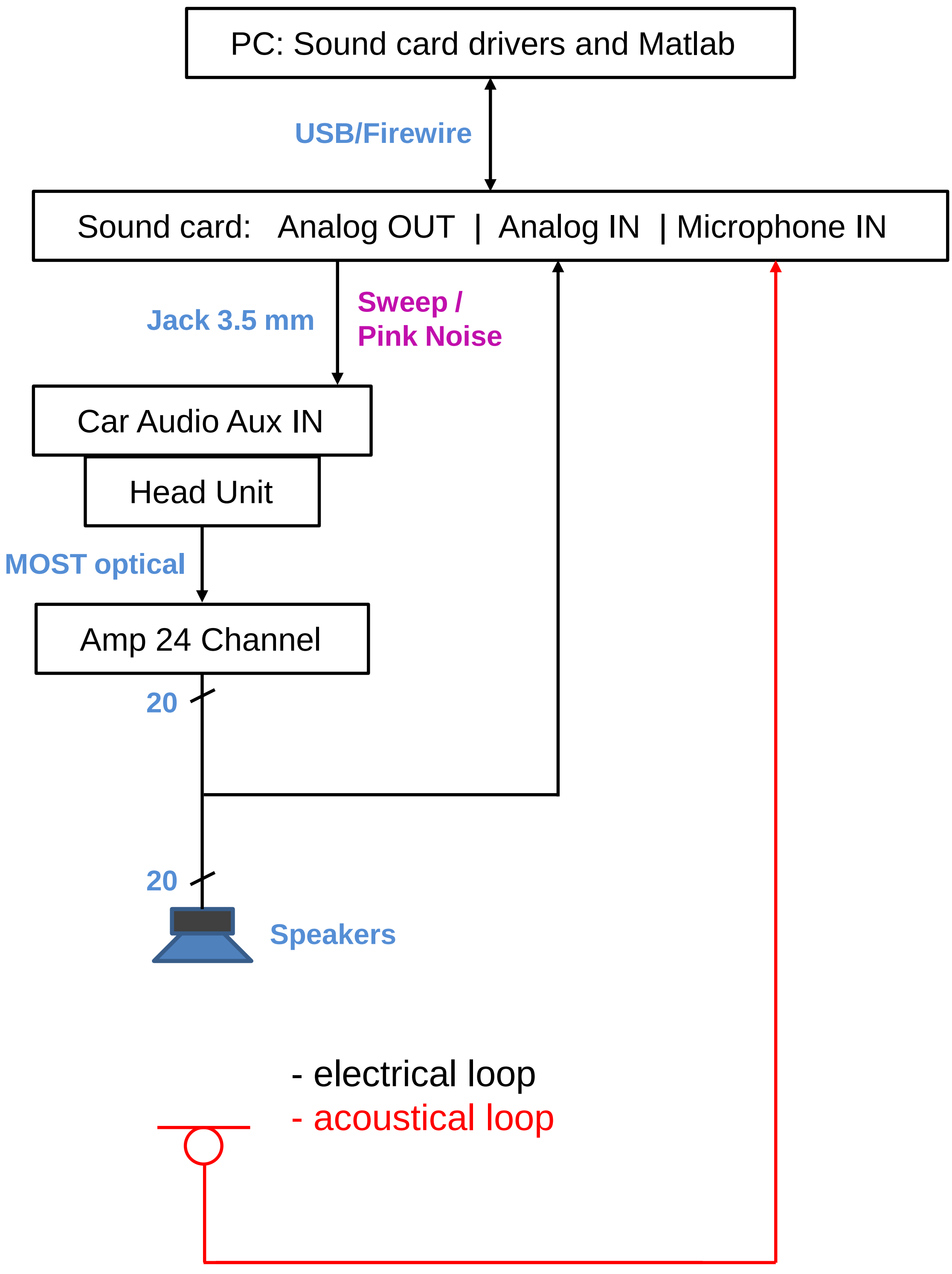


Figure 3.5: Test setup overview with electrical and acoustical loop

3.3.1 Implementation in Matlab

The Matlab program AudioLoopGainCheck.m is the core element of the test process and uses the audio I/O library "Playrec" [HUM14].

3.3.1.1 Initializing and selecting audio drivers

For initializing the test setup, the program detects the available sound configurations on the computer using the function `playrec('getDevices')` and the driver for the desired sound card can be selected, including required input and output channels for playback and recording.

3.3.1.2 Simultaneous audio I/O

For simultaneous audio playback and recording, the function `play_wav`, which already exists in the aforementioned I/O library for audio playback, is extended in order to provide the ability to playback and record audio at the same time. As the name suggests, the command `audioread(filename)` reads data from the file called `filename` and stores the samples in the vector defined as `testsignal`. This vector is played back via the selected output channel of the sound card. At the same time, the selected input channel of the sound card records the incoming sound. Internally, a data buffer is allocated for the recorded data stream and the data is then retrieved by the command `getRec`. The main function for running the test is shown below.

```
function [ rec_data ] = playrecord_wav( fileName_play,
playDeviceID, recDeviceID, playChanList, recChanList, fs)
% fileName_play: name of the file to be played
% playDeviceID: audio device ID for playback
% recDeviceID: audio device ID for recording
% playChanList: channel number for playback
% recChanList: channel number for recording
% fs: sampling frequency
```

3.3.1.3 Test signals

In the test tool, two signals are available for the test run, normalized pink noise for five seconds and an exponential sweep from 20 Hz to 20 kHz for ten seconds at a sampling frequency of 44.1 kHz each. Pink noise was discussed to be better than white noise due to a more balanced power spectrum density across the frequency range.

White noise has a flat power spectrum density, which means that every frequency has the same power. For example, on the logarithmic frequency axis, the octave from 10 to 20 kHz contains a lot more energy than the octave from 20 to 40 Hz because there are more frequencies involved, which makes it sound quite sharp. To counteract the high amount of power in the high frequency range, pink noise was introduced. As a filtered version of white noise, the energy is constant across octaves, so the power spectrum density is decreasing with increasing frequency. Pink noise sounds less sharp and more natural and is not a potential hazard for small sensitive tweeters because the amount of energy in the high frequency range is lower compared to white noise. As an additional aspect, pink noise is also closer to human auditory perception. For these reasons, pink noise is commonly used for audio calibration purposes [FOL14, SWE00].

The test signal will be the same for all types of speakers. The signals are played back through the selected output channels of the sound card.

3.3.1.4 Data analysis: Frequency response and level comparison

When comparing the vector `testsignal` and the recorded data vector `rec_data`, a range of properties of the transmission system between input and output is accessible. In the electrical domain, the system is defined by the amplifier and the connecting cables only. From the acoustical point of view, the system also includes all the wiring, connections, speakers, the car cabin and the microphone. A short overview is shown in figure 3.6.

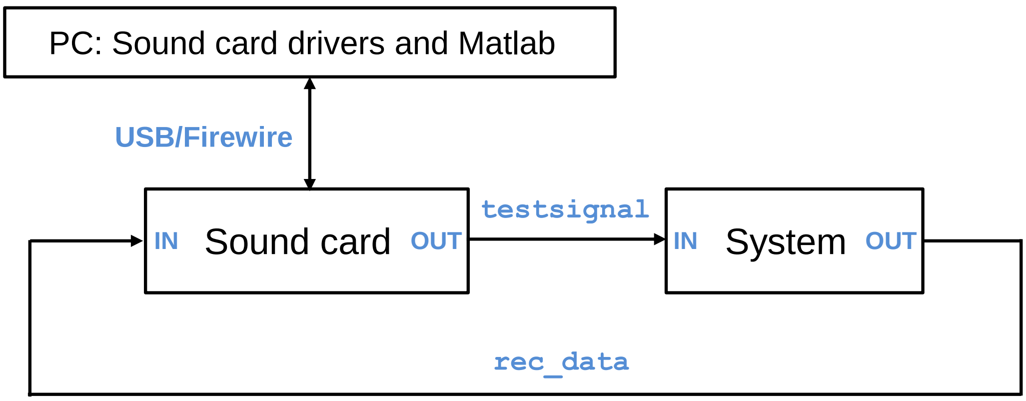


Figure 3.6: Accessing system characteristics using input and output signal

In summary, the test tool is used to measure two system properties.

Frequency response with a sweep

With the acquired input and output data, the frequency response of the system can be calculated by dividing the output spectrum by the input spectrum in frequency domain, using the sweep as input signal.

$$H(f) = \frac{Y(f)}{X(f)} = \frac{FFT\{rec\_data\}}{FFT\{testsignal\}}$$

In order to obtain a clear result, a smoothening filter is applied before the magnitude of the frequency response is plotted.

In order to check the test setup without the device under test, the test is carried out with the output channel directly connected to the input channel to obtain the frequency response of the sound card itself. Since the audio driver and the program are affected by latency, input and output signal are not perfectly aligned. As a consequence, the last approximately 0.5 seconds of the recorded signal are missing since the recording process stops in the moment the playback process is finished and the last samples of the test signal are still on the way to be recorded. To solve the problem and to record the complete sweep, one second of zeros is appended to the input signal in order to extend the recording time. Figure 3.7 shows the spectrograms of the input and the output signal with zero-padding. In figure 3.8, the frequency response of channel 1 of the sound card is presented. As assumed, the curve is perfectly flat in the audible frequency range. At 20 kHz, the anti-aliasing low pass filter can be seen clearly.

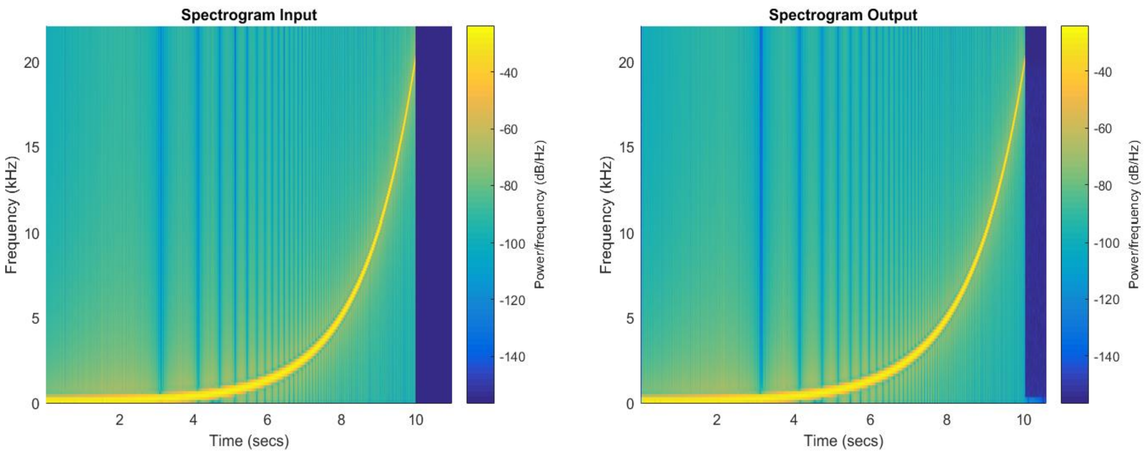


Figure 3.7: Spectrograms of input (testsignal) and output (rec_data)

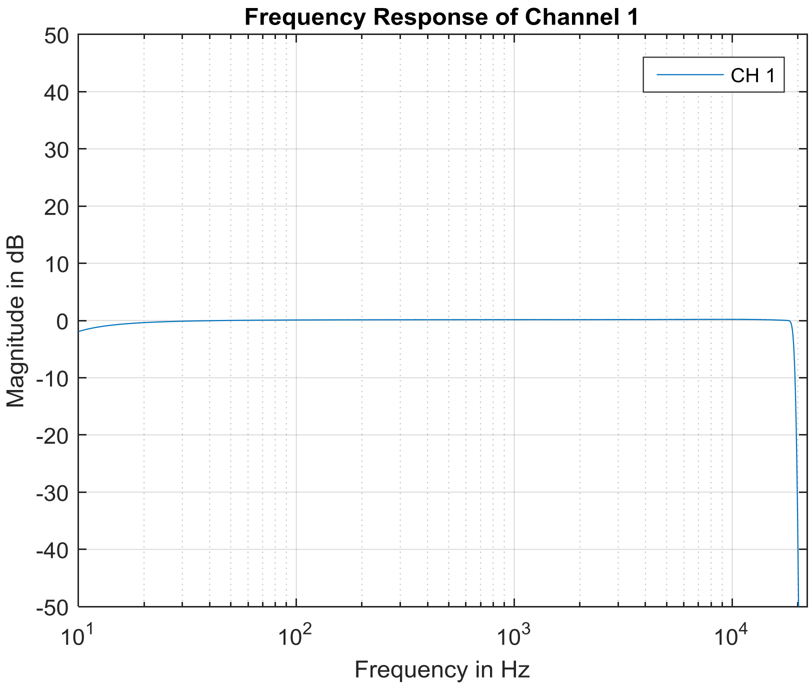


Figure 3.8: Frequency response of channel 1

Level comparison with pink noise

By applying pink noise to the system, the output can be seen as a pink noise signal with a certain root mean square (RMS). Goal of the measurement is to obtain a certain value of each channel which can be taken as a reference value for the comparison of a number of theoretically equal systems – in our case a certain number of amplifiers or car sound systems to be compared. Thus, the results are not absolute values. In order to deal with the latency issue and any transient phenomena, the first and the last 0.5 seconds of the recorded pink noise signal are cut out and the RMS of the windowed signal is calculated as presented in figure 3.9. The RMS is shown by the red line.

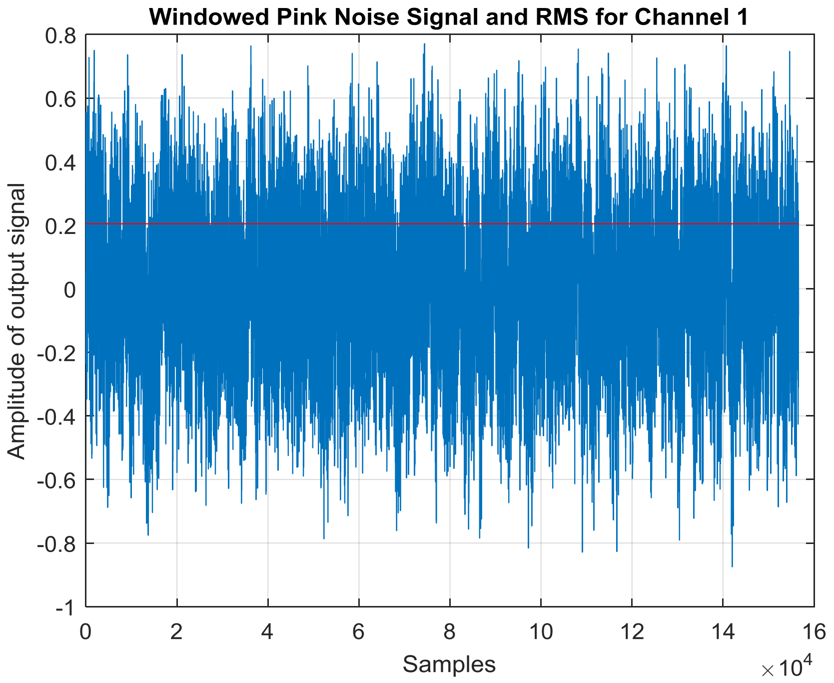


Figure 3.9: Recorded pink noise, windowed, with root mean square (red line)

Now we want to focus on the detailed measurements in the electrical and acoustical loop.

3.3.2 Electrical loop

As already mentioned in chapter 3.3.1.4, the system to be measured in the electrical domain is composed of the amplifier and the connecting cables as shown in figure 3.10. For this measurement, the amplifier is measured outside the car and is operated in a car-like environment by simulating power supply, ignition and data streams to ensure full functionality on the test rig. Figure 3.11 and 3.12 provide more insight into the test setup. The sound card used for this measurement is a Fireface 800 with a firewire connection.

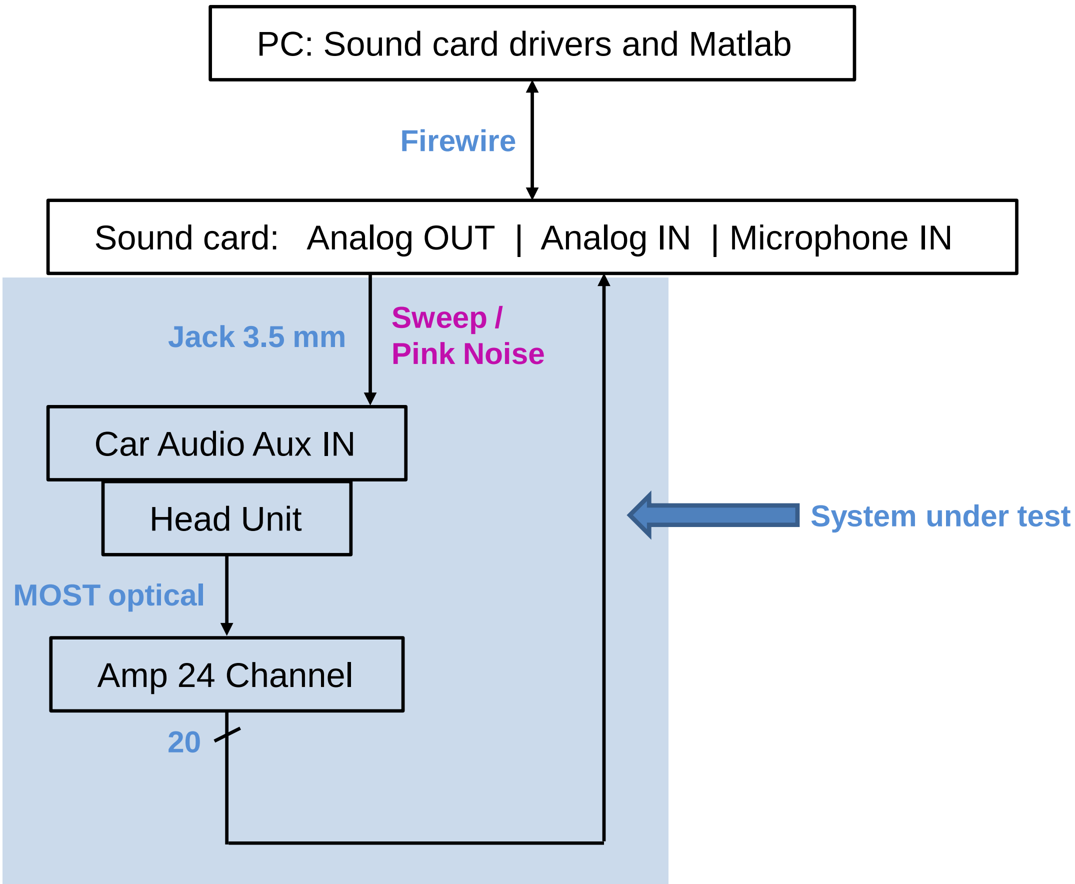


Figure 3.10: Electrical loop: system under test

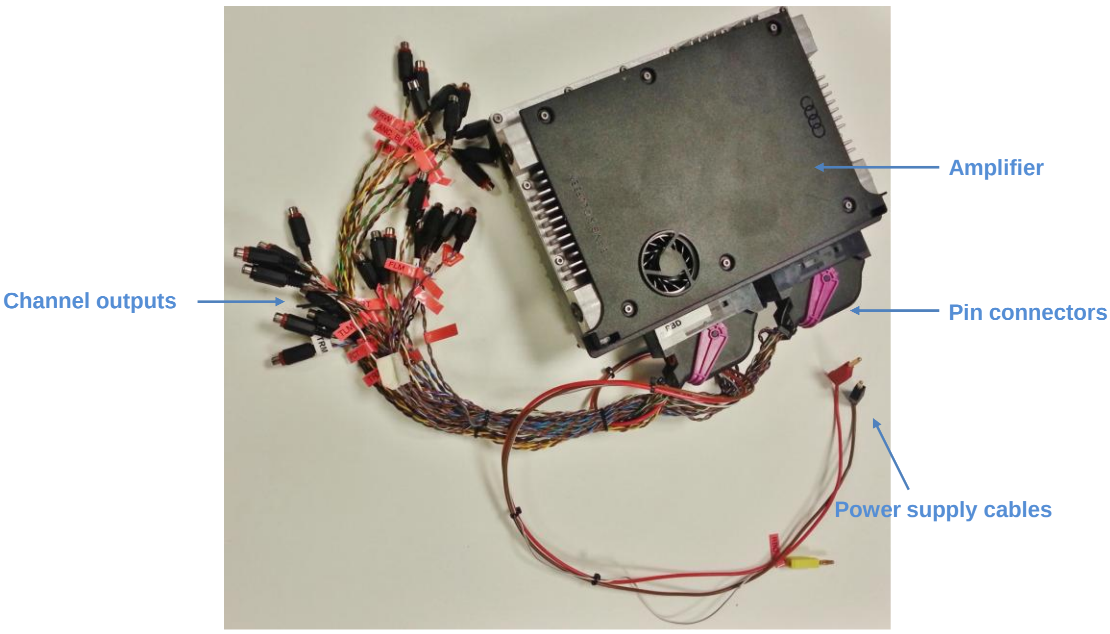


Figure 3.11: Electrical loop: amplifier and pin connectors

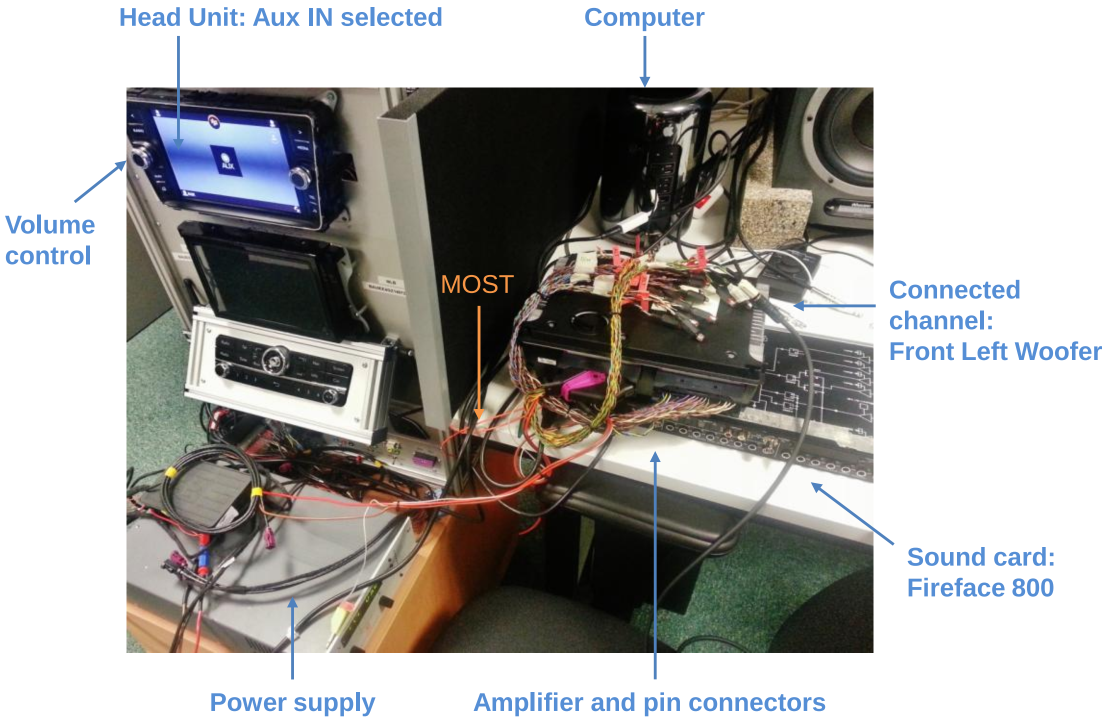


Figure 3.12: Electrical loop: test rig

In this measurement, frequency responses and level comparison values of the amplifier are measured for all channels. The test process and results are described in chapter 3.4.1.

3.3.3 Acoustical loop

For acoustical testing, the sound system is taken as a whole with all its components. Thus, the system is composed of the amplifier, connectors and wiring, speakers, the car cabin and the measurement microphone as illustrated in figure 3.13. Now, the measurement is conducted in a real environment in the actual car. A microphone is mounted on the headliner.

For this measurement, all channels have to play the test signal separately. To achieve this, a special software specified as the tuning tool for the amplifier is used to control each channel on its own. The tuning tool is provided by the supplier and is a computer program which connects to the hardware and allows access to its functions. The name of the tool is derived from its intended purpose. By having access to all channels of the amplifier separately, the sound engineer can tune the sound system, create filters and listen to the results for every channel consecutively. In this specific test case, all channels are muted except for one channel playing the test signal.

Figure 3.14 demonstrates the test setup in the car. For mobility reasons and the necessity of one test channel only, the Fireface was replaced by a 2-channel Tascam US-122MKII sound card connected via USB.

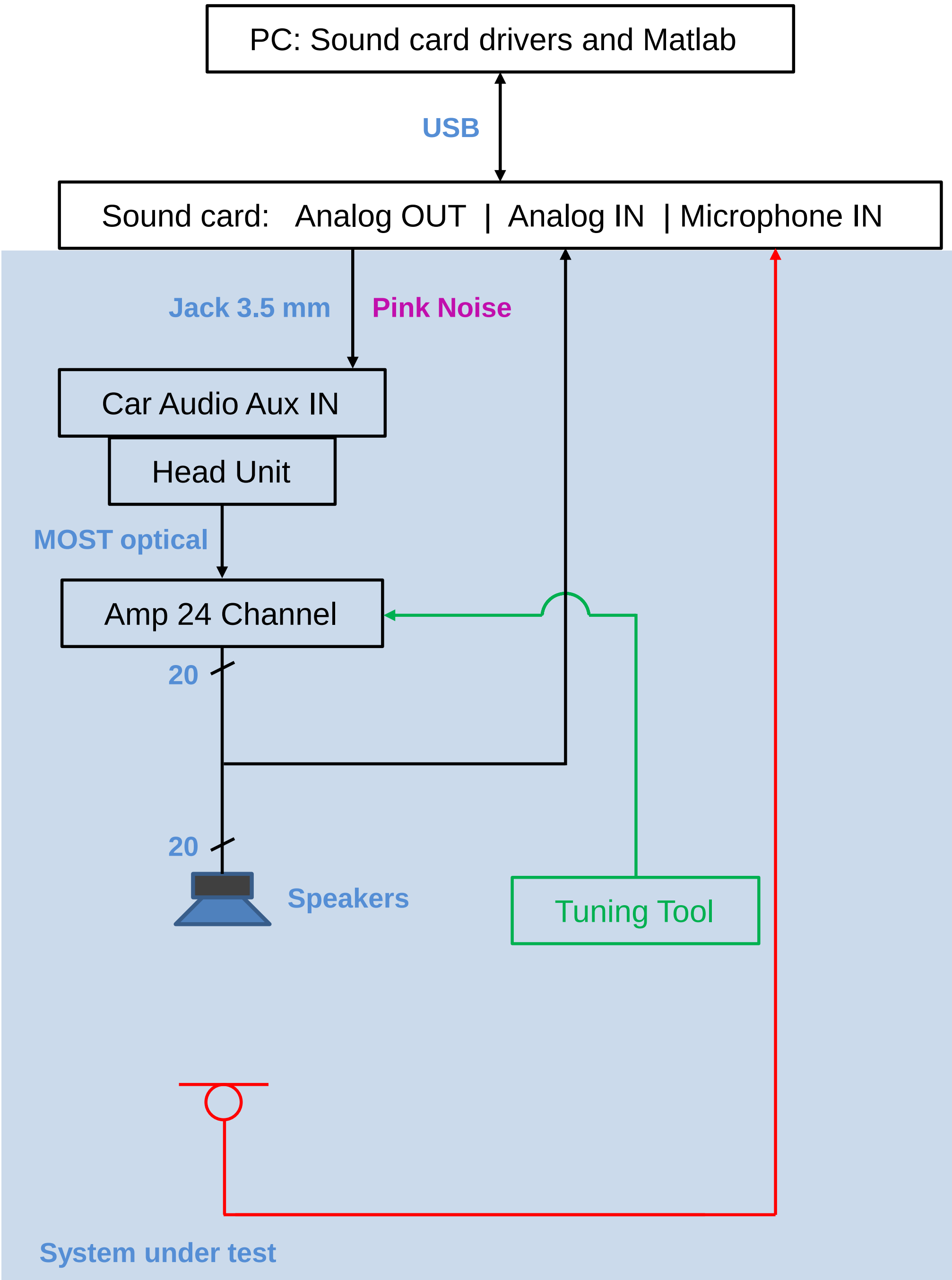


Figure 3.13: Acoustical loop: system under test

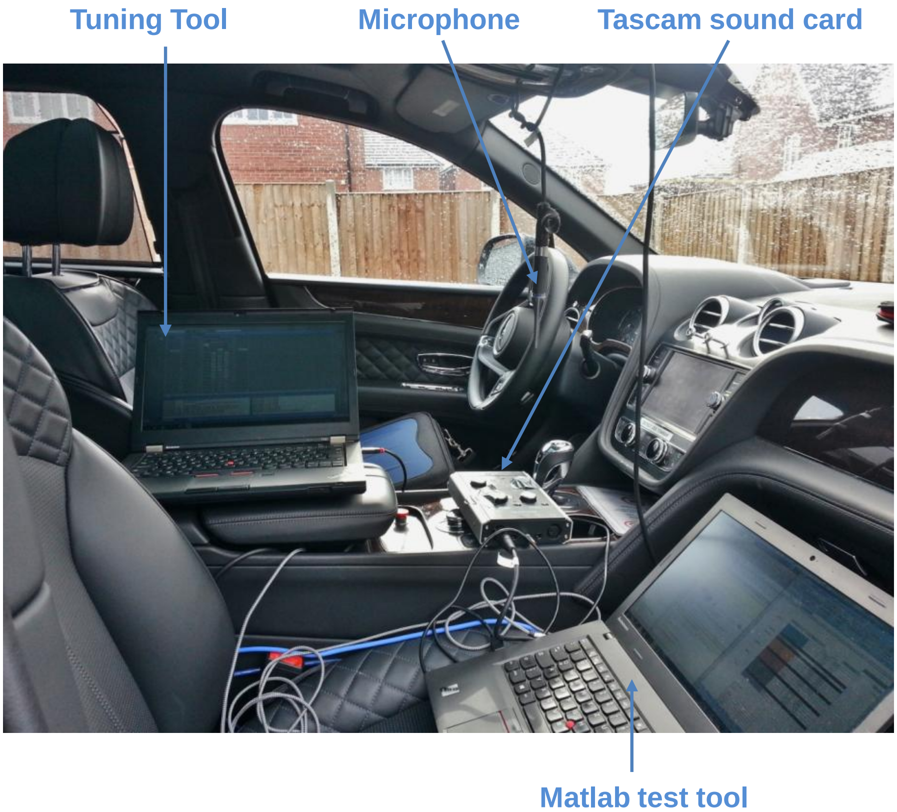


Figure 3.14: Acoustical loop: test setup in the car cabin

In the acoustical domain, only the channel-wise level test is applied. More information about the test process and the results are given in chapter 3.4.2.

3.4 Test results

3.4.1 Electrical loop

For assessing properties in the electrical domain, four amplifiers of the same type are available to be examined on the test rig in regard to frequency response and output levels.

3.4.1.1 Frequency responses

At first, one amplifier is taken to calculate the frequency response for each channel with a sweep. In figure 3.15, the recorded output for woofer, mid-range and tweeter channel can be seen in spectrograms.

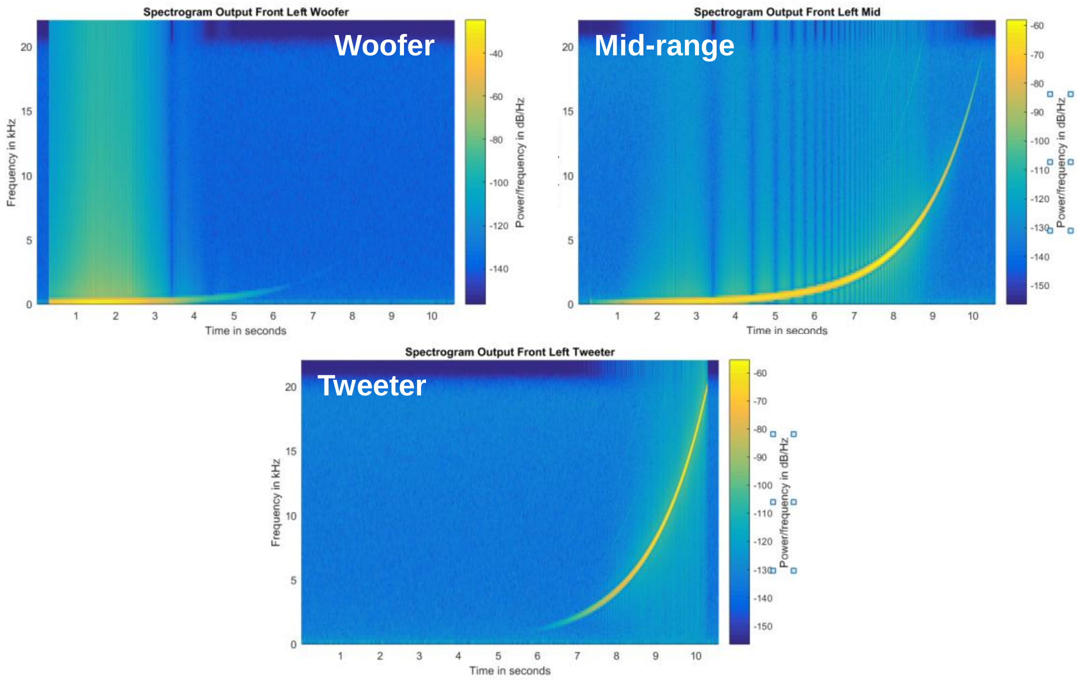


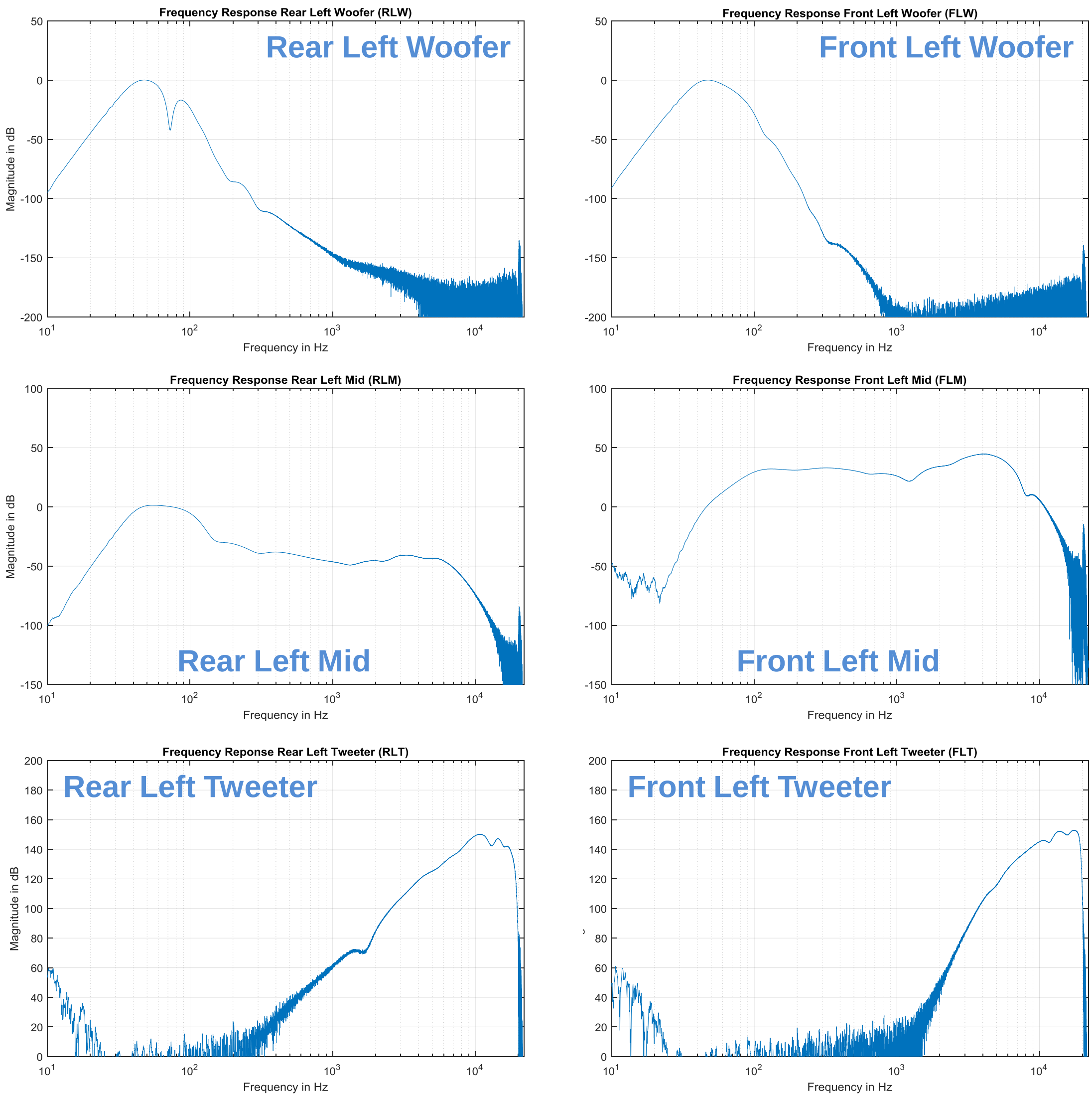
Figure 3.15: Spectrograms: recorded output for woofer, mid-range and tweeter channel

Next, frequency response curves are plotted for every channel. Every channel shows the typical curve depending on the speaker type it will be connected to in the car. Inside the amp, every channel has its assigned equalizer preset tuned by the audio engineer during development of the sound system. This equalizer tool, which is part of the tuning tool mentioned before, can have up to a certain number of filters per channel such as low pass, high pass, peak and notch filter which can be combined all together for designing the desired filter curve as a result.

All frequency responses are symmetrical between left- and right-side speakers. Frequency response curves are checked for four amplifiers of the same type. Results for all four amplifiers show equal behavior and no differences between the devices can be found.

An overview about the equalizer presets and curve characteristics is demonstrated in figure 3.16 focusing on eight examples. It shows two woofers, two mid-range speakers and two tweeters implemented in the front and in the rear of the car with additional two channels for subwoofer and center. Whereas the curve shape in the characteristic frequency range for each speaker is quite clear, noise affects the outcome when the signal is attenuated due to the EQ.

Having a closer look at the filtering, slight notches and peaks can be seen, for example the notch filter for the rear left woofer at about 75 Hz and for the front left mid speaker at a frequency slightly above 1 kHz. This is designed to suppress resonance frequencies or disturbing modes which have been detected in the car cabin.



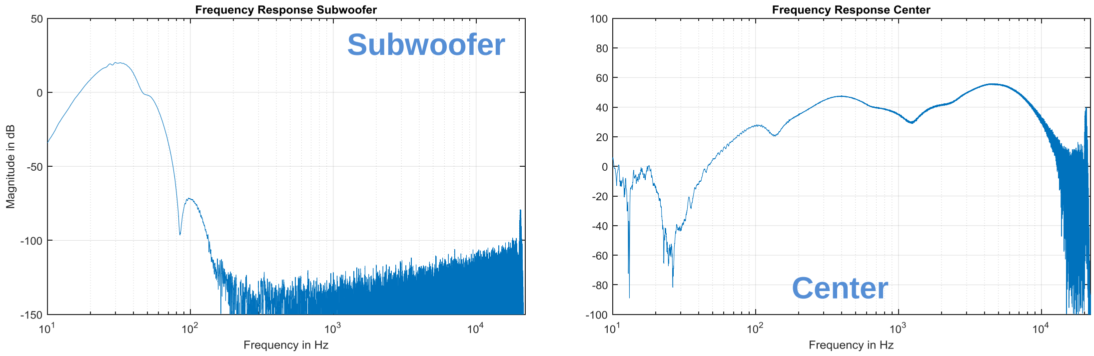


Figure 3.16: Frequency responses for rear and front left woofer, rear and front left mid, rear and front left tweeter, subwoofer and center;
x-axis: frequency in Hz, y-axis: magnitude in dB

Extended statistical model to calculate frequency responses

As we can see in the figures above, noise is affecting the system identification. In these examples on the test rig, the frequency response can be seen clearly despite of the noise in the background. However, if the measurement is conducted in a less stable environment, noise can become dominant and distort the result significantly. For this case, the suggestion is made to extend the frequency response calculation by a statistical noise model using power spectrum densities (PSD) of the signals [SYS14].

As for now, we use a simple spectral division between input and output signal:

$$H(f) = \frac{Y(f)}{X(f)} = \frac{FFT\{rec\_data\}}{FFT\{testsignal\}}$$

This calculation method is highly susceptible to failures, distortion by non-linearities and faces the risk of a division by zero. In the extended model, the frequency response is estimated by averaged power spectrum densities of input and output signal. In order to obtain a PSD of a signal, the autocorrelation of the signal has to be calculated first as shown below for the input signal $x(t)$:

$$R_{xx}(\tau) = \lim_{T\to\infty} \frac{1}{T} \int_{-T/2}^{+T/2} x(t) * x(t+\tau)dt$$

According to the Wiener-Chintschin theorem, the Fourier transformation of the autocorrelation $R_{xx}(\tau)$ is equal to the power spectrum density of the signal $x(t)$:

$$S_{xx}(j\omega) = \int_{-\infty}^{+\infty} R_{xx}(\tau) * e^{-j\omega\tau}\, d\tau$$

Combining input and output signal, the absolute magnitude of the transfer function can be described as follows:

$$|H(j\omega)|^2 = \frac{S_{yy}(j\omega)}{S_{xx}(j\omega)}$$

$S_{xx}(j\omega)$ and $S_{yy}(j\omega)$ are probabilistic properties, based on a finite signal in time domain. They are calculated by averaging a finite number of periodograms. A periodogram is defined as an estimate for the spectral density of a signal, determined by a finite number of non-zero samples taken out of random pieces of the time-domain signal.

This overall approach does not include phase information for the frequency response estimate since the latter is represented by real values only. Therefore, the PSD method is extended. The new model includes two estimates for the frequency response, considering unknown additional noise at the input and the output of the system, respectively. The basic principle of this approach is to have a look at the direct relation between input and output by applying the cross-correlation function in the time domain. The calculation procedure is the same as before. Figure 3.17 describes the signal flow in the model with additional noise at the input and output. $X(t)$ and $y(t)$ are the accessible properties. The noise is described by $n_x(t)$ and $n_y(t)$.

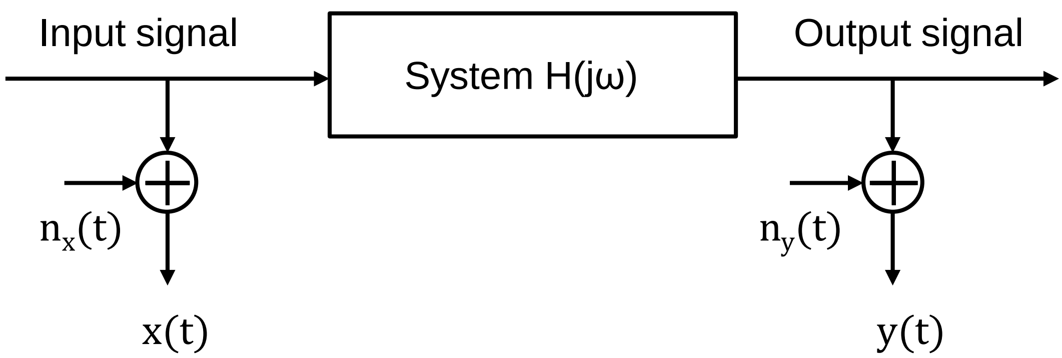


Figure 3.17: Noise model: noise at the input and output of the system [cf. SYS14]

a) Noise at the output of the system:

The estimated frequency response considering noise at the output of the system contains the transformed cross-correlation (CCF) function between input and output divided by the PSD of the input. This estimate of the frequency response minimizes the influence of the unknown noise at the output when measuring the output signal.

$$H_1(j\omega) = \frac{S_{xy}(j\omega)}{S_{xx}(j\omega)}$$

b) Noise at the input of the system:

As specified in the procedure before, the estimate is calculated by the PSD of the output divided by the transformed CCF between output and input, minimizing the effect of the noise at the system input.

$$H_2(j\omega) = \frac{S_{yy}(j\omega)}{S_{yx}(j\omega)}$$

Both estimates $H_1(j\omega)$ and $H_2(j\omega)$ include information about the phase response of the system. By the averaging procedures, the errors at the input and output are eliminated and both estimates converge to the true frequency response $H(j\omega)$.

Generally speaking, by extending the mathematical model for frequency response calculation, the influence of noise on the measurement can be reduced. Within the EOL testing procedure, however, noise reduction is not crucial in our case since the results are clear enough to draw conclusions for the future development of an EOL process.

3.4.1.2 Channel-wise level comparison

In this test section, levels of channels are examined. The RMS level of every channel output is measured in order to be able to compare output levels of different devices directly. Four amplifiers are taken to carry out the comparison. As they already showed the same frequency response curves, it is not surprising that the levels for all 20 channels are almost equal as well.

The bar chart in figure 3.18 presents the direct comparison between amplifier I and II for eight left channels as an example. Amplifier I is taken as a reference. On top of each bar, the level difference is shown in dB. The largest difference can be found for the rear left mid surround speaker with an absolute difference of 0.24 dB. Amplifier II has slightly lower levels than amplifier I.

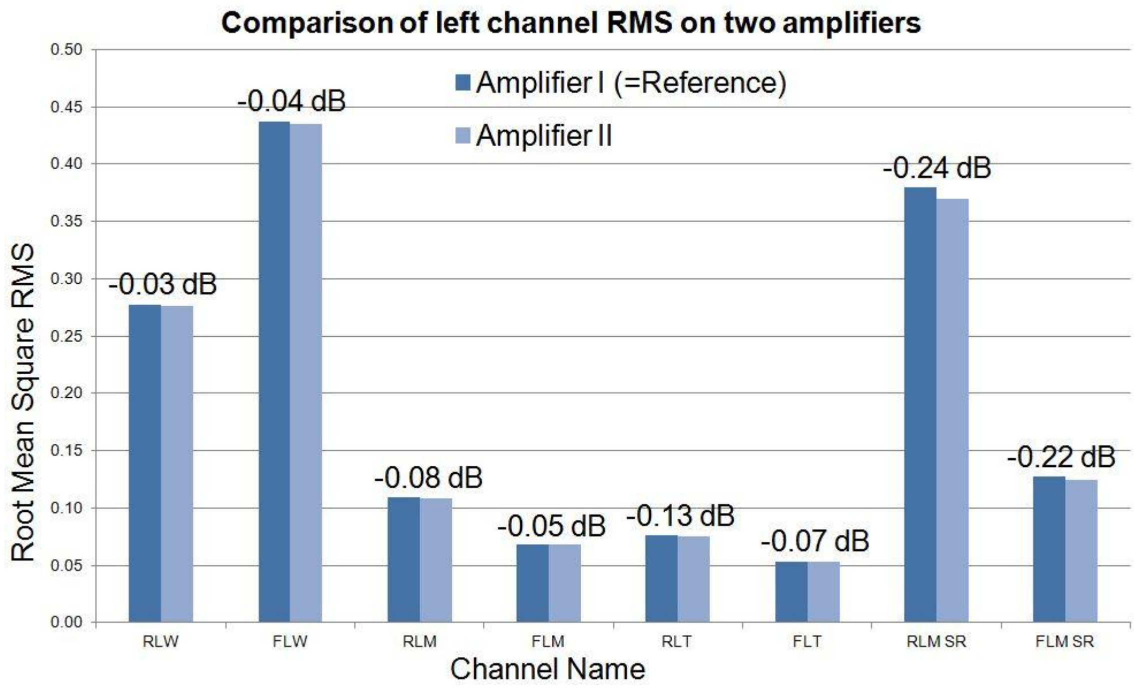


Figure 3.18: Level comparison between two amplifiers for rear and front left woofer, rear and front left mid, rear and front left tweeter and rear and front left mid surround speaker

3.4.1.3 Summary and conclusion

From these results we can draw the first relevant conclusion for further end-of-line testing processes. The measurements show that there are only small differences between devices which can be considered as negligible. Therefore, electrical testing is not crucial within the EOL process because tolerances are considered to be in the order of a few tenths of decibels or even less. This does not affect the auditory experience. It is not necessary for this test to implement the extended noise model explained in chapter 3.4.1.1.

3.4.2 Acoustical loop

In the last section, tolerances in the electrical domain have been stated not to have a significant impact on the overall outcome since the measured level differences between devices are negligibly small. From the acoustical point of view, more components play a role in modifying the acoustical output of the system, so more deviations are to be expected.

Five cars are taken as measurement objects, four left-hand drive and one right-hand drive car. They all have the same sound system implemented with equal configurations. Since there was no anechoic chamber available for testing, an approximately quiet environment had to be found for each of the measurements. Therefore, most of the measurements have been conducted at night and outside the workshops. Before running the test, each sound system had to undergo a subjective listening test with well-known test tracks in order to ensure to test a fully functional installation in every car. The average outside sound pressure level has been calculated as 47 dB SPL.

For acoustical testing, only the channel-wise level comparison is applied for all five cars. In contrast to electrical testing, the RMS is converted into sound pressure levels in dB to provide a better idea about the outcomes since we are more used to values in dB SPL.

The following table in figure 3.19 shows the sound pressure levels for all five cars for all channels without shakers. In all five cars, both tweeters in the front did not play back any output signal. This can be explained by an error in the current software installed on the amplifier. This mistake was eliminated in subsequent software versions. Unfortunately, cars with the updated software version were not available. Thus, relevant test results are based on the measurements of 16 channels only.

In order to conduct level comparisons between cars, a reference car has to be found and defined. In our test case, Car 3 is determined to be the golden reference, marked in light green on the table in the middle column.

Sound pressure levels in dB SPL

Channel	Car 1 (L)	Car 2 (L)	Car 3 (L) - Ref	Car 4 (L)	Car 5 (R)
RLW	76.39	76.14	75.41	76.30	76.41
RRW	76.71	75.88	75.44	75.90	75.86
FLW	85.18	84.64	84.15	84.87	84.02
FRW	76.34	83.31	82.62	83.85	78.38
SUB	88.69	90.09	87.81	88.47	88.05
RLM	59.55	60.10	58.87	59.67	60.79
RRM	60.03	59.97	58.52	59.97	60.60
FLM	75.14	74.36	74.78	74.50	73.48
FRM	73.68	73.05	73.30	73.72	73.93
RLT	63.58	65.85	62.62	64.00	61.28
RRT	62.18	63.58	61.39	62.22	62.54
FLT	no signal	no signal	no signal	no signal	no signal
FRT	no signal	no signal	no signal	no signal	no signal
Center	69.85	68.40	69.97	69.62	69.89
RLM SR	58.89	57.86	57.84	58.22	61.15
RRM SR	58.01	57.97	57.28	57.53	59.37
FLM SR	68.06	71.03	69.53	67.91	70.64
FRM SR	70.22	70.06	69.80	70.06	69.92

Figure 3.19: Sound pressure levels in dB SPL for all channels measured in four left-hand drive cars (L) and one right-hand drive car (R); Car 3 defined as reference (colored in green)

3.4.2.1 Comparison of left-hand drive vehicles

At first, the four left-hand drive cars are investigated in terms of sound pressure levels for every channel. The table in figure 3.20 gives an overview of the calculated level differences between Car 1, 2 and 4 in comparison to the reference car. Level differences greater than 1 dB are marked in orange. If the mismatch exceeds 2 dB, cells are colored in red.

Level differences in dB

Channel	Car 1 ↔ 3	Car 2 ↔ 3	Car 4 ↔ 3
RLW	0.98	0.73	0.89
RRW	1.26	0.44	0.46
FLW	1.02	0.49	0.72
FRW	-6.28	0.69	1.23
SUB	0.88	2.28	0.66
RLM	0.67	1.23	0.80
RRM	1.51	1.44	1.45
FLM	0.36	-0.41	-0.28
FRM	0.39	-0.25	0.42
RLT	0.95	3.22	1.37
RRT	0.79	2.19	0.83
FLT	no signal	no signal	no signal
FRT	no signal	no signal	no signal
Center	-0.12	-1.57	-0.35
RLM SR	1.05	0.02	0.38
RRM SR	0.73	0.69	0.25
FLM SR	-1.47	1.50	-1.63
FRM SR	0.43	0.26	0.26
Average	**1.18**	**1.09**	**0.75**

Figure 3.20: Level differences in dB of Car 1, 2 and 4 in reference to Car 3

When comparing the results, Car 4 provides the best results and closest alignment to the golden reference. Car 1 and Car 2 show deviations higher than 2 dB (red). The average level difference can be stated as 1.18 dB, 1.09 dB and 0.75 dB, respectively.

Listening tests in the car cabin examined the impact of these level differences on auditory perception. With well-known music tracks as a test signal, the gain of one channel in the front has been changed with the tuning tool in the order of 0.5 dB up to 6 dB. In the case of headphones as playback device, the just noticeable difference for level differences between stimuli has been examined to be around 2 dB for a group of subjects with average auditory skills [ZHI12].

In our case considering the car cabin, a change of 0.5 dB for one channel was not perceptible. However, channel gain modification of 1.5 dB had an impact on localization of the sound source and this leads clearly away from the optimal sound reproduction. The test was conducted with two experienced audio engineers as test subjects.

When investigating the level differences in figure 3.20, it is clearly provided that these differences have a significant influence on the overall system performance. Furthermore, it has to be considered that there is not only one channel being affected. Tolerances are added up in total for all channels. For example, if the mismatch for the front left woofer is +1 dB and for the front right woofer -1 dB, the total mismatch adds up to 2 dB. The maximum difference in our test case is 6.28 dB for the front right woofer of Car 1 compared to the golden reference. Additionally, Car 2 shows level differences greater than 2 dB three times. The levels of the subwoofer and the two tweeters in the rear are both too high.

Figure 3.21, 3.22 and 3.23 illustrate the sound pressure level differences between Car 1, 2 and 4 in comparison to Car 3 as a reference for all available channels. The value above each bar shows the level difference in dB. As pointed out in figure 3.18, difference values above 1 dB are marked in orange, differences greater than 2 dB are highlighted in red.

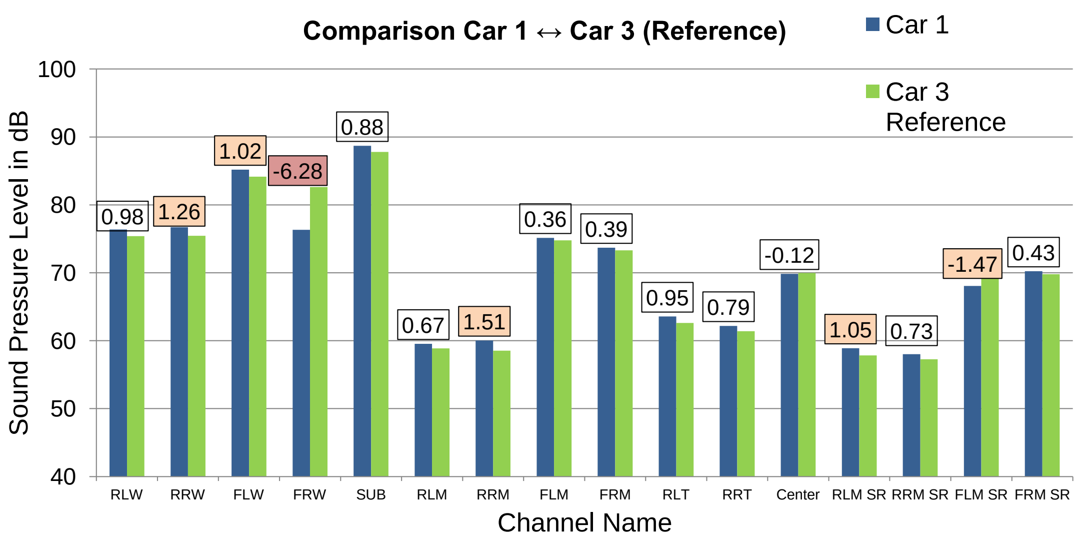


Figure 3.21: Level differences in dB between Car 1 and Car 3 (Reference)

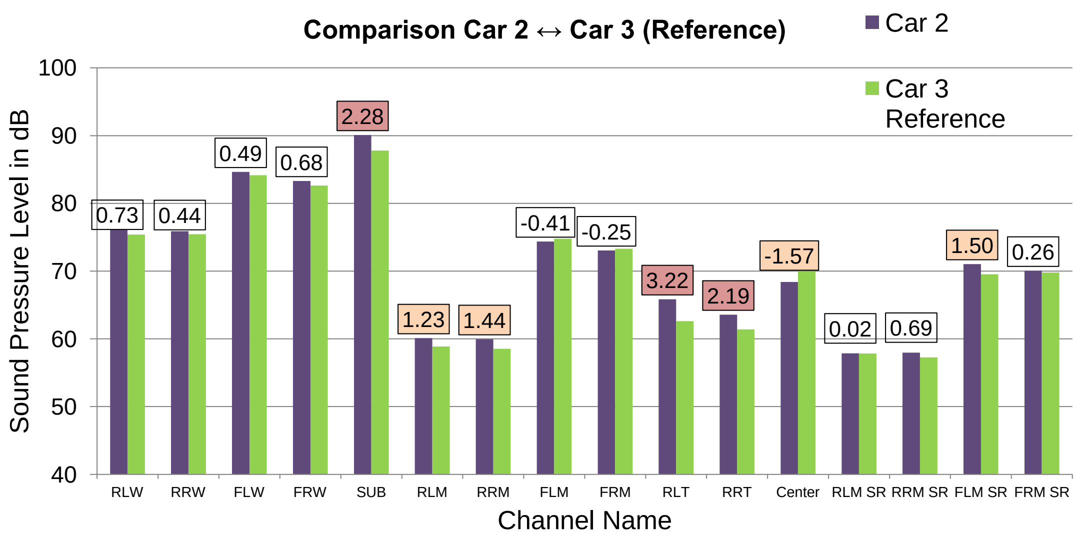


Figure 3.22: Level differences in dB between Car 2 and Car 3 (Reference)

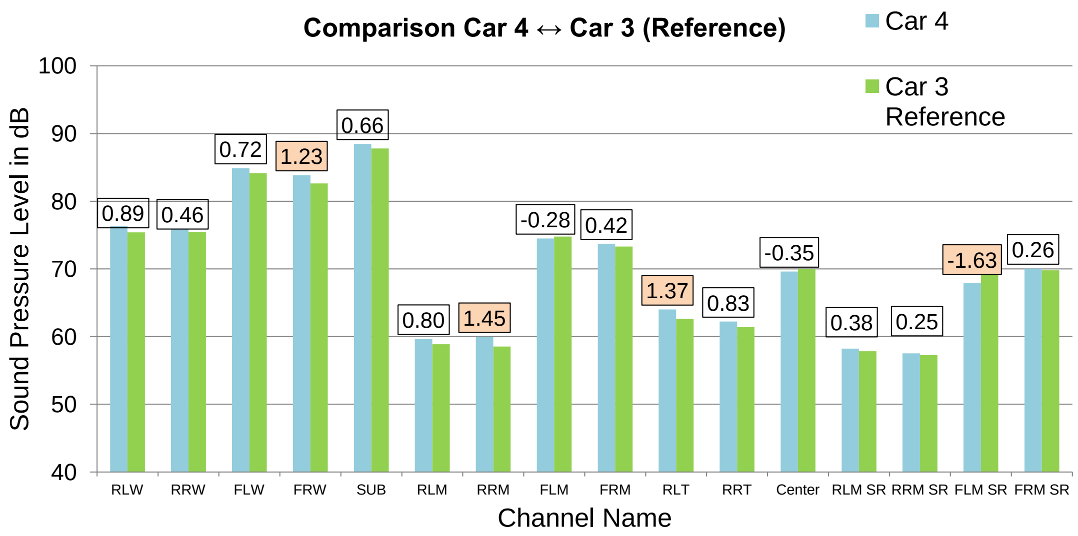


Figure 3.23: Level differences in dB between Car 4 and Car 3 (Reference)

From these EOL simulation results, we can draw the conclusion that with all tolerances added up at the end of the signal chain, we can expect level differences between cars in the order of 1-2 dB on average but with the possibility of mismatches greater than that. As even one channel disturbance alone can have a negative effect on the complete system, the necessity of an EOL calibration system can be shown clearly.

3.4.2.2 Comparison of left- / right-hand drive vehicles

When EOL testing, both left- and right-hand drive vehicles have to be considered. In both vehicle types, the same hardware is used but EQ files and tuning processes are different due to different car cabin geometry and the steering wheel position on the other side. In theory, EQ settings are estimated to be mirrored and thus, the sound pressure levels for all channels are expected to behave in the same way. In order to investigate the circumstances in reality, a right-hand drive vehicle has been taken for running the test and comparing its channel-wise sound pressure levels to the reference car. It is not surprising that level differences turned out to be a lot higher since the golden reference is a left-hand drive car. This can be seen in the middle column on the table in figure 3.24. For further examination of the idea that EQ and channel settings are mirrored, channel levels of Car 5 have been switched as if the car would have a left-hand drive implementation. For example, the front left woofer of Car 5 is directly compared to the front right woofer of Car 3 and so forth.

When having a closer look at the numerical results, the switched version seems to have lower level differences and slightly less orange and red colored cells. But when checking the average level differences of 1.32 dB and 1.30 dB, it turns out that both versions score quite badly in comparison to the table in figure 3.20 with the average level differences much lower.

The overall results of this test lead to the confirmed conclusion that left-hand and right-hand drive vehicles cannot be compared. Even if the EQ files are theoretically mirrored versions for both implementation types, the reference car for right-hand drive vehicles has to be a right-hand drive car as well. In this way, a valid golden reference can be established to obtain correct outcomes.

Level differences in dB

	Original	Switched
Channel	Car 5 ↔ 3	Car 5 ↔ 3
RLW	1.00	0.46
RRW	0.42	0.97
FLW	-0.14	-5.77
FRW	-4.24	1.39
SUB	0.25	0.25
RLM	1.92	1.73
RRM	2.08	2.27
FLM	-1.29	-0.85
FRM	0.63	0.19
RLT	-1.35	-0.09
RRT	1.15	-0.11
FLT	no signal	no signal
FRT	no signal	no signal
Center	-0.08	-0.08
RLM SR	3.31	1.53
RRM SR	2.08	3.87
FLM SR	1.11	0.39
FRM SR	0.12	0.85
Average	**1.32**	**1.30**

Figure 3.24: Level differences in dB of Car 5 in reference to Car 3

3.4.2.3 Summary and conclusion

After obtaining more knowledge and information about the tolerances in numerical values, the necessity of an EOL test concept in the acoustical domain has been proven. But even if the overall test results are clear to be interpreted, they have to be taken with caution.

In the beginning, all cars had to undergo an initial check in order to make sure that test conditions are approximately the same in every vehicle. This initial check includes hardware settings such as the exact adjustment of the microphone position, putting the seats in an easily repeatable position, closing all doors and the trunk and keeping equal settings on the sound card.

On software side, sound settings on the HMI such as bass or treble boost have to be all set to zero and the volume has to be adjusted to a predefined value. However, even if this initial check has been conducted in every vehicle, the test conditions were never exactly the same.

Altogether, there are many influential factors we have to consider such as outside noise and many other different aspects that play an important role when implementing an EOL test tool in real practice. In the next chapter, we want to focus on a working solution for an end-of-line test concept, including a detailed overview of a possible implementation strategy and the challenges on the way towards the realization of a stable concept.

4 End-of-line system calibration in practice

The overall goal of a sound system calibration at the end of the production line is to ensure that the customer experiences the sign-off performance of the audio system in every vehicle being purchased. The system should recognize faults in terms of unacceptably high tolerances or other kinds of damages and compensate deficiencies in a reasonable and moderate way. This chapter introduces a concept to implement a calibration test tool in practice and sheds light on various challenges which have to be faced in order to establish a stable working solution. Finally, solutions for the improvement of the current system are suggested.

4.1 Implementation concept

As being confirmed in chapter 3.4, the EOL test concept only focuses on the acoustical testing. The idea is similar to the test process for the acoustical loop explained in chapter 3.3.3.

A microphone in the car cabin records the test signal which is played back by all channels consecutively. In contrast to the simulation study in chapter 3, the test signal is fed into the system via a specific diagnostics software. This software is connected to the car and acts like an interface between the computer and the vehicle in order to get access to all properties of the car. This affects not only the sound system characteristics but also seat and steering wheel positions, engine properties and all other components with electrical control units. Vehicle data cannot only be observed but also modified.

4.1.1 Measurement

In our case, the diagnostics tool can mute and solo speakers separately and play back predefined test signals such as a sine wave or pink noise. The signal is picked up by a microphone in the car cabin und directly fed back to the amplifier. Inside the amplifier, the amplitude and root mean square of the signal is calculated and defined as the level of the currently active channel. Next, the measured level is compared to a channel-specific value in a lookup table which is stored internally in the amplifier. This lookup table is retrieved from the sound level measurement in the golden reference vehicle and contains all the desired levels for every single channel. The golden vehicle should be the vehicle where the tuning process and the equalization took place in order to make sure that the golden version is exactly the sign-off performance we want to have in every customer car.

4.1.2 Compensation

After comparison of the actual values with the desired levels inside the table, gains should then be updated in the next step. The software inside the amplifier takes the level differences and adds or subtracts x.x dB to or from the existing channel gain in order to compensate the level differences. Figure 4.1 illustrates the signal flow of the test concept. The diagnostics tool will be an external device which has to be put actively into every car to run the calibration.

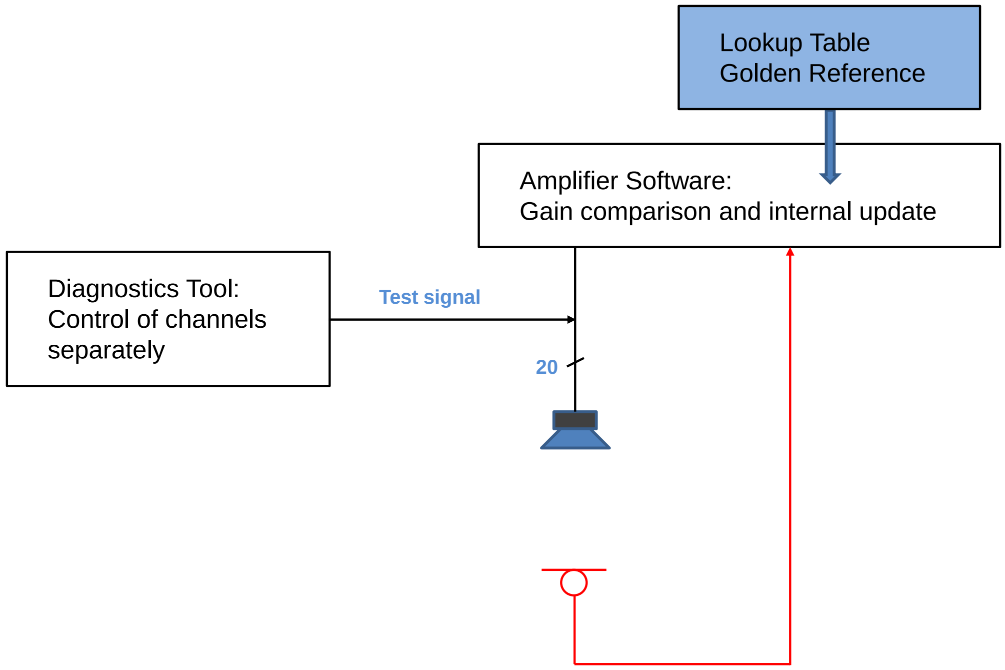


Figure 4.1: Signal flow chart for the EOL test concept with diagnostics

Of course it is not a surprising fact, that there is a variety of challenges and factors which can influence the measurement and calibration in a negative way. In the following chapter, upcoming difficulties are explained in more detail.

4.2 Challenges

4.2.1 Limited resources for development

As already mentioned in the previous chapters, the test tool is specifically developed for vehicles in the luxury car brand sector. In comparison to automotive firms addressing the broader market segment, small luxury companies usually have a shortage of vehicles to run tests with, especially those with the advanced sound system implemented which has to be tested. In most of the cases, only pre-series vehicles are available for testing purposes which are often affected by unknown faults or show differences to the standard equipment in the future car to be released. The sound system can also be affected and thus, been operated far away from the final SOP (start of production) condition and far away from the golden reference. As a consequence, the EOL tool intended for customer cars is often tested on pre-series vehicles which is a common source of errors and wrong results.

4.2.2 Microphone choice

Considering the choice of the testing microphone in the car cabin, an exterior or an interior microphone can be chosen, each with advantages and disadvantages to be discussed.

<u>4.2.2.1 Interior microphone</u>

In contrast to the simulation in chapter 3 where an external microphone is used, an already implemented microphone might be the better choice in many aspects. Nowadays, most vehicles are equipped with already implemented microphones on the roof for active noise cancellation applications, also called ANC. These microphones pick up the interior noise in the car cabin and combined with additional accelerometers in the chassis, the data is sent to a signal processing unit which produces sound that is 180 degrees out of phase to cancel out the primary noise source. These microphone signals can also be used for the EOL testing which would have two essential advantages. First, no external device has to be brought into the car which results in a time-effective test design. Furthermore, the microphone position is standardized in every vehicle, so errors due to uncertainties in microphone positioning are avoided. The downside, however, is the limited quality of this microphone and its bandwidth limitation. Additionally, these microphones often have their own tolerances. This is an important issue since a different microphone is used for every calibration process. In figure 4.2, the allowed tolerance range is shown for the implemented microphone. Except for a frequency of 1 kHz, the magnitude has a tolerance window of 6 dB, or +/- 3 dB for the overall frequency range and it is obvious that deviations of that size influence the measurement results significantly. There are no requirements for frequencies below 60 Hz and above 4 kHz.

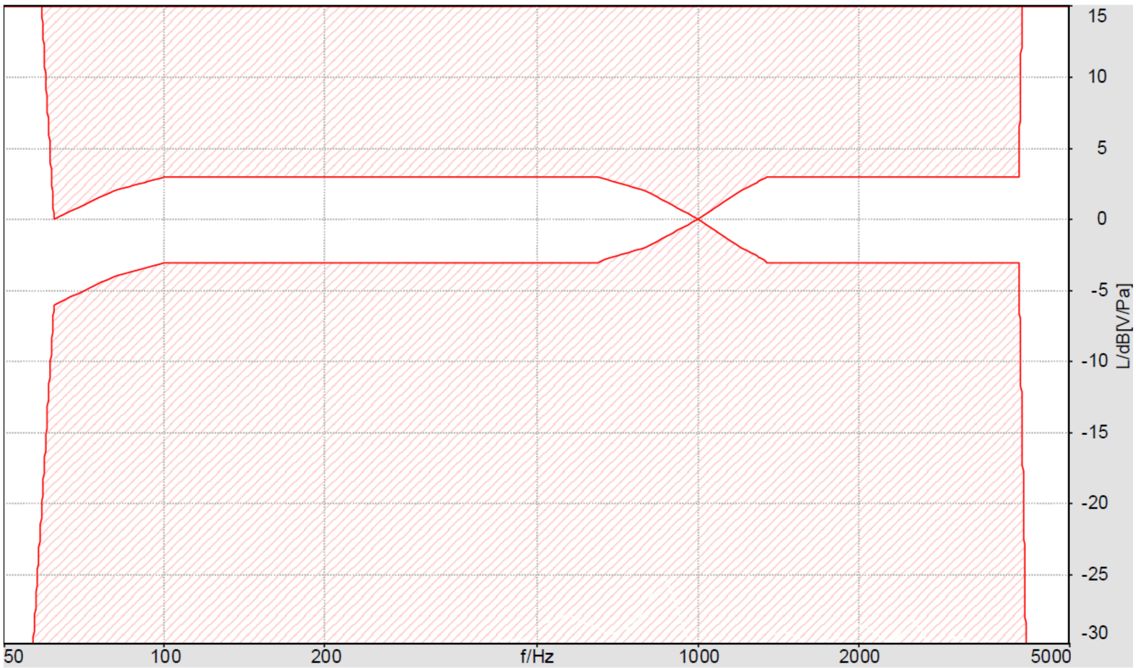


Figure 4.2: Tolerance band for implemented ANC microphones [GAH12]

Out of arising concerns, suggestions have already been made to use better microphones with a higher quality for the ANC application, but this is still in progress.

4.2.2.2 Exterior microphone

In order to overcome the quality and bandwidth limitation, an external microphone can be put into the car at a predefined position as it was carried out in the simulation in chapter 3. A high-quality device can be used with a flat frequency response and a bandwidth which also includes subwoofers and the high frequency range. Furthermore, tolerances will be lower and all sign-off cars are tested on the line with the same device. The disadvantage of the exterior choice is the negative time aspect and the risk of inaccurate positioning.

In order to find out more about the effects of microphone placement inaccuracy, the test microphone from chapter 3 has been mounted on two positions 10 cm above and below the reference position. In this test, the front left mid speaker was active, playing pink noise. Figure 4.3 shows the influence of the microphone position on the measured sound pressure level in a bar chart. Incorrect positioning in the order of 10 cm vertically results in an average level mismatch of 0.26 dB.

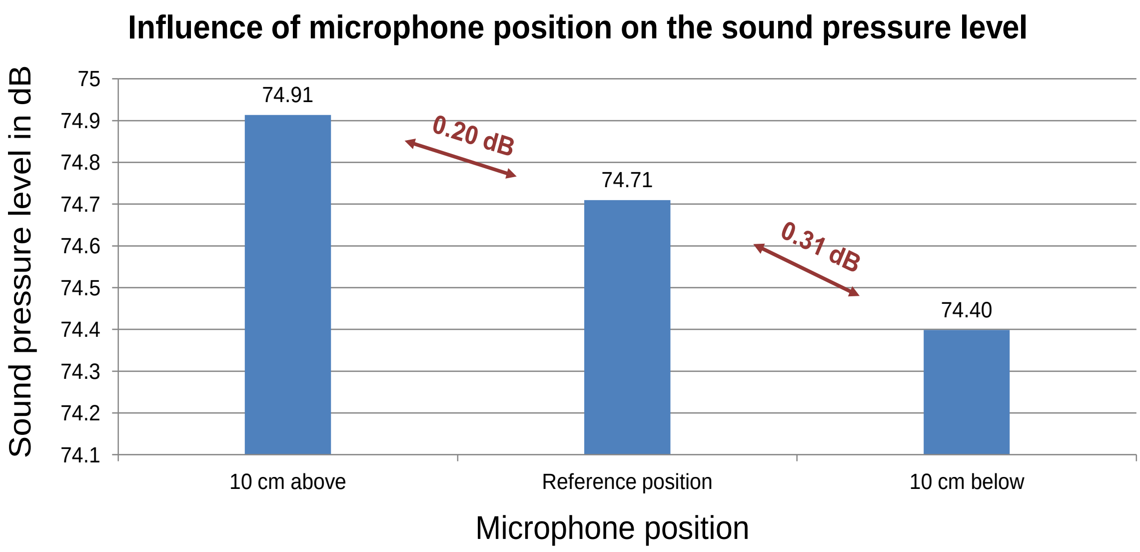


Figure 4.3: Influence of microphone position on sound pressure level

4.2.3 Car interior

Another major influential factor is the equipment of the car interior considering various aspects.

4.2.3.1 Seat positions

Seat positions can have a big influence on the measurement results considering wave propagation and reflections on obstacles. Since seat positions on the line are random, every vehicle is tested under different conditions. The term seat position refers to the seat base position, back and head rest. In order to establish a valid calibration result, seat positions have to be the same for every calibration. The diagnostics tool can get access to the seat properties and adjust the positions to three different presets: fully fore, fully aft and a mid position. Overall, the mid position has been agreed to be the reference for the measurements since it is the most likely one in daily usage. The challenge here is to combine seat position adjustment and the calibration within one process in the diagnostics tool.

A similar situation can be observed for the position of the steering wheel additionally.

4.2.3.2 Windows and doors

During the calibration process, it has to be ensured that all doors, windows, sun roof and the trunk are closed properly. Again, the information about doors and windows open or closed can be retrieved from the diagnostics tool, hence the communication with the tool is essential in this case as well.

4.2.3.3 Vehicle equipment and optional choices

Since there are various equipment options for the car interior to be chosen by the customer, all vehicles to be tested can be very different from each other even if it is the same model. The most important aspect in this case is the interior cover material of the seats, connecting elements and the roof liner. The type of fabric, leather or cloth have different absorption coefficients and thus, a different absorption behavior in the sound field. This mostly affects higher frequencies. Taking the reference frequency of 1 kHz, the absorption coefficient for cloth-upholstered seats is about 45% higher than for seats covered with leather [ACO]. As there can be seen in figure 4.4, high frequencies are increasingly damped in case of cloth in comparison to leather. Therefore, those two cases have to be distinguished in vehicles.

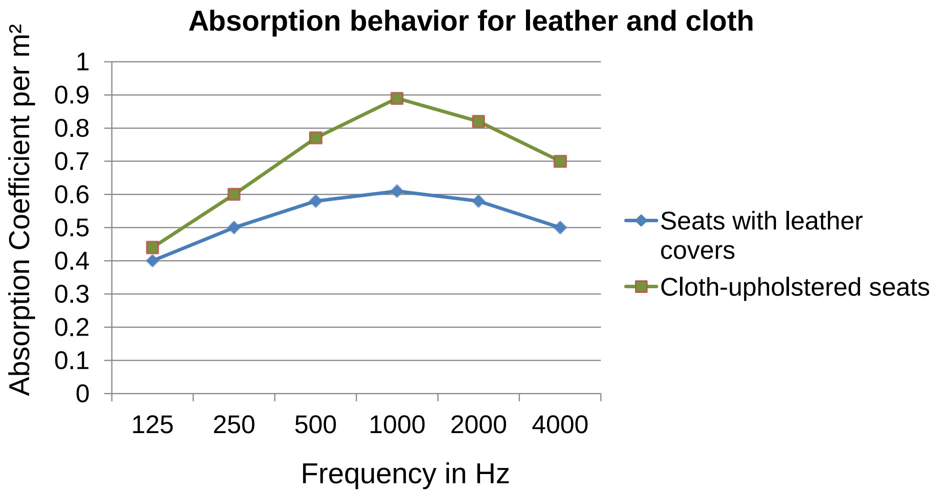


Figure 4.4: Absorption behavior for leather and cloth [cf. ACO]

Apart from clothing, other equipment can have an additional impact on the measurements, for example optional glassy sun roofs and sun shades or curtains on the windows. Furthermore, the vehicles on the line often have protective coverings in and outside the car to avoid scratches and other kinds of damages on sensitive surface structures. These covers and other possible items on the seats or in the trunk can modify measurements. Before running the test, each car should be inspected manually for such items and coverings. It is also a challenging task to do this reliably in a fast-paced environment with employees who possibly do not have extended knowledge about the sensitivity of sound system calibration.

4.2.4 Measurement environment at the end of the line

Speaking of the line itself, it does not seem surprising that the measurement environment is challenging on its own.

4.2.4.1 Time limit

First of all, time is scarce and the calibration process has to be performed as fast as possible. Especially when it comes to manual checks under pressure, this can lead to human failure and measurement errors.

4.2.4.2 Outside noise

Furthermore, a serious problem is the outside noise level. In order to obtain reliable measurement results, the test should be conducted in an anechoic chamber. Of course, this is not the case at the end of the line, but a relatively low outside noise level is essential for getting valid results. It is not impossible to achieve a noise level of 45 dB SPL inside the car with all doors closed. Using a test signal sufficiently loud will result in a practically usable signal-to-noise ratio.

However, we should not forget that in a work environment like this, many workers do their tasks on the vehicles at the same time which can, depending on the task, lead to disturbing noise such as hammering, drilling or knocking. These are also random processes and not predictable. Additionally, with workers being around, one cannot be sure that doors are opened during calibration processes. Therefore, EOL testing should be carried out in an extra room, separated from all other tasks.

4.2.4.3 Environment

There is one part of the signal flow that has not been considered so far: the transmission function from the loudspeaker to the listener's ear, or the microphone, respectively. Following the rules of wave propagation, the transmission of sound waves is affected by the medium in which the wave is traveling. For temperature changes of +/- 20°C, the SPL will be influenced by +/- 0.6 dB. For an operating range of -40°C to 90°C, the variation from 25°C will approximately be +/- 2 dB. Also, static air pressure directly influences the transmitted SPL. A pressure change of +/- 20 hPa will result in a SPL mismatch of +/- 0.2 dB. Furthermore, humidity is another aspect to be considered and can have an impact on absorption characteristics of materials [CHA16].

Finally, these properties of the transmission medium do not only affect the sound field itself, but also the behavior of the microphone, for example the temperature-induced change of microphone sensitivity. In extreme work environments, measurements can be distorted due to extraordinary environmental conditions. This supports the suggestion to provide a separated room which has to meet specific requirements for EOL testing.

4.2.5 Measurement uncertainties

In testing environments, measurement tools can have tolerances themselves. For a valid calibration, these tolerances should be evaluated as precisely as possible to know about the uncertainty of the obtained results.

Furthermore, the golden reference sign-off car itself has speakers that have already been used for several hours, which is not the case in brand new cars. Therefore, studies have to be carried out investigating the behavior of new speakers in comparison to the same speaker after a predefined number of operating hours under normal conditions. It is commonly recommended that the sound system has already been operated before running the calibration process. Overall, further investigations are important to obtain numerical results and more precise relations between operating hours and speaker performance.

4.3 Suggestions for a working solution

After addressing various upcoming challenges to be faced, solutions are suggested how to provide a stable system and how to improve the measurement conditions.

4.3.1 Adjustment window size

With the knowledge about measurement uncertainties, an adjustment window is introduced in which adjustment will not be made. The window size will be directly related to the measurement uncertainty. For example, if the uncertainty lies around +/- 0.5 dB, the system will not adjust level differences within this range. By applying this strategy, it can be achieved that only true errors are corrected.

As mentioned in the chapter about measurement uncertainties on page 60, the speakers have to undergo additional testing procedures to find out more about the influence of operating hours on the overall performance.

4.3.2 Suitable location for calibration

In chapter 4.2.4, the suggestion has already been made to provide a quiet room next to the line to establish a suitable measurement environment for calibration purposes. In general, it is a fundamental question, where the calibration should take place. Up to now, we assumed to carry out the calibration at the end of the line inside the plant of the particular car manufacturer. Premises on-site involve difficulties in terms of limited available space and outside noise. Therefore, it has been proposed to outsource the process to external partners such as authorized workshops or dealers before handing over the vehicle to the customer.

The advantage of this strategy is to provide both more time and a more adequate location to run the calibration. But on the downside, if the sound system has more serious damages than just tolerances to be compensated, the workshop or dealer does not have the possibility to repair the faulty parts. At worst, the vehicle has to be brought back to the manufacturer. One idea is to split the calibration into two parts: the detection of damages on-site before the car leaves the manufacturer and the actual calibration of tolerances carried out by the external partner. Additionally, it has to be considered that dealers or workshop staff need specific trainings to be able to calibrate sound systems.

4.3.3 Communication between vehicle and calibration tool

In order to get valid measurement data to run the calibration, we need to make sure that the obtained data is plausible and valid. Therefore, the tool has to be able to run a few initial checks before starting measurement and calibration. Communication between the vehicle and the calibration tool is therefore an essential aspect in the process in order check if the current conditions allow a valid measurement. If requirements are not met within predefined limits, the process should be cancelled.

This initial test should be able to check if all doors and windows are closed and if seats are in the correct position. If not, the seat position can be adjusted via the diagnostics tool. To get access to external parameters such as temperature, additional sensors have to be brought into the car. Another important condition is the background noise which can be recorded by the calibration microphone. If the outside noise level is above a certain limit, the calibration is stopped.

Additionally, it is recommended to implement the option to cancel the calibration manually and to revert the status to factory settings.

5 Conclusion

With all challenges to be faced on the way towards the realization of an EOL calibration for sound systems in automotive audio, we have to keep one important fact in mind that we should not create solutions to problems which do not exist. If the calibration is unstable and the measurements are not exact enough, a sound system with slight deficiencies can actually become much worse after calibration. Before launching the calibration tool, all critical aspects have to be eliminated to ensure a stable outcome and the successful improvement of the sound system.

In this thesis, tolerances and level differences between devices have been investigated with the outcome that calibration on an electrical level only is not necessary since the level differences between amplifiers have been determined to be negligibly small. On the contrary, however, it was possible to show clearly the necessity of an EOL testing in the acoustical domain since level differences between test vehicles and the reference car exceeded values of 1 or even 2 dB; in one case, a mismatch of 6 dB has been discovered. In order to ensure the reliable delivery of sound systems whose performance matches the golden sign-off vehicle as closely as possible, values in this order of magnitude are not acceptable and have to be compensated by a channel-wise gain update inside the amplifier. For testing purposes in the early stage of development, a Matlab tool was developed which allows simultaneous audio playback and recording and which establishes an easy-to-use measurement process to access sound system properties such as frequency response and sound pressure levels of each channel. For future aspects, suggestions have been made to improve and accelerate the process to create a fully functional and reliable end-of-line test concept in practice.

In the next months, a decision has to be made which microphone should be used and if the existing ANC microphone will be replaced by a device with higher quality and extended bandwidth. If this is done and all equipment is in place, the measurement uncertainty has to be examined to define a suitable adjustment window size for calibration. On software side, the diagnostics tool has to be embedded successfully into the signal flow and internal processes in the amplifier have to be programmed to work reliably, including all conditions from outside.

As an outlook to the future, EOL calibration is a potential instrument to ensure a perfect performance for every sound system. Once established a working solution, all car manufacturers will have access to the possibilities to deliver perfect copies of the golden reference sign-off vehicle. Furthermore, the application can also be extended on the market to be available not only in the automotive sector but also for all audio appliances and installations. All in all, this strategy promises a successful continuous improvement of acoustic experience in the field of high-quality audio.

References

[ACO] Absorption Coefficients
URL:
http://www.acoustic.ua/st/web_absorption_data_eng.pdf
(last access on June 22, 2016)

[AP] User´s manual Audio Precision APx500

[BAY11] Bayer, D.: “Loudspeaker Reliability Testing”, company internal document, 2011

[CHA14] Chapman, P. J.: "Quantifying Acoustic Measurement Tolerances and their Importance in the Loudspeaker Supply Chain", presented on the 136th AES convention, Berlin, April 2014

[CHA16] Chapman, P.J.: "Electroacoustical Aspects of EOL Calibration", company internal document, March 2016

[DIRa] Dirac research signal processing, Solutions for Automotive Audio, Sweden
URL:
http://www.dirac.com/automotive-audio/
(last access on June 22, 2016)

[DIRb] Dirac research signal processing, Online Store, The Dirac Live Room Correction Suite
URL:
http://www.dirac.com/online-store/
(last access on June 22, 2016)

[FOL14] Foley, D.: "White Noise Definition Vs. Pink Noise", article issued by Acoustic Fields, Room Acoustics Design Specialists, 2014
URL:
http://www.acousticfields.com/white-noise-definition-vs-pink-noise/
(last access on June 22, 2016)

[GAH12] Gahner, E., Perl, M.: "Technische Entwicklung, Konzernlastenheft Mikrofone", company internal document, 2012

[HAG15] Hagmüller, M.: "Active Noise Control", lecture slides for "Digitale Audiotechnik 2", Signal Processing and Speech Communication Laboratory, TU Graz, 2015

[HUM14] Humphrey, R.: Playrec Multi-channel Matlab Audio, 2008 - 2014
URL:
http://www.playrec.co.uk/compiling.html
(last access on June 22, 2016)

[INT16] Company intranet, April 2016

[KAI13] Kaindl, F.: “Sound Internal Audio Amplifier: Functional Requirements, Style Guide”, company internal document, 2013

[KLI06] Klippel, W., Schlechter, J.: “Measurement and Visualization of Loudspeaker Cone Vibration“, presented on the 121th AES Convention, San Francisco, October 2006
URL:
https://www.klippel.de/fileadmin/klippel/Files/Know_How/Literature/Papers/Measurement_and_Visualization_Cone_vibration_06.pdf
(last access on June 22, 2016)

[KLI14] Klippel, W.: “Audio Systems Modelling Measurement Control”, lecture slides for the course “Sound Quality of Audio Systems”, Institute of Acoustics and Speech Communication, University of Technology, Dresden, 2014

[KLIa] User´s Manual Klippel R&D System

[KLIb] “R&D System – The Development Tool of Audio Technology“ Specification of the measurement modules of the Klippel R&D System
URL:
https://www.klippel.de/our-products/rd-system.html
(last access on June 22, 2016)

[KLIc] Sketches Setup R&D System by Klippel
URL:
https://www.klippel.de/our-products/rd-system/hardware/sketches-rd-setup.html
(last access on June 22, 2016)

[LAV] URL:
http://www.lavocespeakers.com/design
(last access on June 22, 2016)

[OLI15] Olid Dominguez, B.: “Infotainment”, presentation slides, company internal document, 2015

[SCH] Schlechter, J., Klippel W.: “Visualization and Analysis of Loudspeaker Vibrations”, Klippel GmbH, Dresden, n.d.
URL:
https://www.klippel.de/fileadmin/_migrated/content_uploads/Visualization_and_Analysis_of_Loudspeaker_Vibration_01.pdf
(last access on June 22, 2016)

[SHI13] Shinn-Cunningham, B.: "Auditory Precedence Effect", Encyclopedia of Computational Neuroscience, Springer Science + Business Media New York, 2013
URL:
http://www.cns.bu.edu/~shinn/resources/pdfs/2013/2013_Precedence.pdf
(last access on June 22, 2016)

[SWE00] "Pink Noise Versus White Noise", article issued by Sweetwater, Music Instruments and Pro Audio, August 2000
URL:
http://www.sweetwater.com/insync/pink-noise-versus-white-noise/
(last access on June 22, 2016)

[SYS14] "Systemidentifikation - Schätzung des Frequenzganges aus den Leistungsdichtespektren", lecture slides for "Schwingungs-messtechnik Vorlesung", Institute of Electrical Measurement and Measurement Signal Processing, TU Graz, 2014

[TRU99] Truax, B.: “Handbook for Acoustic Ecology”, 2nd Edition, 1999
URL:
http://www.sfu.ca/sonic-studio/handbook/Phon.html
(last access on June 22, 2016)

[ZEL12] Zeller, P.: "Handbuch Fahrzeugakustik. Grundlagen, Auslegung, Berechnung, Versuch", Kapitel 9, Vieweg + Teubner, 2. Auflage, 2012

[ZHI12] Zhijun Zaho, L.X.: "The Influence of Visual Stimuli on Just Noticeable Difference of Loudness", 5th International Congress on Image and Signal Processing, Chongqing, 2012

Annex

A AudioLoopGainCheck.m

```
clear all;
close all;
clc;
addpath('PlayrecForMatlab');
addpath('playrec_testmatlab');
%% *** bernadette Schreyer AudioLoopGainCheck May 2016 ***
% Multichannel I/O Loop
%% Test audio I/O device ID
% Required: Fireface 800 or Tascam US-122MKII
fs = 44100;
disp('Audio Device is being checked, read struct and look for correct
Device ID. Change it if necessary');
devices = playrec('getDevices');
device_ID = 6; % Select required sound card driver in the struct
disp(strcat(devices(1,device_ID+1).name));
disp('Set volume on HMI to 15');

%% Audio Loop Playback and Record: Set input and output channels (only
1 channel active in this setup, max 16 channels under test)
fs = 44100;
chmin = 1; % Min Channel
chmax = 1; % Max Channel
channel = chmin:1:chmax;
input_CH = [1];
output_CH = [1]; % output_CH = [channel];

%% Measurement of Frequency Response with a Sweep
filename_play_s = 'Sweep.wav';
filename_rec_s = 'RecordedSweep.wav';

rec_data_s = playrecord_wav(filename_play_s, device_ID, device_ID,
output_CH, input_CH, fs);
% figure; plot(rec_data_s); ylim([-1.0 1.0]); grid on;
% str ['Recorded Signal Channel ' num2str(i)];
% xlabel('Samples'); ylabel('Amplitude'); title(str);
audiowrite(filename_rec_s, rec_data_s, fs,'BitsPerSample',16);
data_s = rec_data_s;
testsignal_s = audioread(filename_play_s);

%% Measurement of RMS with Pink Noise
filename_play_n = 'PinkNoise_Stereo_5s_corr_norm.wav';
filename_rec_n = 'RecordedNoise.wav';

rec_data_n = playrecord_wav(filename_play_n, device_ID, device_ID,
output_CH, input_CH, fs);
audiowrite(filename_rec_n, rec_data_n, fs,'BitsPerSample',16);
data_n = rec_data_n;
```

```
testsignal_n = audioread(filename_play_n);


%% DATA ANALYSIS

for i = chmin:1:chmax

% RMS and Gain Calculation
output_n = data_n(:,i);
input_n = testsignal_n;
output_n_cut = output_n(22050:length(output_n)-22050); % windowing:
cut first and last 0.5 second
rmsdata = sqrt(mean(output_n_cut.^2));
rmsredline = rmsdata*ones(1,length(output_n_cut));
str = ['Windowed Signal and RMS for Channel ' num2str(i)];
figure; plot(output_n_cut); grid on; hold on;
plot(rmsredline,'r');title(str);

refvector =
[0.013330,0.013382,0.036482,0.030581,0.055554,0.001908,0.012393,0.0104
53,0.003059,0.002655,0.007125,0.006986,0.001763,0.001654]; % channel-
wise RMS of Reference Car 3
% calculating required gain to comply with reference
gain = -20*log(rmsdata/refvector(i));
str1 = ['RMS for Channel ' num2str(i) ':'];
str2 = ['Gain for Channel ' num2str(i) ' in dB:'];
disp(str1); x = sprintf('%0.6f',rmsdata); disp(x);
disp(str2); disp(gain);

% Gain limit for adjustment window size and creating vectors for
excelport

if abs(gain) < 1          % definition of adjustment window size: here
+/- 1 dB
    gainvector(i) = 0;    % no gain update
else
gainvector(i) = gain;     % vector with required channel gains for
updating
end

channelvector(i)= i;      % channel number
rmsvector(i) = rmsdata;   % vector with measured RMS values for every
channel


% Frequency responses
output_s = data_s(:,i)
input_s = testsignal_s;

figure;spectrogram(output_s,256,128,256,44100,'yaxis');
title('Spectrogram Output');
figure;spectrogram(input_s,256,128,256,44100,'yaxis');
title('Spectrogram Input');
% figure; stem(data); title('Stem data');
% figure; stem(testsignal); title('Stem testsignal');
nges = length(output_s) + length(input_s) - 1;
fr = fft(output_s,nges)./fft(input_s,nges);
acoeffs = ones(1,100)/100;
```

```
smoothfr = filter(acoeffs,1,fr); % smoothing
len = length(fr);
fax = 0:fs/len:fs-fs/len;
magnitude = 20*log(abs(smoothfr));
scale = 20*log(abs(smoothfr(1000))); % scale to 0dB (reference 1 kHz)
magnitude = magnitude - scale;
magmatrix(:,i) = magnitude;
% phase = (unwrap(angle(smoothfr))*(180/pi);
str4 = ['Frequency Response of Channel ' num2str(i)];
str5 = ['CH ' num2str(i)];
figure; grid on;
semilogx(fax,magnitude); grid on; xlabel('Frequency');
ylabel('Magnitude in dB');
xlim([20 22050]); ylim([-100 200]); title(str4); legend(str5);

% % subplot(2,1,1); semilogx(fax,magnitude); grid on;
xlabel('Frequency'); ylabel('Magnitude in dB');
% % subplot(2,1,2); semilogx(fax,phase); grid on; xlabel('Frequency in
Hz'); ylabel('Phase in Degrees');

end

%% Visualizing Results
% Bar Chart
str6 = ['Required Gains in dB for Channel ' num2str(chmin) ' to '
num2str(chmax)];
figure; bar(channel,gainvector,0.1); grid on;

for i = chmin:1:chmax
text(channel(i),double(gainvector(i)+10),num2str(gainvector(i)),'Horiz
ontalAlignment','center','FontSize',14);
end
ylabel('Gain in dB'); xlabel('Channel Number'); title(str6);

%% Creating Excelport
filename = 'listofgains.xlsx';
% colums
column1 = {'ChannelNr.'};
xlswrite(filename, column1,1,'B2');
column2 = {'Reference'};
xlswrite(filename, column2,1,'C2');
column3 = {'Measured'};
xlswrite(filename, column3,1,'D2');
column4 = {'RequiredGain'};
xlswrite(filename, column4,1,'E2');
% data
xlswrite(filename,channelvector',1,'B3');
xlswrite(filename,refvector',1,'C3');
xlswrite(filename,rmsvector',1,'D3');
xlswrite(filename,gainvector',1,'E3');
```

B playrecord_wav.m [cf. HUM14]

```
function [ recData ] = playrecord_wav( fileName_play,
playDeviceID,recDeviceID, playChanList,recChanList, Fs, startPoint,
endPoint)

% fileName_play: name of the file to be played
% playDeviceID: audio device ID for playback
% recDeviceID: audio device ID for recording
% playChanList: channel number for playback
% recChanList: channel number for recording
% Fs: sampling frequency

%PLAY_WAV Play a wav file
%
%play_wav( fileName, playDeviceID, chanList, startPoint, endPoint )
%   Plays the file fileName on Playrec device playDeviceID.  chanList
%   specifices the channels on which speakers are connected and must
be a
%   row vector.  startPoint and endPoint are optional, and can be used
to
%   limit playback to a particular range of samples.

pageSize = 4096;    %size of each page processed
pageBufCount = 5;   %number of pages of buffering

runMaxSpeed = false; %When true, the processor is used much more
heavily
                     %(ie always at maximum), but the chance of
skipping is
                     %reduced


[fileSize] = audioinfo(fileName_play);
Fs = fileSize.SampleRate;
fileLength = fileSize.TotalSamples;
fileChanCount = fileSize.NumChannels;

recBuffer = [];
recData = [];

if((fileChanCount < 1) || (fileChanCount > 2))
    error ('File must contain either 1 or 2 channels');
end

if(nargin<7)
    startPoint = 1;
else
    startPoint = max(1, startPoint);
end

if(nargin<8)
    endPoint = fileLength;
else
    endPoint = min(endPoint, fileLength);
```

```
end

if(startPoint > endPoint)
    fprintf('(startPoint > endPoint) so no samples to play\n');
    return
end

if ~isreal(playChanList) || length(playChanList) < 1 ||
length(playChanList) > 2 ...
    || ndims(playChanList)~=2 || size(playChanList, 1)~=1

    error ('chanList must be a real row vector with 1 or 2 elements');
end

%Test if current initialisation is ok
if(playrec('isInitialised'))
    if(playrec('getSampleRate')~=Fs)
        fprintf('Changing playrec sample rate from %d to %d\n',
playrec('getSampleRate'), Fs);
        playrec('reset');
    elseif(playrec('getPlayDevice')~=playDeviceID)
        fprintf('Changing playrec play device from %d to %d\n',
playrec('getPlayDevice'), playDeviceID);
        playrec('reset');
%     elseif(playrec('getPlayMaxChannel')<max(chanList))
%         fprintf('Resetting playrec to configure device to use more
output channels\n');
%         playrec('reset');
    end
end


%Initialise if not initialised
if(~playrec('isInitialised'))
    fprintf('Initialising playrec to use sample rate: %d,
playDeviceID: %d and no record device\n', Fs, playDeviceID);
    %playrec('init', Fs, playDeviceID, -1);
    playrec('init', Fs, playDeviceID, recDeviceID);

    % This slight delay is included because if a dialog box pops up
during
    % initialisation (eg MOTU telling you there are no MOTU devices
    % attached) then without the delay Ctrl+C to stop playback
sometimes
    % doesn't work.
    pause(0.1);
end

if(~playrec('isInitialised'))
    error ('Unable to initialise playrec correctly');
elseif(playrec('getPlayMaxChannel')<max(playChanList))
    error ('Selected device does not support %d output channels\n',
max(chanList));
end

if(playrec('pause'))
    fprintf('Playrec was paused - clearing all previous pages and
unpausing.\n');
```

```
    playrec('delPage');
    playrec('pause', 0);
end
pageNumList = [];

fprintf('Playing from sample %d to sample %d with a sample rate of %d
samples/sec\n', startPoint, endPoint, Fs);

for startSample = startPoint:pageSize:endPoint
    endSample = min(startSample + pageSize - 1, endPoint);

    y = audioread(fileName_play, [startSample endSample]);

    if length(playChanList) == 1 && fileChanCount == 2
        y = (y(:, 1) + y(:, 2)) / 2;
    end

    if length(playChanList) == 2 && fileChanCount == 1
        y = [y, y];
    end

    %pageNumList = [pageNumList playrec('playrec', y, chanList)];
    pageNumList = [pageNumList playrec('playrec', y, playChanList, -1,
recChanList)];


    if(startSample==startPoint)
        %This is the first time through so reset the skipped sample
count
        playrec('resetSkippedSampleCount');
    end


    % runMaxSpeed==true means a very tight while loop is entered until
the
    % page has completed whereas when runMaxSpeed==false the 'block'
    % command in playrec is used.  This repeatedly suspends the thread
    % until the page has completed, meaning the time between page
    % completing and the 'block' command returning can be much longer
than
    % that with the tight while loop
    if(length(pageNumList) > pageBufCount)
        if(runMaxSpeed)
            while(playrec('isFinished', pageNumList(1)) == 0)
            end
        else
            playrec('block', pageNumList(1));
        end
        recBuffer = [playrec('getRec', pageNumList(1))];
        %
        if size(recBuffer,1)>0
            recData = [recData;recBuffer];
        end;
        %
        pageNumList = pageNumList(2:end);
    end

end
```

```
fprintf('Playback complete with %d samples worth of glitches\n',
playrec('getSkippedSampleCount'));
```