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Impact of Automotive Glass Subwoofer Technology on Vehicle Interior Sound

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ABSTRACT

An innovated design for the automotive subwoofer system is proposed where the rear glass functions as the dynamic driver of the subwoofer system. The rear glass is mechanically excited using two piezoelectric actuators located along the bottom edge. The glass is fixed along the top and is free to move along the other three sides. The actuators exert a force perpendicular to the glass surface which is proportional to the low frequency input signal taken from the audio system.

A study was undertaken to evaluate and compare the acoustic performance and characteristics of the rear glass subwoofer system relative to a conventional subwoofer system. Acoustical properties including frequency response, total harmonic distortion, and loudness are characterized and compared for both subwoofer designs. A subjective evaluation was conducted to correlate with objective measurements. An evaluation procedure suitable for evaluating the glass subwoofer system performance is recommended for future implementation.

DEDICATION

If it wasn't for my little angels,

That kept me awake day and night,

For their precious cry and laugh,

For: "Tata, let's go and play out",

This work would never be thrilling enough.

To my kids Stefan and Ana

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NOMENCLATURE

ADC	analog to digital converter
CA	charge amplified (accelerometer)
CPB	constant percentage bandwidth
dB	Decibel
FFT	Fast Fourier Transform
FM	frequency modulation
f_m	maximum frequency
f_n	Nyquist frequency
Hz	Hertz (cycles per second)
ICP	integrated circuit piezoelectric (accelerometer)
in	inch
lb	pound
m/s	meter per second
NVH	noise vibration and harshness
RMS	root mean square
RPM	reviews per minute
SNR	signal to noise ratio
SPL	sound pressure level
STP	standard temperature and pressure conditions
V	Volt
W	Watt
Ω	Ohms

CHAPTER I

INTRODUCTION

Along with modern technology and the competitive nature of today's automotive industry, the demand for sound quality has become paramount. The need for sound quality now influences both the strategy of auto-makers and the customer perception of the overall quality of the vehicle. Due to the complexity and versatility of a vehicle cabin, considering numerous noise and vibration sources, many challenges need to be overcome in order to refine the acoustic comfort of today's vehicles. While primarily focusing on the reduction of the overall interior noise and vibration within the vehicle cabin, NVH engineers have now recognized the importance of vehicle acoustical package and audio system. An accurate representation of a produced sound becomes an important factor of consideration for a more enjoyable listening experience and overall consumer appreciation.

In more recent years, much design effort is being directed towards the development and tuning of high-end automotive audio systems [1]. Various audio components including the radio head unit, separate amplifiers and premium loudspeakers are engineered to produce high quality level of audio performance.. Among others, the most significant component in premium audio systems, which greatly contributes to the overall listening experience, is the audible low frequency component of the subwoofer system.

Some background on the subject of different subwoofer systems is presented in the following text.

1.1 Conventional Subwoofer

A subwoofer is an electro-acoustic transducer which translates an electrical energy into sound and is dedicated for low audio frequencies or "bass". The first subwoofer was introduced in the 1960's in order to add the low frequency content to home stereo systems and to enhance the sound performance. It became popular in the 1970's with the introduction of "*Sensurround*". Later in the 1980's and 1990's, with the introduction of compact cassette and compact disc technology, reproduction of low frequency content was no longer limited by the capability of phonograph record stylus to track the groove [2]. This created a great opportunity for music producers to add more bass to the recordings, and there was an increase in the demand for subwoofers. By the beginning of the 21st century, subwoofers became increasingly popular in aftermarket car audio systems and almost a standard sound reinforcement component in nightclubs and concert venues.

Conventional subwoofers vary in size, weight, power consumption and frequency range. Based on product information from leading automotive audio system companies, automotive subwoofers are typically rated for a frequency range of 20 Hz to 100 Hz and some as high as 200 Hz. The reality of these manufacturer's specifications may vary depending on the vehicle cabin design and enclosure volume for where the subwoofer is installed which is usually the luggage compartment. Based on the experimental analysis conducted in this study, frequencies above approximately 110 Hz are not desired if the intent of the subwoofer is to reproduce accurate bass without sound leakage from the vocals and other higher-pitched frequencies. The diameter of the subwoofer can also significantly contribute to the capable frequency range of the speaker with a larger diameter speaker being capable of generating lower frequencies. For automotive applications, factory installed subwoofers usually range in size from 8 inches up to 10 inches with aftermarket automotive subwoofers being up to 15 inches in diameter. The packaging of large subwoofers can impose significant restrictions on the vehicle interior design, particularly for compact and medium size vehicles. With an increase in speaker size comes also an increase in weight. Automotive subwoofers can weigh up to 20 lbs which can attribute to increased fuel consumption of the vehicle. Large dynamic speakers may also consume relatively large amounts of electrical power requiring large external amplifiers. Given the significance of electric power consumption in electric and hybrid vehicles, the addition of such large subwoofer drivers to car audio systems can add additional electrical loads to such vehicles.

The conventional automotive subwoofer system is comprised of one or more dynamic drivers as illustrated in Figure 1.1 with the controlled excursion of the diaphragm, or cone, being the significant contributor to the quality of the acoustic output. However, as the driver moves outward during large excursions, the voice coil can often be extended out of the magnetic gap thereby causing a drop of magnetic force. This can have negative audio effects with less control of the voice coil which can cause the subwoofer to sound sloppy and introduce high levels of distortion into the acoustic output.

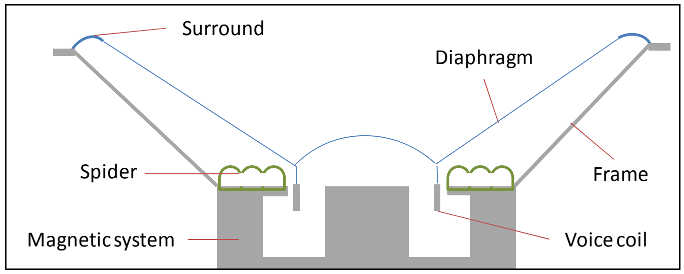


Figure 1.1 – Conventional Subwoofer Cross-section

It is due to the inherent disadvantages of the conventional subwoofer system which demonstrate the merits of an alternative subwoofer design. Specifically, a design with significant weight savings, lower power consumption and having less negative impact on vehicle fuel consumption is discussed. This alternative design uses the glass of the vehicle as the dynamic driver for the vehicle's subwoofer system.

1.2 Glass Subwoofer

The glass subwoofer system is an innovative approach to generating the audible low frequency sound in an automotive audio system. The operating concept is based on piezoelectric actuated exciters which serve as the dynamic driver for the windshield or rear glass of the vehicle. The use of piezoelectric actuators to excite a vehicle's glass to produce low frequency sound is novel. However, the technology of piezoelectric actuators has been used for the generation of high frequency sound for other applications. Due to the output amplitude limitations of the piezoelectric elements they are typically used in low cost high frequency applications such as electronic beepers as well as less expensive speaker systems including computer speakers and portable radio tweeters. Examples of patents involving piezoelectric technology applications are given in the literature survey section. Several patents for piezoelectric loud speakers for automotive applications are also introduced in the literature survey section, but none of them incorporate a glass panel of the vehicle as the sound source. It is also necessary to note that all of the proposed design solutions target either middle or high frequencies but not the bass frequencies.

For this investigation, two exciters are mounted along the bottom edge of the rear glass of a Chrysler 300 sedan as illustrated in Figure 1.2. The glass is fixed along the top edge and is free to move along the other three sides which are sealed using a special dynamic seal which allows for the movement of the window. The exciter is comprised of a piezoelectric element laterally compressed in a fishbone spring structure as shown in Figure 1.3. When an electrical signal is supplied to the piezoelectric element, it expands laterally and forces the spring system to push against the rails of the actuator. Since the base rail is mounted on the vehicle structure, the upper rail rises up and down, exerting an oscillating force perpendicular to the glass surface which is proportional to the low frequency input signal taken from the audio system. In order to drive the piezos the signal is amplified by a piezo amplifier. The careful design of the piezo amplifier is necessary for optimized actuator performance. Due to the relatively small size, the packaging of the piezoelectric exciters has little negative impact on vehicle cabin space. The lower mass of the exciter system compared to a conventional subwoofer also results in better vehicle fuel consumption for the vehicle. The required power consumption to drive the glass subwoofer system is also a fraction of the demand of a conventional subwoofer.

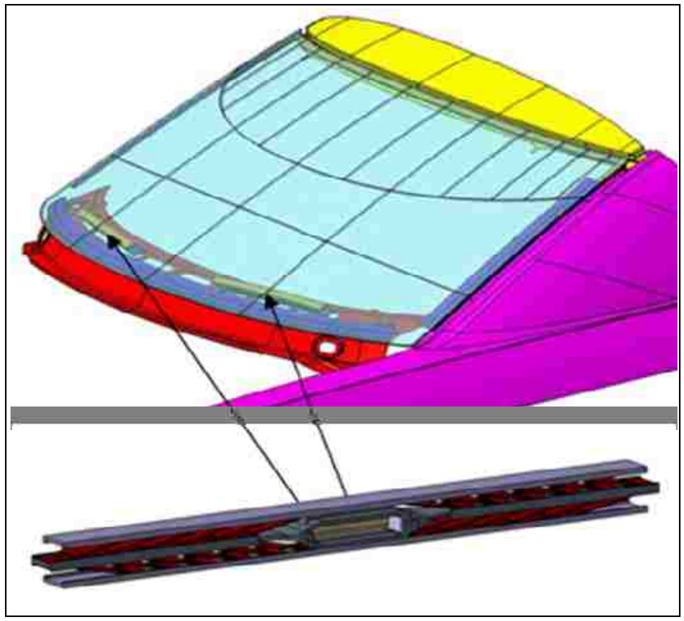


Figure 1.2 – Rear Glass Subwoofer System [Courtesy of Magna Exteriors and Interiors Division]

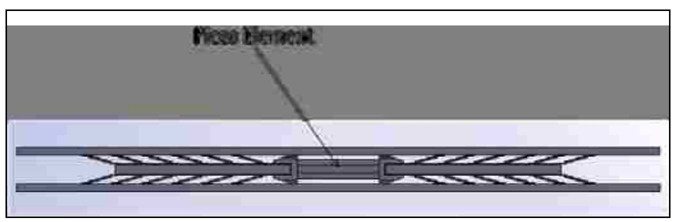


Figure 1.3 – Piezoelectric Actuated Exciter [Courtesy of Magna Exteriors and Interiors Division]

1.3 Objectives

The research objectives of this study are as follows:

- Investigate the effect and evaluate the contribution of the glass subwoofer system on vehicle interior sound quality
- Compare the glass subwoofer system to a conventional subwoofer system and obtain relationships between objective measurements and subjective evaluations of both systems

- Develop a standard testing procedure suitable for the measurement and evaluation of the glass subwoofer system acoustic properties

As this research explores the novel idea of utilizing the rear glass as the driver for the vehicle's subwoofer system, different measuring and evaluation techniques are used in order to develop a standardized testing guidelines to be used to rate the system. Physical indices of sound including basic signal analysis, level and spectrum, frequency response and total harmonic distortion are implemented together with more aurally adequate indices including binaural loudness and subjective response. The vehicle selected for this work is a full size Chrysler 300 sedan equipped with both the baseline glass subwoofer system and upgraded conventional factory installed subwoofer system. The intent is that any guidelines provided in this work can be easily implemented in any vehicle type and model for successive estimate of glass subwoofer impact on vehicle interior sound quality.

CHAPTER II

REVIEW OF LITERATURE

Presently, no published studies on the acoustical performance of the glass subwoofer technology have been found in the literature. There are also no published studies on the comparison of the glass subwoofer versus the conventional subwoofer technology for in vehicle applications. Therefore, the following literature survey was undertaken in order to investigate the fundamentals of the automotive audio system and to understand the available measurement methods in order to quantify and compare the sound characteristics of the automotive subwoofer systems under consideration. Ultimately, the goal of the literature survey is to select the relevant and applicable analysis methods to compare the acoustical performance of the conventional and the glass subwoofer technologies for in-vehicle applications. Any testing methodology would be applied to both subwoofer systems. Lastly, it is important to acknowledge that the glass subwoofer system incorporates alternative technologies, which are different from a conventional subwoofer system. As such, some common electroacoustic measurements such as electrical impendence are omitted from any analysis.

This chapter describes the historical background related to the automotive audio system development. This is followed by a review of publications on the subject of piezoelectric technology for automotive applications. Next, the applicability of typical electroacoustics measurements for loudspeaker performance specification is discussed as it relates to this study. The theories of psychoacoustics analysis, including loudness metric analysis, and subjective evaluation, are introduced and described as potential methods of comparison between the two subwoofer systems. The next section is a summary of a publication describing component level analysis in terms of the contribution of the various sound sources to the receiver at the ear of the listener inside a vehicle, with special emphasis to the vehicle audio system. Although the component level analysis is not a subject of this study, the methods described in this publication serve as a suggestion for future work, or next steps, related to the future development and improvement of the glass subwoofer technology, and for that reason its inclusion in this literature survey is justified. Lastly, a literature survey of monaural and binaural vehicle interior measurements is presented in order to assess the potential contribution of each in the evaluation of the subwoofer systems considered in this study.

2.1 Automotive Audio System Development

In this section, the historical background of subwoofer design and specifications are discussed as they relate to the conventional subwoofer used for automotive applications.

2.1.1 Historical Background

The roots of the automotive audio system go back to the 1930's. The first to introduce an in-vehicle car radio were the Galvin brothers [3]. They named their system "*Motorola*" which was derived from words "*motor*" meaning motion and "*ola*" meaning sound. The first car speaker was only one centre speaker located in the dashboard. Soon after, other companies from around the world, including the German company *Blaupunkt*, began to develop automotive stereo systems [4]. Significant development was achieved in the early 1950's with the introduction of *FM* stereo broadcast and the launch of radio systems with more than just one loudspeaker. These loudspeakers were simply home audio speakers simply installed in the vehicle. The problem with this was that they were not well suited for the vibration and temperature conditions within the vehicle and further development was necessary to adapt speakers to such extremes. Advancements in the electrical system, including switching from 6.3 V to 12 V vehicle batteries, allowed for the further development of automotive radio systems and the introduction of first *16* $^{2}/_{3}$ *RPM* disc players by *Motorola* in 1956 [3]. A few years later the *45 RPM* record player was introduced followed by the 4 track tape player which became the first commercially available car stereo system. In a quest to develop a more powerful audio system, Jim Fosgate manufactured in 1978 the first 12 V amplifier for use in a car stereo system. Not long after in 1983, *Zed Audio* developed a 200 W per channel amplifier. These advancements led to the integration of low frequency loudspeakers called subwoofers into a vehicle audio system. Bass reproduction became one of the most significant differences between low-cost and premium audio systems. Throughout the last three decades, automotive subwoofer systems went through continuous improvements and modifications of its original design while still sustaining its primary components. Design and specifications of typical automotive subwoofers are discussed in the next section.

2.1.2 Automotive Subwoofer Design and Specifications

Subwoofer systems are intended for limited low frequency range (20 Hz – 200 Hz), and as such, they require careful design consideration. In order to accurately reproduce low frequencies without distortions caused primarily by unwanted resonances, subwoofers must have a solid well braced construction. Better subwoofer systems are typically quite heavy and include power amplifiers with additional controls relevant to the low frequency reproduction [5]. These amplifiers can be either active built in the subwoofer system, or passive external amplifiers. A typical subwoofer system design is shown in Figure 1.1 of Chapter 1.

The diaphragm, or cone, is connected to a stiff frame through a flexible suspension system which consists of a spider and surround [5]. The spider, or damper, provides a restoring force and functions as a voice coil and cone centering mechanism through its range of travel. Additional control is provided by the surround which greatly contributes during the long subwoofer excursions. Attached at the bottom of the cone is the voice coil which extends into the magnetic gap between magnets and the pole piece. When an electrical signal from the amplifier is fed to the voice coil, it becomes an electromagnet which interacts with the speaker driver's magnetic system. Mechanical force is then generated which causes the voice coil to move the diaphragm axially back and forth, thus disturbing the immediate air pressure and producing a sound [5]. The subwoofer's excursion which is visually seen as cone extensive displacement inward and outward, together with the driver diameter, is a primary contributor to high acoustic output in any conventional driver on the market [5].

In order to rate a subwoofer system, general electrical, mechanical and acoustical characteristics are selected and include [6]: subwoofer system weight (lb), size of the driver (in), electrical impedance (Ω), rated power (W RMS), sensitivity (dB at one meter distance and 1W RMS input), frequency response (dB at Hz), and total harmonic distortion (%).

Rated power, defined as the maximum power that a subwoofer can handle before being damaged, will usually range between 50 and 400 W RMS for a premium subwoofer system. This information is based on data from the leading automotive audio system companies and the fact that a maximum available power of 500 W for in-car sound systems is limited by the alternator output [7]. It should be stated that this is not a true measure of the sound output which a subwoofer can produce. Further, the speaker driver can be damaged at much less rated power if driven extensively beyond its mechanical limits, especially at very low frequencies. This measure is omitted from the comparison of the two subwoofer systems involved in this study as they incorporate different electromechanical concepts.

The sensitivity, or speaker efficiency, is defined as the sound pressure level generated by a speaker and measured under the free-field conditions one meter away from the source at 1 watt RMS power input at selected frequencies. Studies have shown that a speaker rated 3 dB more than another requires only half of the rated power for the same output [6]. Premium automotive subwoofer sensitivities range anywhere between 85 dB and 95 dB at 1W RMS. Since these measurements are conducted at 1 W RMS power under free field conditions [8], this specification will not be used for in-vehicle testing and comparison of two different subwoofers of interest.

Frequency response characterizes a speaker's output for the constant input level over the frequency range of interest. As mentioned in the introduction, the typical frequency range of an automotive subwoofer based on the specifications provided by the leading automotive audio system companies is between 20 Hz and 200 Hz, although these figures may vary for in-vehicle measurements as is the case in this study [5]. The frequency response specifications should be supported by the corresponding graphs to adequately characterize the response.

Total harmonic distortion is an optional specification sometimes included on a subwoofer label. This metric is quite important because it describes a non-linear behavior of the subwoofer system output.

2.2 Piezoelectric Technology and Automotive Applications

Piezoelectric technology incorporates crystalline materials which deflect and change shape when a voltage is applied to it. These materials tend to perform well under a compressive load, but are weak and break when subjected to a tensile load. These materials are quite often used in low cost, high frequency applications that do not require high output levels. Typical examples of piezoelectric components used for sound production purposes are found in the patent, *Piezoelectric Acoustic Speaker System*, 1976 [9] by Kinoshita. The inventor introduced a piezoelectric speaker comprised of the piezoelectric diaphragm enclosed in a cylinder with multiple vibrating regions. The purpose of this invention was to present a piezoelectric speaker which is capable of altering the directional characteristic of sound. Kumada et al. disclosed in his patent, *Transparent Flat Panel Piezoelectric Speaker*, 1982 [10] the integration of a transparent flat panel mounted to a piezoelectric actuator to produce high frequency sound in watches. Another example of utilizing piezoelectric drivers to produce mid/high frequency sound outputs can be found in *Piezoelectric Speaker*, 1990 [11] by Takaya. The application of multiple piezoelectric elements used to drive a flexible panel and produce high intensity sound outputs under extreme environments is demonstrated in patent *Piezoelectric Panel Speaker*, 1993 [12] by Shields.

There are also several attempts in the automotive industry to incorporate the piezoelectric technology into automotive sound systems. For example, the patent, *Piezo*

Speaker for Improved Passenger Cabin Audio Systems, 1999 [13] by Parrella et al., introduced piezoelectric actuators mounted on the vehicle structure, door panels, roof, and deck lid. This proposed solution was intended for mid and high frequency range sound. In his patent, *Vehicular Loudspeaker System*, 2003 [14] Warnaka proposed the use of piezoelectric actuators within the headliner and trim components to generate mid to high frequency sound. The utilization of multiple piezoelectric actuators within the headliner is also found in the patent *Vehicular Audio System and Electromagnetic Transducer Assembly for Use Therein*, 2006 [15] by Emerling et al. Since no amplification is used for the actuators excitation, the displacement amplitude is low and limited to mid and high frequency sound.

2.3 Electroacoustic Measurements for Loudspeaker Performance Specification

Typical loudspeaker performance specifications are based on electroacoustic measurements at a one meter distance from the loudspeaker axis in free field [8]. Such electroacoustics measurements cannot be fully utilized in this study given that the rear glass subwoofer system is an integrated part of the vehicle, and as such, incapable of being tested separate from the vehicle under free field conditions.

The two most common electroacoustics parameters are the frequency response and total harmonic distortion. These two parameters describe the dynamic behaviour and linearity of the subwoofers under the consideration and as such will be used in this study [16].

2.3.1 Frequency Response

In the automotive industry, frequency response measurements are commonly performed for various combinations of vibroacoustic inputs and outputs and are not strictly associated with loudspeakers and electroacoustics measurements. Several studies used frequency response as a performance parameter for the conventional automotive subwoofer [16, 17, 18]. There are no published studies dealing with the frequency response of the automotive glass subwoofer system. Therefore, for this study the frequency response is used as an objective analysis method for evaluating and comparing the performance of the conventional and the glass subwoofer system.

2.3.2 Total Harmonic Distortion

Studies have shown that the sensitivity of human hearing to nonlinear woofer distortions is around 5% for real signals [19]. A typical automotive subwoofer system commonly produces up to 10% total harmonic distortion [20]. This type of distortion is tightly related to trim and panels which are the main contributors of sound distortion inside the vehicle cabin [16]. The rear glass of the rear glass subwoofer system may potentially behave as one of these panels. Therefore, the total harmonic distortion appears to be a relevant objective performance parameter that would eventually be used in this study to evaluate and compare any non-linear behaviour of the two subwoofer systems.

2.4 Psychoacoustics

Psychoacoustic is the science of the human perception of sound. It involves not only the physical science of acoustics but also a psychology of human hearing. Psychoacoustics employs metrics which provide a more meaningful insight of sound as perceived by humans [21]. At this time there are no published studies on the psychoacoustic evaluation of automotive subwoofer systems based on any of the currently available psychoacoustic metrics. This section provides an overview of literature related to the most common psychoacoustic metric loudness. The description of the development of loudness in this section is significant because it uncovers its applicability for the purpose of evaluating vehicle interior sound quality of the two automotive subwoofer designs considered in this study. This will potentially provide a more detailed insight into the effects of this vehicle system on the interior vehicle sound quality as perceived by an automotive customer. The same argument applies to any subjective evaluation of the two subwoofer systems.

The roots of the psychoacoustics doctrine start in early 1930's with the first known paper on sound perception presented by Fletcher and Munson [22]. The area of psychoacoustics becomes well established by the 1950's, and it gained a high attention in the last decade. Two major techniques utilized in the psychoacoustic evaluation of sound are: the use of objective metrics such as loudness to estimate sound perception by the listener, and the subjective evaluation where the listener's subjective opinion is used to describe the characteristics of perceived sound. There are many objective metrics developed to date, but only loudness will be discussed here since other psychoacoustic metrics such as sharpness, roughness, etc. do not have a significant association with this type of low frequency sound.

2.4.1 Loudness

According to Zwicker [21], loudness is a metric which closely matches the perceived intensity of a sound. Since the first notable introduction of loudness by Fletcher and Munson [22] in 1933, there has been extensive research and steady progress in the understanding of the loudness model. It has been defined through experiments that the loudness level of a sound is equal to the sound pressure level (SPL) of a 1 kHz tone.

Perhaps one of the most significant contributions to understanding the relationship between perceived loudness and sound pressure level is the creation of the equal loudness-level contours by Stevens in 1956. These curves have since been improved and are given as ISO 226: 2003 [23] as different models have been proposed and standardized. Another noteworthy improvement in loudness characterization was made by Zwicker which involved the use of critical bands where frequency is defined in "Barks" as opposed to "Hertz". It is important to note that the bark scale corresponds linearly to the Hertz scale for lower frequencies up to approximately 500 Hz [21] meaning that the bandwidth is constant at 100 Hz (1 Bark = 100 Hz, 2 Bark = 200 Hz., etc.). Table 2.1 illustrates the relationship between "Bark" scale and "Hertz" scale.

Table 2.1 – Critical Band Rate According to Zwicker and Fastl, 1990 **[21]**

Critical Band [Bark]	Centre Frequency [Hz]	Critical Band [Bark]	Centre Frequency [Hz]
0	0	13	2000
1	100	14	2320
2	200	15	2700
3	300	16	3150
4	400	17	3700
5	510	18	4400
6	630	19	5300
7	770	20	6400
8	920	21	7700
9	1080	22	9500
10	1270	23	12000
11	1480	24	15500
12	1720		

2.4.1.1 Equal Loudness-Level Contours

Given that sound is not perceived equally across the entire audible frequency range [21], equal loudness level contours are developed based on the experimental data to account for these differences. Figure 2.1 below illustrates equal loudness level contours.

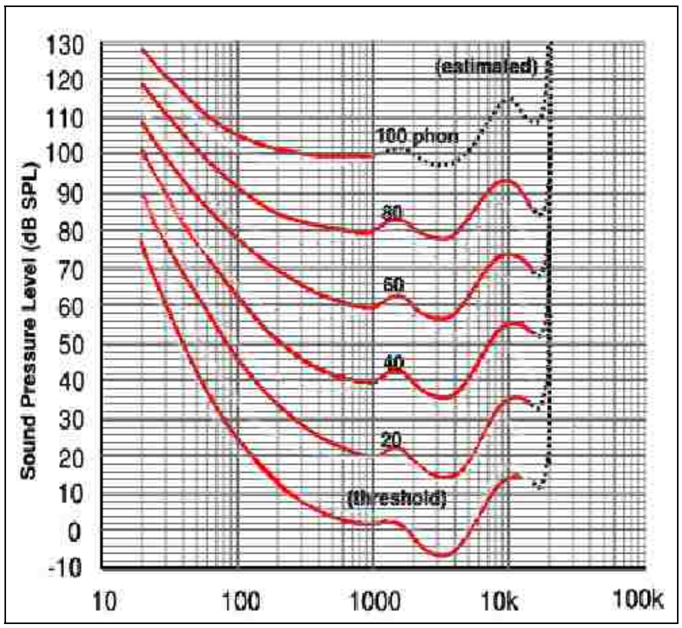


Figure 2.1 – ISO 226:2003 Equal Loudness-Level Contours **[23]**

2.4.1.2 Binaural Loudness

In order to evaluate the sound characteristics, the human auditory system employs two receivers; the left and the right ear. This allows for not only identification of sound sources, but also their localization in the tridimensional field [24]. Binaural loudness can be described as an additional step involved in loudness calculation for more precise estimate of perceived sound characteristic. According to Noumura [24], each ear receives a different sound pressure signal from each different source. In his paper Noumura explains that humans localize a sound image based on the differences in amplitude and phase at each ear. It is necessary to calculate the loudness level at both ears and to account for simultaneous masking effect which is calculated based on the centre frequency and sound pressure level at each critical band for both ears. These values are further summed together via binaural add method, described in [24], and binaural loudness is calculated.

2.4.2 Subjective Evaluation and Paired Comparison

A well known set of guidelines for acoustical subjective evaluation in the automotive industry was published by Otto [25]. A selection of a long list of best practices learned from the experience of automotive NVH engineers over the years are summarized in this section. These guidelines were followed in the subjective evaluation component of this study.

Subjective evaluation is a vital factor for assessing a product’s competitiveness. It is the final stage of sound quality evaluation and it involves a group of jurors in a listening test. The test must be conducted through the entire design process with the greatest care and accuracy. Several critical aspects involved in subjective testing may then be generalized and put into practice.

One area of concern is the proper selection of the testing environment. The environment must be carefully selected and be free of excessive background noise or any other sources influencing the evaluation procedure. Besides permissible ambient noise, there are many factors which can affect the juror’s preference during a test including the room’s acoustics, ambience, temperature, and humidity, and as such they each need to be addressed.

The selection of jurors also plays an important role in the proper execution and desirable outcome of the subjective test. Listeners must be carefully chosen for their demographic position, economic status and the probability that they are potential customers to the product under test. It is desired that jurors be trained or at least familiar with the product to some extent. The appropriate number of jurors can range anywhere between 25 to 50, mainly depending upon the time constraints and the availability of the subjects.

Adequate presentation of the evaluated sounds or systems, as well as the proper specific instructions given to the listeners, is necessary to warrant consistent and valid test results. Studies have shown that subjective evaluations are best done blind [26], since listeners with certain brand preferences tend to rank those systems as better in sound quality despite various audible shortcomings [27].

Two methods of subjective evaluation can be utilized; semantic differential test, in which each recording is rated on an absolute scale, or paired comparison testing in which sound of preference is to be chosen. A paired comparison method was chosen for this study due to its simplicity and the fact that only two sound sources with limited frequency range are compared. Another reason for choosing this method is its effectiveness when employing untrained jurors. The paired comparison technique allows the listener to be presented with a sequence of pairs of sounds where the listener has to decide on the sound of preference.

The final step involves the process of the validation of the test results and correlation of the objective matrices with the subjective preferences. Several different correlation techniques such as linear regression can be utilized to obtain confident levels of relationship between objective and subjective indices. Data plots of "Actual versus Predicated" values or scatter diagrams are usually used to present and validate the results.

2.5 Transfer Function Model of Automotive Audio System

The acoustic of a vehicle cabin is characterized by a combination of materials with different acoustic properties, reflecting surfaces and relatively small air volume. Sound reflections in such enclosure can significantly contribute to the direct signal of a sound at its early stage of propagation and will have a negative effect on its colouration [17]. Employing proper techniques, near-field measurements can provide meaningful comparison of acoustic quantities independent of vehicle interior environment [18]. However, not to acknowledge the effect of the room acoustics and associated influence on a subwoofer performance would be short-sighted. Having in mind that the complex design of a vehicle cabin corresponds neither to a free field nor a diffuse field, it is necessary to investigate and point out all possible sources of sound contamination on its path from the source to the receiver. In other words, it is necessary to define a transfer function model of the automotive audio system. "The relationship that exists in the steady state between the output signal and the input signal of a two-port device is called the transfer function" [28].

The first transfer path (T1) is the electronics of an audio system with its frequency response, loudness curve and distortion characteristics [29]. For this study T1 is not of a major concern since the same head unit is used for both subwoofer systems' comparison. The second transfer path (T2) corresponds to the loudspeaker, in this case subwoofer, and its wiring harness [30]. The loud speaker alters the input signal in such a way that its frequency response is never perfectly flat, especially in vehicle measurements. The next transfer path (T3) is the mixture of the trim and panels that the loud speaker is attached to. As stated earlier this transfer function is one of the primary contributors to sound distortions inside the vehicle. It adds “rattle” and “buzz” noises which correspond to dips and peaks in the frequency response measurements [29]. The following transfer path (T4) includes mounting brackets, grills and accompanying cavities. T4 also alters the sound and causes distortions in the loud speaker output. Both T3 and T4 merge into the room acoustic of a vehicle cabin which is the transfer path T5. It represents a sound package of the vehicle cabin including interior dimensions and surfaces all made up from different materials with dissimilar absorption and diffusion characteristics. Taking into account the listener’s close proximity to the reflective surfaces, an important extension to T5 is the location of the listener himself, or transfer path (T6). It confines the effect of room acoustics between the source, loudspeaker, and the receiver, listener. The next transfer path (T7) is the listener himself who also contributes to the overall acoustic result. The presence of body, especially the shadowing effect of the head and ears, modifies the sound field [31]. This is one of the reasons why it is recommended to use head and torso simulator for aurally correct measurements. The last and optional transfer path is the ear pinna shape (T8). Since being specific to each individual, it is usually omitted from the sound system transfer function model.

2.6 Summary

The literature survey presented in this chapter summarizes the present state-of the-art dealing with automotive subwoofer system acoustical performance characterization, including its frequency response, total harmonic distortion, loudness and subjective analyses. There are presently no published studies related to the acoustical performance of the automotive glass subwoofer system, an alternative green technology when compared to the conventional automotive subwoofer system. Therefore, the results presented in the following chapters address this shortcoming by presenting an objective and subjective evaluation and comparison between both the conventional and glass subwoofer systems. In addition, a standardized testing procedure suitable for measuring and evaluation of the glass subwoofer system acoustic properties is presented as a recommendation for future implementations.

CHAPTER III

THEORY

This chapter describes the theory associated with the digital signal processing parameters used to measure, store and analyze the acoustical signals for the objective analysis and the comparison of the two subwoofer systems under consideration. It also describes the theory behind the sensors selected for the study used to obtain the measurements including a discussion of the advantages and disadvantages of each, particularly as it relates to the in-vehicle measurement environment. The chapter concludes with a description of the vehicle interior sound pressure and vibration measurement techniques utilized.

3.1 Analog to Digital Signal Conversion

This section discusses the methods used to reduce analogue to digital conversion errors such as aliasing and leakage which are associated with the frequency and chosen sampling period of the acquisition process. The measurement apparatus is typically comprised of: a) a sensor which has some characteristics that are sensitive to the measured variable, and b) a transducer which converts change in characteristics to a detectable signal [32]. This acquired signal is generally a time varying voltage which is converted to a digital form by sampling and quantization process utilizing an analog to digital converter (ADC). Sampling represents a method used to convert a time varying signal to a discrete time signal of continuous amplitude by collecting discrete data values at equal time intervals [32]. Quantization refers to a process of converting a continuous amplitude signal to a discrete amplitude signal. In other words, it is a measure of precision of amplitude conversion from analog to digital domain. For this study, the continuous analog signals of interest, sound pressure and acceleration were converted to discrete signals, that is, they needed to be approximated or sampled. Sampling can be considered as a product of a continuous analog signal and a discrete valued sampling function of unit amplitude, resulting in a discrete time signal with equally spaced amplitude values in time [32]. One of the main assumptions associated with this approximation is to correctly identify a maximum frequency of interest and the sampling frequency. As an example, the maximum frequency (f_m), of a sinusoidal sound wave, is equal to the inverse of its period (T) and is expressed as follows:

$$f_m = \frac{1}{T} \tag{3.1}$$

Sufficient number of samples is necessary to obtain a valid reconstruction of this analog signal, or a sound wave [32]. In other words, time intervals between samples must be small enough to maintain the maximum frequency. The sampling frequency (f_s) is defined as follows:

$$f_s = \frac{1}{\Delta t} \tag{3.2}$$

Where: $\boldsymbol{\Delta t}$ represents the time between the samples.

3.1.1 Sampling Frequency Considerations

Based on *Shannon's Sampling Theory* and *Nyquist Criterion*, in order to extract valid frequency information of the analog signal, the sampling frequency must be at least 2.56 times greater than the maximum frequency [32].

$$f_s > 2.56 f_m \tag{3.3}$$

Simply, the higher the sampling frequency, the higher is the likelihood of capturing the maximum frequency contained in an analog signal. On the other hand, if a sampling frequency is too low, it can generate a false frequency and can lead to an aliasing error. The PULSE LabShop version 15 software used in this study uses the Nyquist Criterion to determine sampling frequency based on the maximum frequency of interest specified in software settings before each measurement. The proper selection of the sampling frequency was critical in this study to ensure accurate frequency content of measurement data. The next section describes the consequences of a common measurement error associated with an improperly selected sampling frequency.

3.1.1.1 Aliasing

Sampling at too low a frequency can lead to the problem called aliasing which can cause erroneous results and invalid representation of original analog signal as illustrated in Figure 3.1. The green continuous waves represent the analog signal, for example sound pressure or acceleration, while the red dotted line shows sampling signal with too low a sampling frequency. One can see that this sampling frequency does not capture all necessary discrete points in order to accurately represent original signal.

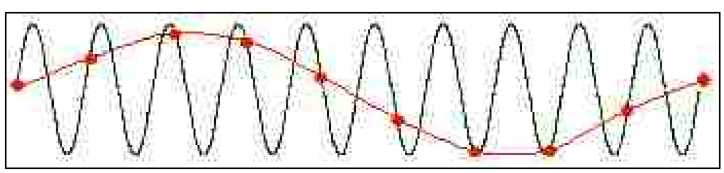


Figure 3.1 – Aliasing Effect in the Time Domain **[33]**

This problem can be overcome by implementing the Nyquist Criterion which stipulates a proper sampling frequency previously defined in Equation 3.3. The effect of aliasing is also applicable in the frequency domain as illustrated in Figure 3.2. All multiples of Nyquist frequency (f_n) act as the folding lines for the frequency components labelled as

f_1, f_2, f_3, f_4. Frequency f_4 is folded back on f_3 around line $3 f_n$, frequency f_3 is folded back on f_2 around $2 f_n$, and frequency f_2 is folded on f_1 around f_n. Thus all the signals at these frequencies are seen as the signals at f_1 and it can be concluded again that the lowest frequency at which aliasing can occur is approximately half of the sampling frequency (f_s).

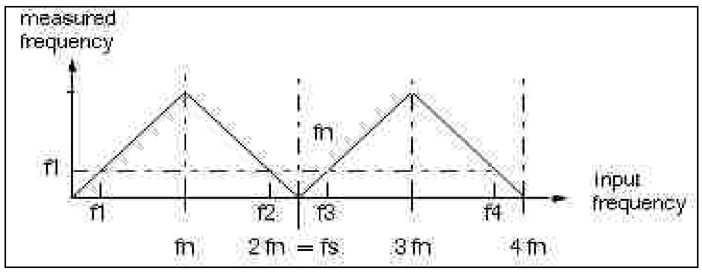


Figure 3.2 – Aliasing Effect in the Frequency Domain **[33]**

3.1.1.2 <u>Filtering</u>

Filters are used in this study to attenuate and remove the undesirable frequency content from the dynamic signal. For example, low pass filter cuts off higher frequencies above the specified cut-off frequency, whereas high pass filter removes lower frequencies below cut-off limit. Band pass filter removes frequencies above and below a selected frequency band, whilst notch filter cuts off frequencies within specified frequency band. Illustration of different types of ideal filters is shown in Figure 3.3.

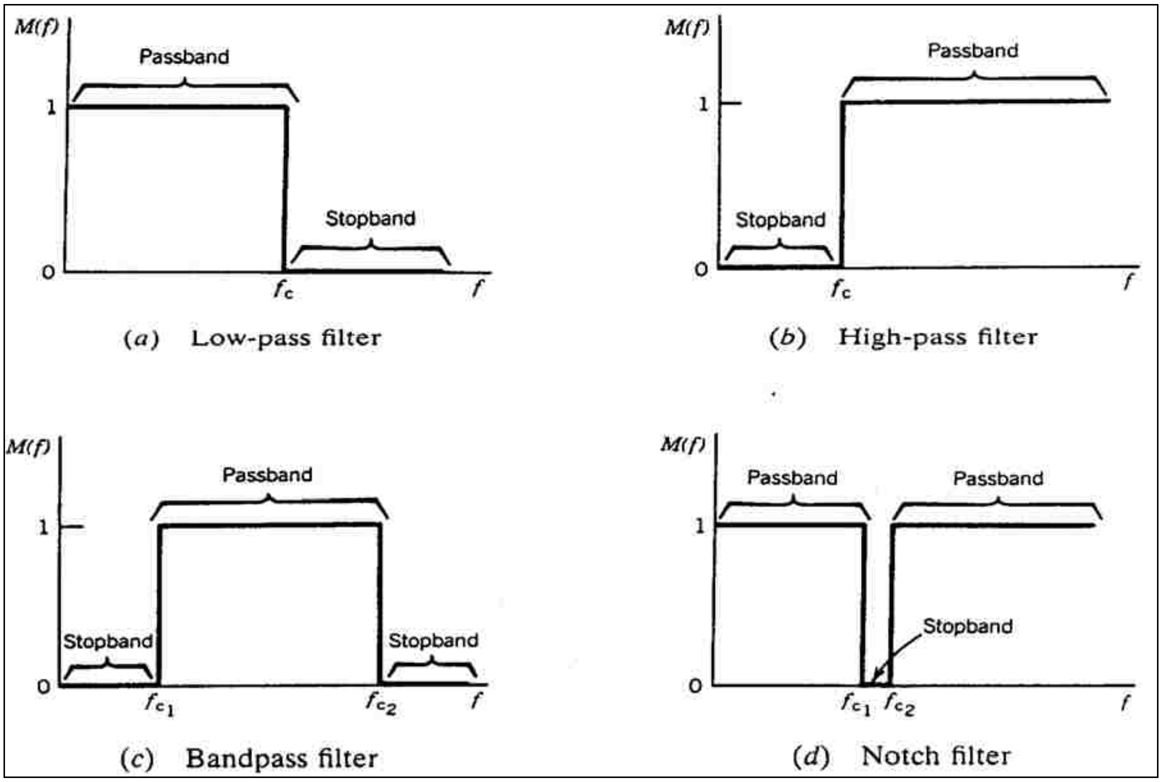


Figure 3.3 – Different Types of Ideal Filters **[32]**

Real filters such as the ones used in this study, are less than ideal. An example shown in Figure 3.4 illustrates that the position of the cut-off frequency must be made with respect to maximum frequency and the roll-off characteristics of the filter. Typical roll-off point occurs at 80 percent bandwidth so the rest of bandwidth might contain faulty data.

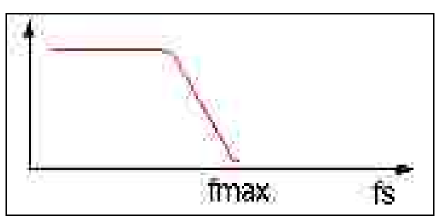


Figure 3.4 – Low Pass Real Filter **[33]**

Generally, analog to digital converters apply low pass real filters to the analog signal prior digitization in order to prevent aliasing. For this study, when the equalizer filter was turned off the low pass filter was utilized. Its decay slope was -12 dB per octave. The explanation of this real filter specification mentioned in the next section is now clarified.

3.1.2 Sampling Period Considerations

The discrete time sampling associated with digital signal processing is characterized with a certain sampling period [32]. For example, a continuous sine wave, potentially representing a sound pressure or acceleration wave, should result in the single spectral line, as shown in Figure 3.5.

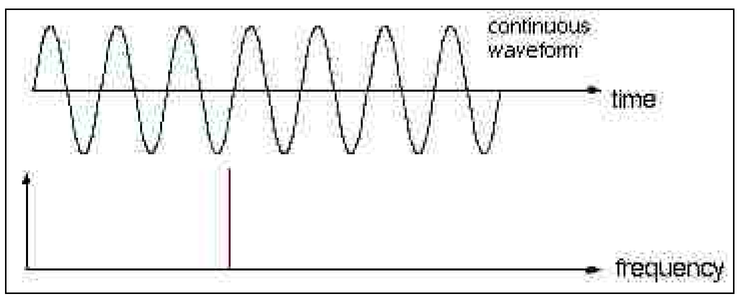


Figure 3.5 – Single Spectral Line as a Result of Periodical Waveform **[33]**

In reality, this is only attainable if the sine wave is periodical in the time domain, otherwise a leakage of energy occurs. As this is one of the most common issues associated with digital signal processing it is an important consideration in this study.

3.1.2.1 Leakage

Leakage of energy in the frequency domain, as described in Figure 3.6, is a consequence of taking only a finite length of time data history. This issue is unavoidable when dealing with digitally sampled signals. As a result, the goal is to minimize the errors associated with leakage.

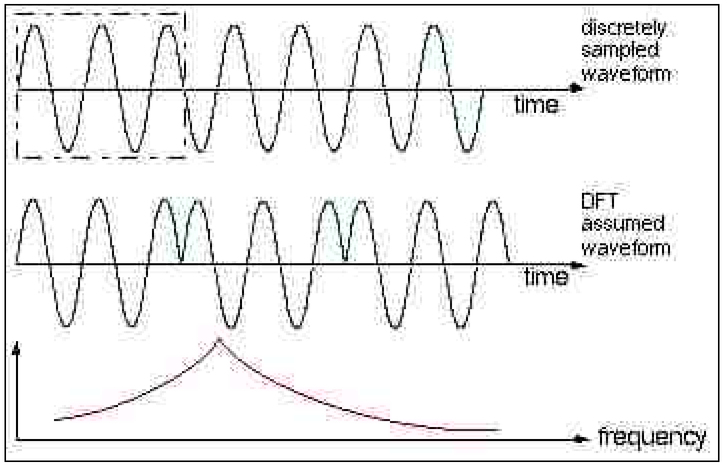


Figure 3.6 – Leakage of Energy as a Result of Finite Length of Time Data **[33]**

Although leakage errors cannot be completely eliminated they can be greatly reduced by employing various excitation techniques and by increasing the frequency resolution [32]. This effect can also be reduced by proper windowing method. Windowing was also employed in this study and is explained in the next section.

3.1.2.2 Windowing

Windowing techniques are used to reduce the leakage of energy which can mask the presence of small signals. It is due to the discontinuities at the edges of sampling period which cause the leakage problem. This can be overcome by ensuring that the sampled value is multiplied by zero at the beginning and the end of the sampling period, thus creating the periodic sampling signal [32]. Although useful, windowing techniques also give rise to errors itself by disturbing energy content of the data. Several different types of windows exist. The most common ones include: rectangular (uniform) window, Hanning window, and flattop window which are illustrated in Figure 3.7.

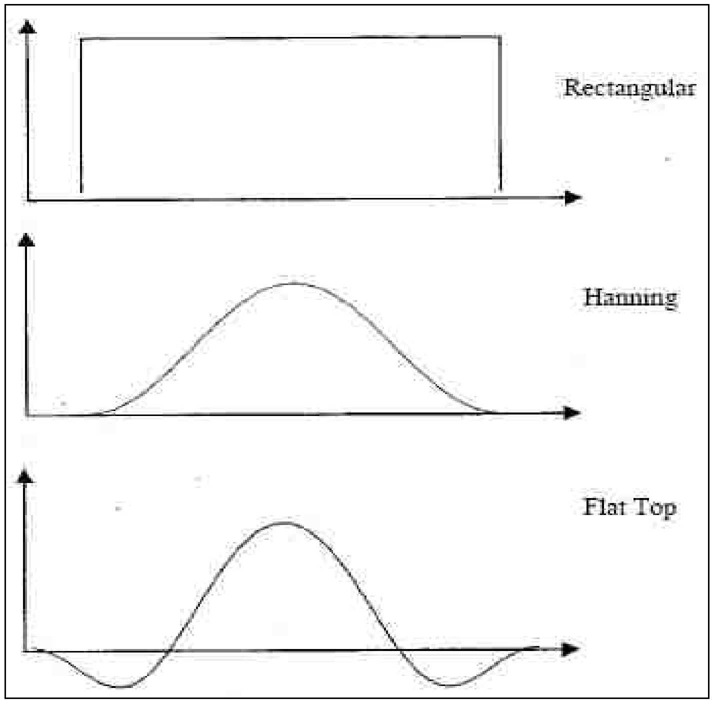


Figure 3.7 – Different Types of Weighting Windows **[33]**

Rectangular windowing is generally used when leakage is not an issue since it does not affect the energy distribution. The Hanning window is commonly applied to random signals with the discrete frequency components, whereas flattop windowing is mainly suited for calibration purposes. Their application is based on the type of excitation signal and desired trade-off between dynamic range and resolution. Based on the above description, the Hanning window was the most appropriate window for use in this study as mentioned in Section 4.3.1.1.

3.2 Excitation Signal Types

The following sections will briefly discuss the types of excitation signals used in this study. The purpose of introducing various input signals was to investigate whether or not they have an effect on the response of the two subwoofer systems, individually and as compared to each other. It is important to emphasize the physical characteristics of these signals in terms of their similarities, but more importantly, their differences. Before performing any testing and analysis involving excitation techniques, it is necessary to select the type of the signal to be analyzed. The selection of a proper excitation signal has an influence on the type of analysis and choice of analysis parameters. The most fundamental division of signals is into stationary and non-stationary signals [34]. Average properties of stationary signals do not vary with the time and are independent of the sample record used to determine them. This analogy applies to both deterministic and random stationary signals. On the other side, instantaneous values of non-stationary signals, both continuous and transient, are function of time.

3.2.1 Pseudo Random (Stationary Signal)

The use of random noise as a test source has the characteristic to spread the signal's energy uniformly over the desired audio spectrum [34]. The main disadvantage, which makes the use of truly random noise impractical, is its inherent nature of randomness. To overcome this issue and to yield absolutely accurate measurements it is necessary to average the values over an infinite time interval. However, in practice, a balance has to be made between averaging time and desired accuracy. The most efficient way to accomplish this is by the use of a pseudo random signal shown in Figure 3.8. Although similar to random noise, pseudo random signal is periodic in nature and produces discrete power spectrum.

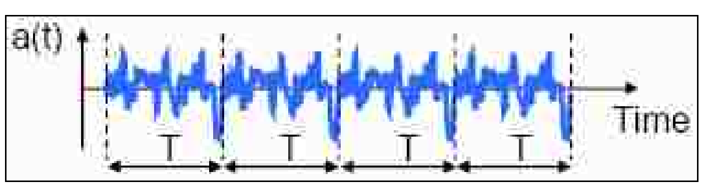


Figure 3.8 – Pseudo Random Signal Illustration **[35]**

Generally there is no need for extensive averaging since the impulses repeat with every period of time T which is the FFT record length [34]. A pseudo random signal can be reproduced exactly and there is no spectral leakage if rectangular weighting is used. This may be of benefit in the standardization of testing.

3.2.2 Swept Sine (Non-stationary Signal)

A sine wave can be described as a continuous cyclic wave form in which amplitude fluctuates according to the sine function of the elapsed time [34]. Since it contains only a single fundamental frequency it may be portrayed as the simplest sound. When a sine wave is gradually varied in frequency value (typically from low to high) over a specified frequency range, it is referred to as a swept sine or simply sweep. It is the most common non-stationary signal utilized in practice and it is shown in Figure 3.9.

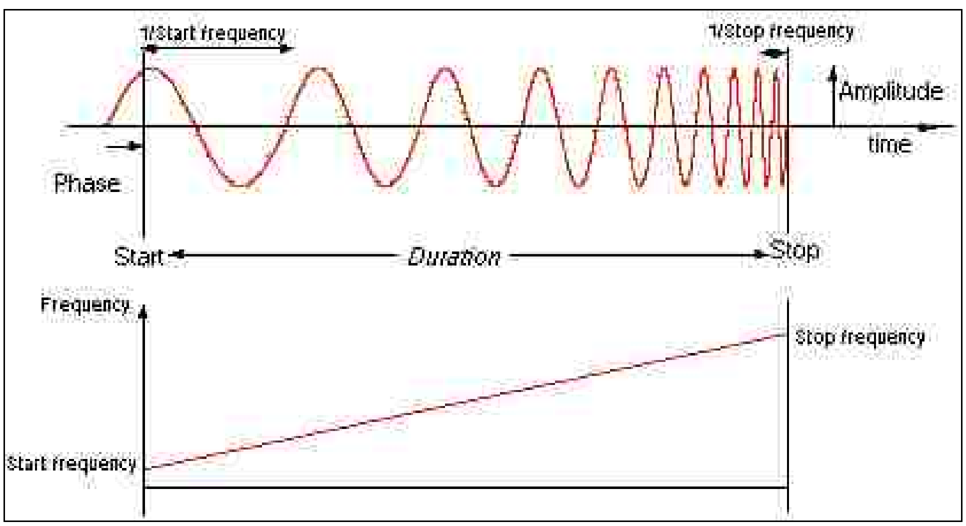


Figure 3.9 – Swept Sine Signal Illustration **[33]**

Because of its high immunity against distortion, low crest factor and high signal to noise ratio, the swept sine signal is most readily used in frequency response measurements

[36]. Compared to random noise, the swept sine signal provides much better coherence characteristics between the input and output signal. Swept based measurements are also less prone to negative effects of time variance.

3.2.3 Stepped Sine (Non-stationary Signal)

A stepped sine is another variation of a sine tone commonly used in electroacoustics. As oppose to a general sine wave where the amplitude gradually fluctuates up and down over the course of the cycle, the stepped sine signal has a series of steps associated with the voltage variations at specified frequencies. All energy is concentrated at the single frequency at the same time and a high SNR is realized [36]. After each individual measurement frequency is incremented by an arbitrary value depending upon desired spectral resolution. Despite the considerable processing time required, the stepped sine is a well established method when it comes to precise distortion measurements [36].

3.3 Sound and Vibration Transducers

A proper selection and physical setup of sound and vibration transducers in this study was an essential step to minimize the undesirable noise effects and to collect valid data of the response to the physical excitations being measured. A thorough understanding of the sensor capabilities and limitations, as well as the type of the desired output signal was established in this section, as related to the sensors used in this study.

3.3.1 Microphones

A microphone is the most commonly used transducer for acoustic measurements which transforms small-amplitude pressure fluctuations into corresponding voltage values. Microphones may include one of the following types of transducers: carbon, ceramic, condenser, moving coil, inductor, ribbon, magnetic, and semi-conductor [37]. The most common microphone design is the condenser microphone which is used in this study and discussed in more detail. It is comprised of a microphone casing, protection grid, and capacitor which incorporates a pair of metal plates, known as diaphragm and backplate, separated by an insulating material [37]. When a small fluctuation in pressure is sensed by the diaphragm plate, it deflects slightly and results in a change of the air capacitance between the diaphragm and the backplate. This is due to the opposite charges being formed on the plates by a polarization voltage. Depending upon the charge formation, microphones can be characterized as pre-polarized, those which incorporate internal charge, or externally polarized, those which require external power supply via preamplifier [37]. Prepolarized microphones were used in this study. Diameter size is another microphone characteristic that should be considered when selecting a sensor. An increase in directional and amplitude sensitivity is achieved with larger diameter sized microphones, whereas smaller diameter microphones have less influence on the sound field. The ½" microphone is the most commonly used size for both high and low SPL. Based on the testing environment, the microphones can be designated into two categories: normal incidence microphone, utilized under the free-field conditions, and random incidence microphone, utilized under the diffuse filed conditions. Free field microphones were used for this study in order to capture the sound pressure in a particular direction of interest, that is, the direction associated with the most sensitive axis of the microphone(s), as explained in experimental details sections and illustrated in figures in Chapter 4. For this study, the Bruel & Kjaer Type 4189 microphone was used (see Appendix B for detailed specifications).

3.3.1.1 Head and Torso Simulator (HATS)

Sound recordings were needed in this research to correctly represent the sound perceived by a listener inside a vehicle. For this reason, recordings were obtained using a Head and Torso Simulator (HATS) unit. The HATS is a standardized model representing the human upper body and head where two free-field microphones are placed at the left and right ears of the head. The HATS compensates for the shadowing effects of the upper body and the head and gives a spatial impression of the sound perceived [38]. It also allows for binaural replay of recorded signals for the purposes of improved evaluation of sound quality.

3.3.2 Accelerometers

Accelerometers were used in this study to quantify the vibration of the rear glass window. The accelerometer operating principle is based on the relationship between a force applied on the mass and resulting acceleration. A typical accelerometer is comprised of a housing, seismic mass and a piezoelectric sensing element [39]. When the housing is accelerated, seismic mass exerts a force on the piezoelectric crystals. The crystals generate a charge proportional to the force created by the acceleration of the mass which is converted to voltage. Careful mounting of the accelerometer is a necessity for obtaining accurate measurements. Depending on various constraints, accelerometers can be mounted in several different ways. Most commonly utilized technique includes stud and adhesive mounting. The accelerometers selected for this study were wax mounted to minimize movement between the sensor and the glass, as well as to minimize loading error due to their small size and weight.

3.3.2.1 Integrated Circuit Piezoelectric Accelerometers

An integrated circuit piezoelectric accelerometer (ICP) is comprised of microelectronic chip built into the transducer and signal conditioner which provides constant current. Due to the low impendence voltage through the system, an excellent signal quality can be achieved even if long cables are used. However, this type of accelerometer is not suitable for extreme temperature and humidity environments due to its electronic limitations. The ICP accelerometers, Type 4507 B selected for this study provided the above mentioned advantages without any sacrifices to the accuracy of data as harsh environmental conditions were not an issue in the experimental setup, described in the next chapter.

3.4 Vehicle Interior Sound Pressure and Vibration Measurement Techniques

In order to evaluate characteristics of a subwoofer sound source, good quality sound recordings were required. These measurements are the starting point for description of perceived sound and they involve different measurement techniques depending on the type of sensor and the purpose of the experiment. Several methods are briefly discussed in the following sections.

3.4.1 Monaural Recordings

The sound pressure field of a vehicle cabin can be characterized by using an array of microphones spread throughout the vehicle passenger space [40]. Recordings conducted via monaural method are suitable for quantifying physical indices of sound, such as SPL and frequency response but are not aurally accurate for psychoacoustic evaluation since the human hearing perception is different from that in the actual sound field due to the effects of the human head and torso.

3.4.2 Binaural Recordings

Differences in sound level and phase between the left and the right ear are due to the complexity of the signal processing in human hearing and sound shadowing by the human head. The binaural method utilizes recordings which are collected via two microphones placed in the physical model of the human head and torso. These microphones simulate human ears and take into account the combined effects of the diffraction of the sound waves reaching the eardrums. As a result, binaural recordings offer an advantage in terms of hearing sound in an aurally correct way [41].

3.4.3 Vibration Measurements

Vibration data was acquired using high quality accelerometers to evaluate the vibration characteristics of the excited window structure to get more precise description of the physical response of a glass subwoofer system.

CHAPTER IV

EXPERIMENTAL DETAILS

One has to consider the properties of the instrumentation and the limitations Iin order to obtain valid and repeatable results as well as the proper selection of measuring techniques corresponding to the different stages of NVH testing. When performing any type of experimental measurement, an attempt must be made to minimize all possible extraneous sources to reduce any uncertainty errors. It is also very important to maintain consistent operating conditions and other parameters which may influence the accuracy of the measured data.

The following chapter describes the instrumentation, experimental set-up and testing procedure related to sound and vibration measurements required to characterize the impact of the rear glass subwoofer system on the vehicle interior sound quality. The methodology used are based on experimental techniques employed in electroacoustics and psychoacoustics intended for the prediction and evaluation of vibro-acoustic characteristics of sound sources.

4.1 Equipment and Instrumentation

The equipment and instrumentation employed in the experimental procedure can be classified in three categories:

- Test vehicle and original audio system which has been modified to allow for proper comparison of two different subwoofer systems
- Testing environment used to facilitate the experimental procedure
- Testing instrumentation and data acquisition system used to acquire and process the experimental data

4.1.1 Test Vehicle and Audio System

The primary goal of this study was to accurately define the acoustic characteristics of the rear glass subwoofer system. In order to compare the alternative technology to the existing one, a similar process must be applied for both systems. The vehicle used for this study and shown in Figure 4.1 was a full sized sedan, 2008 Chrysler 300C, equipped with a basic audio system.

Figure 4.1 – Test Vehicle – 2008 Chrysler 300C

The audio system was comprised of a factory installed head unit with a built-in amplifier and eight speakers, including the 10 inch factory installed subwoofer system. The basic subwoofer specifications are found in Table 4.1 below with additional information provided in Appendix B.

Table 4.1 – Chrysler 300C Factory Installed Subwoofer Specifications

Speaker Size	10"
Rated RMS power handling	300 watts
Nominal Impedance	Dual 4-ohm/ Single 4-ohm
Mounting Cutout Diameter	9-1/4" (235mm)
Mounting Depth	6-9/16" (166mm)
Linear Excursion	2"
Recommended Enclosure	0.5ft ^{3} (14.2 L) volume sealed

4.1.2 Test Environment

The ideal subwoofer test environment would be to isolate the speaker in an environment free of any immediate obstacles where the radiating sound from the source is uniform in all directions and the sound pressure level decreases 6 dB per doubling of distance from the source. This can be simulated in a fully anechoic room. Since the two subwoofer systems had to be tested within a vehicle cabin which does not resemble free-field conditions, it was important that the background noise did not influence the measurements. Because of the size constraints and unavailability of a fully anechoic room, the vehicle was stationed in the University automotive research laboratory where the background noise was within tolerable limits and had no effect on the measurements. The background noise within the vehicle was measured to always be at least 15 dB lower than the measured signal's lowest sound pressure level, SPL, throughout the frequency range of interest.

4.1.3 Data Acquisition Hardware and Analysis Software

The acquisition system and software used for the study was Bruel & Kjaer PULSE and version 15 of LabShop. This analysis software is capable of performing acoustical acquisition and analysis including overall SPL and frequency analysis for both steady state and transient signals. It also enables recording for future signal post processing. An additional module of PULSE, Sound Quality Type 7698, is used for the sound quality analysis. This Sound Quality module is capable of analysing, editing and playing monaural or binaural product sounds. It also allows for setting-up a subjective evaluation and correlation to the objective results. More information can be found in the Appendix B.

The data acquisition front end, a Bruel & Kjaer B-Frame Type 3560 B is utilized in the experimental analysis. This unit includes five BNC input ports, one BNC output port for the generator signal, and one BNC Tacho channel as well as a LAN port to connect to PC.

One set of measurements was conducted using Bruel & Kjaer Type 4189 microphones placed in specially designed microphone fixtures located on all four headrest positions in the vehicle. The other set of measurements involved a Bruel & Kjaer Type 4100 head and torso simulator (HATS) mounted on a specially designed fixture intended to replicate the natural position and height of a passenger. A set of 12 miniature DeltaTron Type 4507 B accelerometers were mounted on the rear glass surface to acquire the vibration measurements.

Prior to the in-vehicle measurements, all microphones and HATS were calibrated using Bruel & Kjaer sound level calibrator Type 4231 and Bruel & Kjaer calibrator exciter Type 4294. More information on the data acquisition system and instrumentation can be found in the Appendix B.

4.2 Audio System Modifications and Set-up

In order to allow for the testing and evaluation of both subwoofer systems, modifications to the original sound system had to be performed. These modifications are classified in two groups pertaining to the subwoofer system being modified. The following sections discuss changes being made to the factory installed audio system as well as additional components required to integrate the rear glass subwoofer system.

4.2.1 Integration and Set-up of the Upgraded Audio System

A factory installed head unit was replaced with the Kenwood KDC-X794. This unit incorporates CD, MP3, USB and AUX inputs and allows for detailed digital set-up of the sound output. This audio unit also features a 5-band equalizer, time alignment, digital E's-crossover, high and low pass filters with adjustable slope, and speaker size optimization for better sound. For this study the equalizer was turned off and the low pass filter with the cut-off frequency of 120 Hz was used together with the decay slope of -12 dB per octave.

The factory installed built-in amplifier was replaced with JL Audio XD 600/6 amplifier which is a full range 6 channel car audio amplifier dedicated for all speakers inside the test vehicle with the exception of the subwoofer. A separate single channel amplifier, JL Audio XD 600/1 is used to amplify the signal to the conventional subwoofer. Both amplifiers were professionally installed in the vehicle's luggage compartment and wired to the audio system. It is important to mention that no modification was done to the factory installed speakers and subwoofer system.

Additional information of the installed equipment is given in Appendix B. Detailed wiring diagram of modified audio system can be found in Appendix C.

4.2.2 Integration and Set-up of the Piezoelectric Actuators and Amplifier

The piezoelectric actuators used to excite the glass, shown in Figure 4.2, were designed and built by Magna International. Two of each of the piezoelectric actuators are mounted along the bottom edge of the automobile's rear window as shown in Figure 1.2. The actuators are electrically connected to the vehicle audio system using the specially designed piezoelectric amplifier shown in Figure 4.3. The piezoelectric amplifier has a current input BNC connector and monitor as well as a voltage BNC input with monitor. The amplifier allows for the setting of a mean piezoelectric voltage and incorporates warning lights for over and under voltage. Detailed manufacturers specifications for this instrument are not currently available.

Figure 4.2 – Magna Piezoelectric Actuator

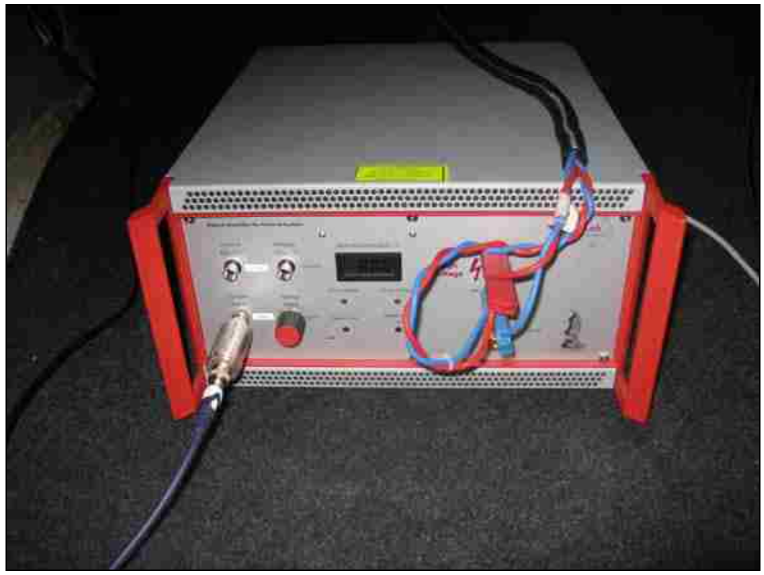


Figure 4.3 – ScienLab Hybrid Amplifier for Piezo Actuators

To allow for interchangeable switching between the conventional and the glass subwoofer system while being engaged, a switch board which is shown in Figure 4.4 was installed in the centre console of the vehicle. This includes 4 switches that can simultaneously turn on or turn off a group of selected speakers as well as a switch between the conventional subwoofer system and the glass subwoofer system. In order to make the operation of the switchboard possible, an LC8i Audio Control unit, shown in Figure 4.5 was installed in the vehicle luggage compartment and connected to the rest of the audio system as shown in the wiring diagram provided in the Appendix C. General features of LC8i Audio Control unit can be found in Appendix B.

Figure 4.4 – Custom Made Switchboard to Switch between Subwoofer Systems

Figure 4.5 – LC8i Audio Control Unit

4.3 Test Procedure

This section describes the technical procedure used to evaluate the impact of the rear glass subwoofer system on the vehicle interior sound quality. It can serve as a testing guidance for future tests of this kind to be performed on any type of a vehicle which incorporates the glass subwoofer system. The execution of the experimental procedure suitable for measuring and evaluation of the glass subwoofer system acoustic properties involves objective and subjective testing described in the following sections. The same testing procedure is used to obtain results for the conventional subwoofer system in order to make comparisons.

4.3.1 Objective Evaluation

An objective evaluation that allowed the commonly employed standard procedure is conducted in all 4 seats inside the vehicle in order to acquire both the physical indices of sound and the psychoacoustic sound quantities of two different subwoofer systems. Work is divided in four sections which include the set-up and analysis of generated signal, as well as the monaural, binaural, and vibration measurements.

4.3.1.1 Excitation Signal Generator and Analyzers Set-up

Before starting the measurement process it was necessary to correctly set-up a generated signal and the analyzer used to process the signals subsequently. During the objective part of the experimental procedure, three different signal types are utilized to excite the subwoofer systems which include: swept sine, pseudo random noise, and a music wave file with a strong bass content. Each of these is described below.

Swept Sine Excitation

- In the swept sine excitation, the generator's signal level is predetermined and set to 500 mV_{rms} in order not to overload a subwoofer system while assuring adequate input level.
- The signal frequency is set to start at 1 mHz and finish at 200 Hz, which corresponds to the low frequencies produced by typical subwoofers.

- The sweep is linear and set to a rate of 3Hz/s in order to assure a gradual propagation of sound wave thorough the frequency of interest.
- The recorder was set for full frequency range with maximum recording length of 70 seconds since it tooks 66.6 seconds for the sweep to finish.
- The frequency of FFT analyzer is set to be 200 Hz since it corresponds to the frequency of interest.
- The number of spectral lines, based on which the frequency resolution, time block, and sampling time are calculated, is also set to be 200, but other values can be used. However, in order to conduct valid data, the product of bandwidth and measurement time value must be at least one or greater. For example, if a very small frequency span is chosen, then a corresponding measurement time must be large.
- For the spectrum averaging, a peak mode is selected and with the time being fixed to 70 seconds it will produce 208 averaging samples. Peak mode is chosen since the spectral energy of the sine wave is concentrated into one frequency and the sine wave reaches its peak value at each cycle. Peak mode indicates the largest amplitude of each spectral line. When a new sample is included, values are compared at each frequency and the largest one is preserved.
- The overlap required to obtain a real time analysis is set to be 66.67%. This gives a uniform overall weighting when employed with a Hanning weighting function which is a type of weighting commonly used for transient signals.

- The constant percentage bandwidth, CPB, analyzer is set to 1/3 octave filter bandwidths with lower centre frequency of 1 Hz and upper centre frequency of 200 Hz.
- The averaging mode is set to exponential in order to place emphasis on the latest sample.
- Averaging time is set to 1 second and no weighting is used.

Pseudo Random Excitation

- For the pseudo random excitation a generator's signal level is predetermined and set to 500 mV_{rms} in order not to overload a subwoofer system while assuring adequate input level.
- The signal frequency span is set to 200 Hz, with number of spectral lines set to 200 as well.
- The recorder is set for full frequency range with maximum recording time of 20 seconds, which is the adequate time length for deterministic random signal.
- The frequency of FFT analyzer is set to be 200 Hz since it corresponds to the frequency of interest.
- Similarly to the swept sine excitation, the number of spectral lines is set to 200.
- The linear mode is selected for the spectrum averaging and with the time being fixed to 20 seconds to produce 58 averaging samples. Linear mode is chosen since the spectral energy of the pseudo random noise is evenly distributed across all frequencies and it reaches its peaks rarely. Linear mode places equal emphasis on all samples.
- The overlap is set to be 66.67%.

- The CPB analyzer settings are left the same as in the swept sine excitation.

Music Wave File Excitation

- For the sound quality and loudness analysis, a music recording with strong bass content is played in the car audio system and recorded for approximately 30 seconds.
- The recorder was set for full frequency range.

4.3.1.2 Monaural Measurements

The monaural measurements employ both swept sine and pseudo random excitation settings since both types of wave forms are used one at the time to conduct the experimental procedure. A set of four microphones are placed in a specially designed fixture located at all four headrest locations inside the vehicle as shown in Figure 4.6. The microphones were placed on the side closer to the windows to capture the highest sound pressure levels. The vehicle was running at idle speed and all doors and windows as well as the sun-roof were closed during the measurements. First the conventional subwoofer was excited and measurements were recorded. The same procedure was then repeated for the glass subwoofer system. Each set of measurements was repeated three times to verify adequate measurement repeatability. Using these, quantitative evaluations including frequency response and total harmonic distortion were determined.

Figure 4.6 – Microphone Set-up for Monaural Measurements

4.3.1.3 Binaural Measurements

The binaural measurements used the same music wave file which was used for the subjective evaluations. This is done to allow for the comparison and correlation of the objective results with the subjective responses. The head and torso simulator was placed in a specially designed fixture located in the driver seat location inside the vehicle as shown in Figure 4.7. As was done for the monaural measurements, the vehicle was running at idle speed and all doors, windows, and sun-roof were closed during the measurements. Measurements are also conducted for both subwoofer systems. Each set of measurements were repeated three times to verify adequate measurement repeatability. Using these measurements, the psychoacoustic quantity of loudness was determined.

Figure 4.7 – HATS Set-up for Binaural Measurements

4.3.1.4 Vibration Measurements

A swept sine excitation signal was used for the conducted vibration measurements. A set of 12 uniaxial accelerometers were located the outside surface of the rear glass as shown in Figure 4.8. The vehicle was turned-off and all doors and windows as well as the sun-roof were closed during the measurements. Only the glass subwoofer system was excited for this test. Each set of measurements were repeated three times to verify adequate measurement repeatability. These additional measurements were conducted to better investigate the total harmonic distortion and sound contamination between the input and output signal. They were also used to identify dissimilarities between the two piezo actuators and uneven displacement of the rear glass.

Figure 4.8 – Accelerometers Set-up for Vibration Measurements

4.3.2 Subjective Evaluation

Subjective tests which involved 27 jurors consisting of University students aged 18 to 25 were performed inside the vehicle in the driver seat position. The jurors performed a paired comparison of sound by switching between the baseline glass subwoofer system and the upgraded conventional subwoofer system. Each evaluator was instructed to select their preferred system based on their subjective experience while listening to a musical composition with significant low frequency content. The tests were blind since the jurors did not know which sound corresponded to which subwoofer system while they manually switched between the two systems during the test. Each juror sat in the vehicle and listened to each individual subwoofer system for approximately 30 seconds. The listening environment was free from any influences including excessive background noise or other participants.

CHAPTER V

DATA ANALYSIS METHODS

Various analysis techniques were used to investigate the impact of the rear glass subwoofer system on the vehicle interior sound quality and to evaluate the acoustical characteristics of the sound source. Due to the unconventional nature of the low frequency source found in this study, a traditional electroacoustic evaluation was modified and combined with a psychoacoustic investigation. A brief foreword of the analysis methods has already been provided in the earlier sections in the form of general characteristics and the developmental stage. This chapter will focus on the theoretical aspects of data analysis methods and their relevance to this study. The analysis is divided into five categories and includes the following:

- Basic frequency analysis (FFT and CPB) used for determination of frequency content and SPL of a sound produced by subwoofers,
- Frequency response function and coherence used to obtain and validate the relationship between input content and output characteristics of the subwoofer system,
- Total harmonic distortion used to grade the linearity and distortion of the subwoofer system,
- Loudness and sound quality used to closely predict the subwoofer's sound perception by the listener in the vehicle, and
- Subjective evaluation and paired analysis used to validate the objective parameters and to gain a better understanding of how different subwoofers are appreciated by potential customers

5.1 Fast Fourier Transform (FFT) and Constant Percentage Bandwidth (CPB)

The fundamentals of a traditional frequency analysis are related to applications of Fourier analysis. This methodology is based on the assumption that real world signals are periodic in nature and contain a finite number of discontinuities in a cycle. For such signals, the Fourier series apply and can be described as follows:

$$x(t) = A_0 + \sum_{n=1}^{\infty} S_x \sin(\frac{2\pi n}{T} t + \phi_n{*}) \tag{5.1}$$

where:

T represents a period of the function, and

$\boldsymbol{A_0}$, $\boldsymbol{A_n}$, $\boldsymbol{B_n}$, $\boldsymbol{S_x}$, and $\boldsymbol{\phi_n{*}}$ are constant coefficients

the constants can be further defined as follows:

$$A_0 = \frac{1}{T} \int_{-T/2}^{T/2} x(t)dt \tag{5.2}$$

$$A_n = \frac{2}{T} \int_{-T/2}^{T/2} x(t) \cos \frac{2\pi n}{T} tdt; \quad n = 1,2,3... \tag{5.3}$$

$$B_n = \frac{2}{T} \int_{-T/2}^{T/2} x(t) \sin \frac{2\pi n}{T} tdt; \quad n = 1,2,3... \tag{5.4}$$

$$S_x = \sqrt{A_n^{\ 2} + B_n^{\ 2}} \tag{5.5}$$

$$\phi_n{*} = \tan^{-1} \frac{A_n}{B_n} \tag{5.6}$$

Based on the integration limits, it is observed that all coefficients are evaluated over one cycle which emphasizes the requirement for a function to be periodic. However, most real world signals are not periodic in nature and certain mathematical transformations are needed to evaluate their frequency content. To be able to investigate transient signals and their frequency content, a Fourier series must be rewritten in alternative form as shown in equation 5.7:

$$x(t) = \sum_{n=-\infty}^{\infty} S_x e^{j\frac{2\pi n}{T}t} \tag{5.7}$$

S_x can be defined as:

$$S_x = \frac{1}{T}\int_{-T/2}^{T/2} x(t)e^{-j\frac{2\pi n}{T}t} \tag{5.8}$$

where:

Δf *represents* the frequency resolution, and ***T*** represents the time period which approaches infinity for non-periodic function

Perhaps, the most useful representation of a Fourier transform is in its numerical form called the Discrete Fourier Transform (DFT) which is used for digitally sampled data and is defined as:

$$x(t) = \lim_{\Delta f \to 0} \sum_{n=-\infty}^{\infty} S_x e^{j\frac{2\pi n}{T}t} \Delta f \tag{5.9}$$

Due to computational intensiveness of the DFT, this transform is not often practical if the number of collected samples is large. The Fast Fourier Transform (FFT) is another algorithm which is used to greatly reduce the number of computations and to obtain the DFT more efficiently. Due to the nature of this algorithm, it is required that the number of sampled data is of the order 2^n where *n* represents the number of samples. One of the great advantages of an FFT is in the fact that it preserves phase information thus allowing for the transformation in either direction. It is desirable to use an FFT with a long time window for the better frequency resolution at low frequencies [38].

Constant percentage bandwidth (CPB) is another representation of data analysis. This analyzer consists of a group of filters whose bandwidth is a fixed percentage of its centre frequency and thus expands on a logarithmic scale at higher frequencies allowing for the better resolution. Depending on the percentage of bandwidth relative to its centre frequency, these filters can be distinguished as 1-octave bands, $^1/_3$ octave bands, etc. For example, the 1-octave band is typically a 70.7 % filter since its bandwidth is always 70.7 % of its centre frequency, whereas the $^1/_3$ octave band filter is always 23 % of its centre frequency. This is defined in the equation below:

$$CPB = \left(\frac{BW}{fc}\right) \bullet 100\% \tag{5.10}$$

where:

BW is the bandwidth, and

Fc represents the centre frequency defined as:

$$fc = \sqrt{f1 \bullet f2} \tag{5.11}$$

5.2 Frequency Response Function (FRF) and Coherence

As a critical evaluation parameter for the subwoofer's acoustical characteristics, a frequency response analysis was performed for the two different subwoofer systems. The frequency response function (FRF) can be defined as the ratio between the output and input signal in the frequency domain and is used to describe the dynamic behaviour of the system. Theoretically, the FRF is developed based on the linear spectra, autopower spectra and crosspower spectra of the input and output signals. The linear spectrum is simply a Fourier transform of a time spectrum whose real and imaginary components correspond to frequency content. The autopower spectrum is a very useful form of computed FFT frequency spectra and is equivalent to the square of the magnitude of the linear spectrum. It is very helpful in identifying key frequency components, but since all imaginary content is removed (thus resulting in spectrum composed of real values only), the phase information is lost and the original time signal cannot be recreated. Autopower spectra can be defined as following:

$$S_{xx}(\omega) = X(\omega) \cdot X^*(\omega) \tag{5-12}$$

where:

X(ω) is a real component of linear spectrum

X*(ω) is an imaginary component of linear spectrum

The crosspower spectrum is commonly used in an analyzer to calculate frequency response and coherence. It can be defined as the product of the signal's linear spectra and complex conjugate of the other one, as indicated in the equation below:

$$S_{xy}(\omega) = X(\omega) \cdot Y^*(\omega) \tag{5-13}$$

were:

X(ω) and ***Y(ω)*** are the specific frequencies of two signals

As oppose to autopower spectrum, the crosspower spectrum includes the phase information.

The frequency response function can then be defined as following:

$$H_1(\omega) = \frac{S_{xy}(\omega)}{S_{xx}(\omega)} \tag{5-14}$$

where:

$\boldsymbol{S_{xy}}$ is a product of linear spectrum of one signal and complex conjugate of the linear spectrum of another,

$\boldsymbol{S_{xx}}$ is a product of real and imaginary components in auto-spectrum

As stated earlier, the main purpose of this analysis was to measure the input/output relationship of the two systems and describe the dynamic behavior. This applies only if there is no noise contamination of the signal and direct relationship between output and input exists. To verify that a linear relationship exists, the coherence parameter was calculated.

Coherence expresses a degree of linearity between two signals and is defined as:

$$\gamma^2 = \frac{\left|S_{yx}\right|^2}{S_{xx} \cdot S_{yy}} \tag{5-15}$$

where:

$\boldsymbol{S_{yx}}$ is the product of linear spectrum of one signal and complex conjugate of the linear spectrum of another,

$\boldsymbol{S_{xx}}$ is the product of real and imaginary components in the auto-spectrum,

$\boldsymbol{S_{yy}}$ is the product of real and imaginary components in the auto-spectrum

A coherence value ranges between 0 and 1 where the value of unity indicates an ideal system and measurement conditions. If the value is less than one, which is typically due to the presence of noise, the quality of the frequency response function is affected. Figure 5.1 demonstrates a valid estimate of the frequency response function with respect to the associated frequency range.

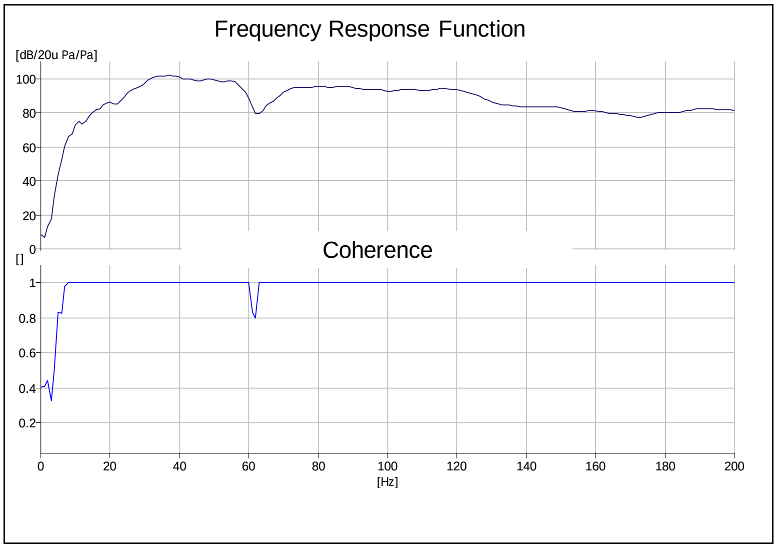


Figure 5.1 – Example of Valid Estimate of System's Frequency Response Function

5.3 <u>Total Harmonic Distortion (THD)</u>

In addition to frequency response, linearity is another valuable parameter for the overall acoustic characterization of a subwoofer system. Most real life systems demonstrate linear characteristics within a certain range of input level, but once that level is exceeded, other spurious frequencies different from the ones applied at the input appear at the output of the system. One method of evaluating these frequencies is to obtain the harmonic distortion values and calculate the total harmonic distortion percentage. The total harmonic distortion percentage can be calculated as follows:

$$\%THD = 100 \cdot \sqrt{\frac{(x_2^2 + x_3^2 + \ldots x_n^2)}{(x_1^2 + x_2^2 + x_3^2 \ldots x_n^2)}} \tag{5-16}$$

where:

$\boldsymbol{X_j}$ represents detected level response at its distortion order

In order to obtain THD values, one tone is used as an excitation signal and the frequencies measured are integer multiples of the excitation frequency. The total harmonic distortion was calculated for both of the subwoofer systems in order to evaluate and compare the non-linear behaviour of the two systems.

5.4 Sound Quality and Loudness

Sound quality is an analysis method used in this study to quantify the qualitative characteristics of the subwoofer speakers. This analysis employs different sound quality metrics to correlate the perceptual characteristics of sound to the physical quantities that can be measured and categorized. It helps to identify if a sound is pleasant or unpleasant to humans. As a subjective estimate of sound perception, loudness is an important sound quality metric which is part of the analysis. Loudness is the only sound quality measure used in this study, since the other common sound quality metrics including sharpness and roughness, do not have a significant association with this type of sound source. A brief historical background and theoretical explanation of loudness has already been given in Chapter 2 of this thesis.

Since loudness accounts for temporal and masking effects and thus being frequency dependent, the same analysis criteria as for FFT analysis must be applied in order to compute loudness. Loudness at each ear is obtained by the following equations:

$$L_N = 40 + 10\log_2 N \quad \text{for } N \geq 1 \text{ sone} \tag{5.17}$$

$$L_N = 40 \cdot (N + 0.0005)^{0.35} \quad \text{for } N < 1 \text{ sone} \tag{5.18}$$

The software then averages loudness at each ear to obtain binaural loudness value.

5.5 Subjective Evaluation and Paired Analysis

A subjective evaluation was conducted to verify the results found by the objective analysis and to better understand how potential customers would rate the sound of the two different subwoofer systems. Some theory about the subjective evaluation and analysis details of the paired comparison has been previously given in Chapter 2. For the subjective analysis, a music recording with a strong bass content was played through the car audio system for approximately 30 seconds. Two arguably different sounds were produced, corresponding to the different subwoofer systems, and played in pairs for evaluation. A paired comparison was used in which the jurors were asked to select their preferred sound. Two pairs were generated with one being in the reverse order from the other one. This allowed for the results of each juror to be checked for consistency. Psycho Acoustic Test Bench (BZ 5301), a tool of the Sound Quality Type 7698 module was used to collect and analyze the scores. The scores were then automatically stored in an Excel sheet which computed the correlation between objective and subjective values via a linear regression method. As a result, the predicted values were compared against the actual.

CHAPTER VI

RESULTS AND DISCUSSION

The following chapter is a summary of the results for this study including the single and dual channel frequency response, total harmonic distortion, loudness and the subjective evaluation for both subwoofer systems. The prototype glass subwoofer system is acoustically characterized and compared to an upgraded factory installed conventional subwoofer system. The following discussion begins with a background noise evaluation and discusses the data repeatability results. The output response of the two systems is discussed next followed by the dual channel frequency response where both output and input signals are evaluated simultaneously. The discussion continues with the total harmonic distortion results, after which the binaural loudness for both subwoofer systems is quantified and correlated with the subjective evaluation results.

Figure 6.1 compares the results obtained for pseudo random signal used to excite both subwoofer systems relative to the background noise during the measurements. It is demonstrated in this graph that background noise does not have a significant influence on the measured signal since noise within the vehicle is at least 15 dB lower than the lowest SPL for the measured signal throughout the frequency range of interest. Similar results are shown in Figure 6.2 where a swept sine signal is used to excite the subwoofers. A greater difference of approximately 25 Hz between the measured signal's lowest sound pressure level and highest peak of background noise is obtained throughout the frequency range of interest. These results demonstrate sufficiently low background noise for the objective and subjective acoustic evaluation of the two subwoofer systems.

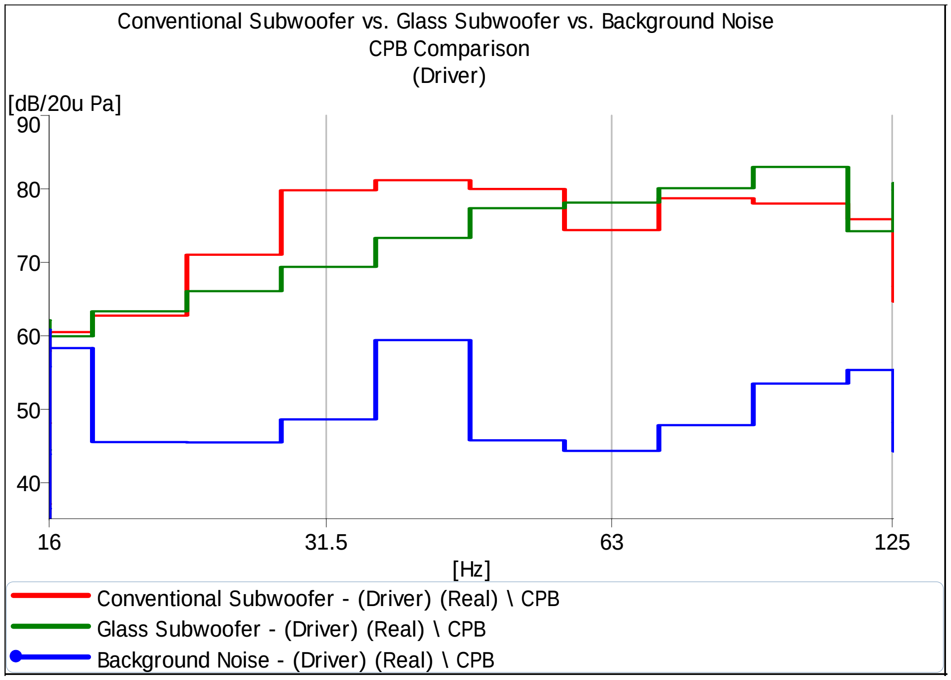


Figure 6.1 – Background Noise vs. Pseudo Random Signal - 3rd Octave CPB Comparison

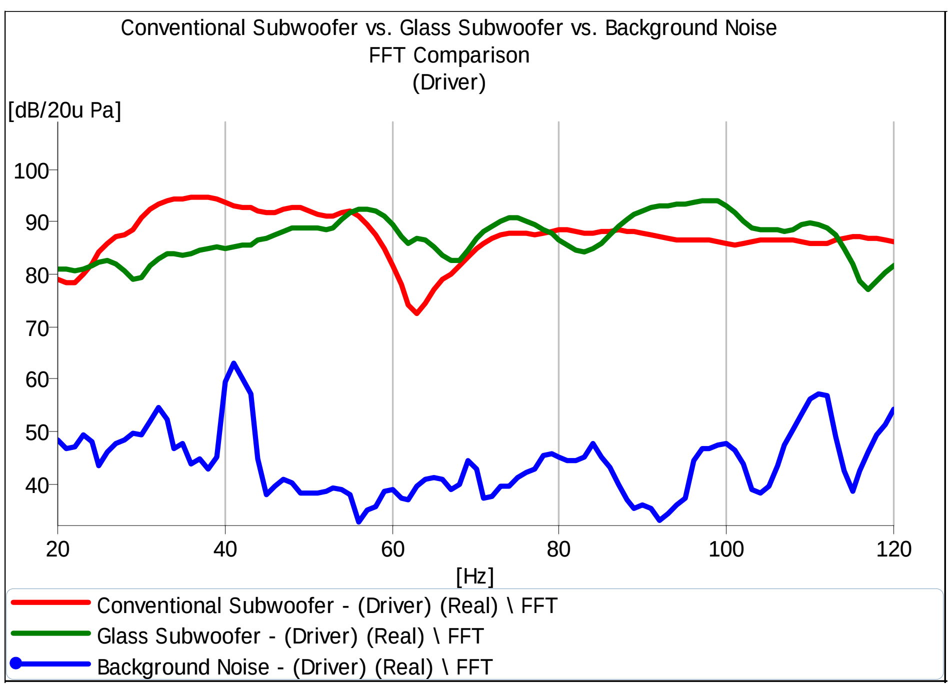


Figure 6.2 – Background Noise vs. Swept Sine Signal - FFT Comparison

As mentioned in the experimental set-up section, each set of measurements was repeated three times to verify adequate measurement repeatability. Figure 6.3 below illustrates three test runs for the conventional subwoofer system excited by a pseudo random signal. Results for all four measurement locations in the vehicle are computed simultaneously and they clearly demonstrate a high level of data repeatability. Minor differences are observed at the lowest frequencies of interest, approximately between 20 Hz and 25 Hz, for the second run relative to the first and third run. As such, the first run data is selected for the further discussion of results.

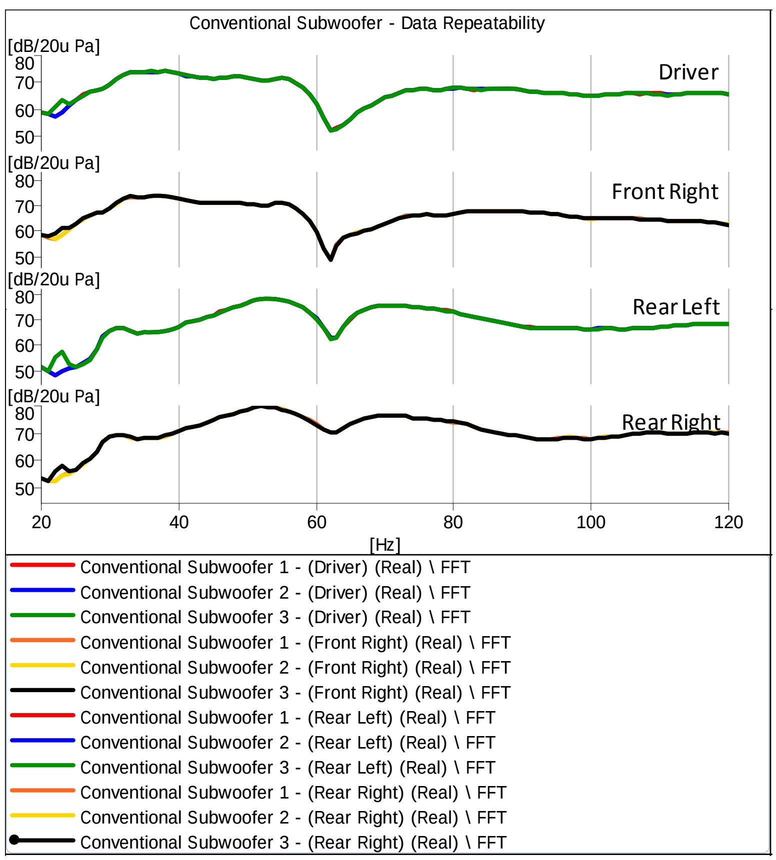


Figure 6.3 – Data Repeatability for Conventional Subwoofer System – Pseudo Random Signal - FFT Comparison

Similar results are obtained for the glass subwoofer system test runs as shown in figure 6.4. Due to similarities to the other two test runs, the first run is selected as a primary collection of data for further discussion of results. This demonstrates the repeatability of the data collection and concludes reliable and repeatable results. Data repeatability for the swept sine measurements can be found in Appendix A.

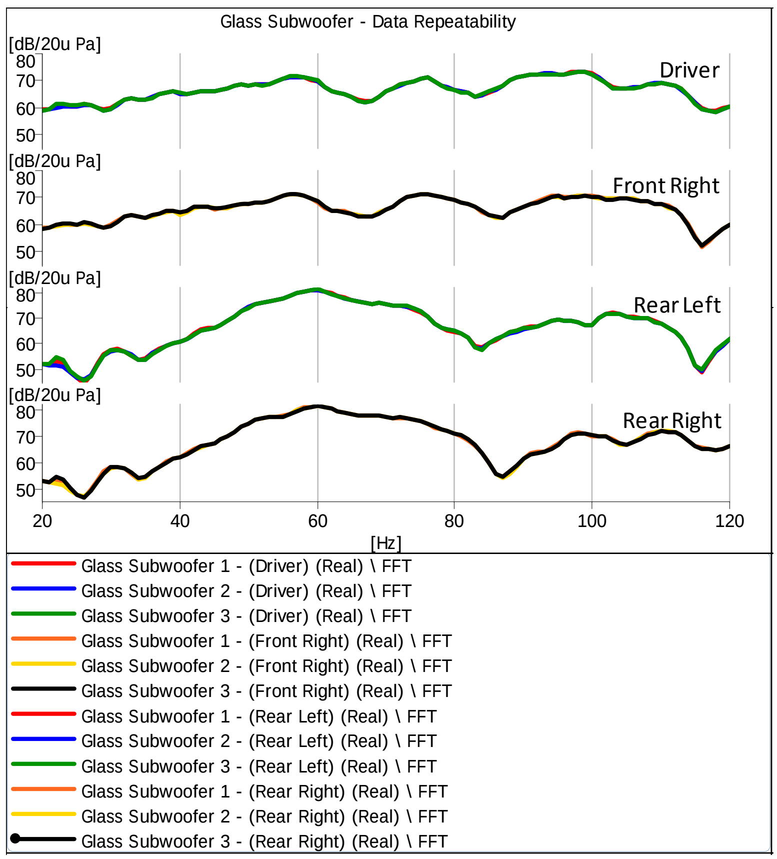


Figure 6.4 – Data Repeatability for Glass Subwoofer System – Pseudo Random Signal - FFT Comparison

It is important to acknowledge in this study that a subwoofer is an omni-directional source with a path of sound propagation which is uniform 360 degrees. Having said that, most of the frequency response differences between the left and right side measuring locations in the vehicle are due to the room acoustic of the cabin. More obvious differences are seen between the measurements taken at the front measuring locations and those taken at the rear measuring locations. Much higher sound pressure levels are experienced at the rear locations which are the result of the shorter distance between the sound source and a receiver. Discussion of the results concentrates at the driver measurement location and compares the acoustic characteristics of both subwoofer systems at that location. Results for the remaining three measurement locations can be found in Appendix A.

6.1 Single Channel Frequency Response Results

A comparison of the measured CPB spectra revealed the output characteristics of the two subwoofer systems. The amplifier gain was set to obtain the same sound pressure level for both the conventional and glass subwoofer systems. Even so, it can be observed from Figure 6.5 that the amplitudes are different for the two systems at corresponding frequencies throughout the frequency range. Higher sound pressure levels, as much as 10 dB, are found at the 31.5 Hz and 40 Hz frequency bands which indicate more dominant performance of the conventional subwoofer over the glass subwoofer system. At the 63 Hz and 100 Hz frequency bands, the glass subwoofer system becomes more dominant and overcomes the sound pressure level of the conventional subwoofer system with a difference in amplitude of up to 8 dB.

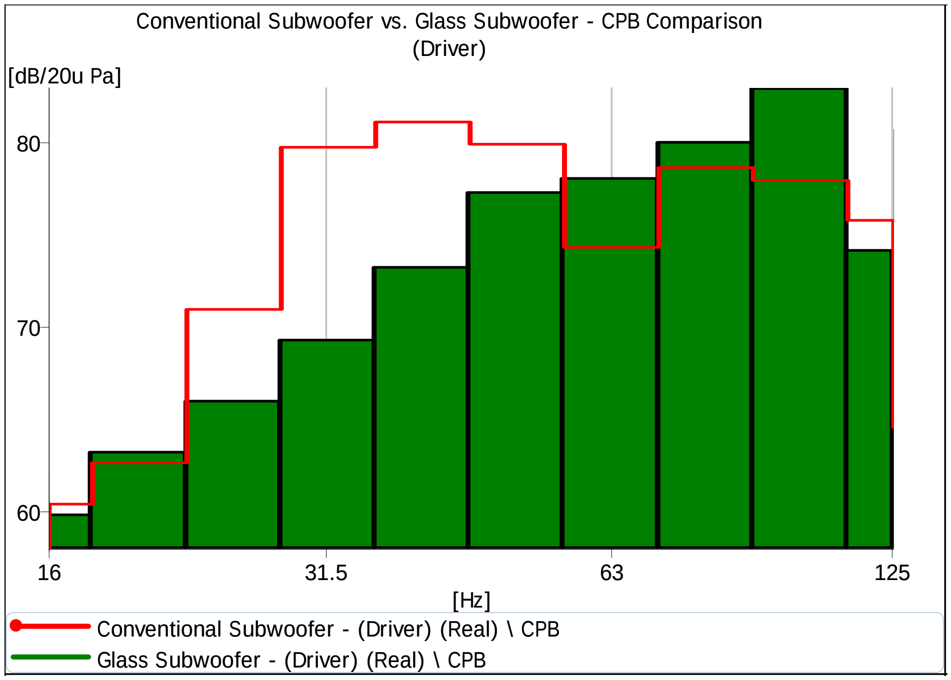


Figure 6.5 – Single Channel Frequency Response – Pseudo Random Signal – 3rd Octave CPB Comparison

Referring to the output response, and considering that the frequency region where popular music has most of its bass energy is between 60 Hz and 125 Hz [42], the glass subwoofer system appears to be preferred over the conventional one.

6.2 Dual Channel Frequency Response Results

A dual channel frequency response was undertaken as a more realistic approach to compare the sound characteristics of the two subwoofer systems. The output signal was referenced to its original input and simultaneous measurements at the input and output are performed, revealing slight deviations from the true flat response. Nevertheless, both systems demonstrate a reasonably flat frequency response with gentle variations in amplitude. For this study, a flat response is not necessarily expected for a real system as it is measured at the driver's ear and not in a free field directly in front of the loudspeaker which is the approach normally used for the measurement of loudspeaker specifications.

Instead, the real response is influenced by the automotive interior acoustics from the source of the sound to the receiver at the driver's ear. The seating arrangement, interior materials and location of the listener all affect the absorption and transmission loss characteristics of the perceived sound. The frequency response for both subwoofer systems was within approximately ± 8 dB from 20 Hz to 120 Hz as shown in Figure 6.6, if the anti-node at 63 Hz is excluded. As indicated by the dip in the coherence function, it is evident in Figure 6.6 that a sharp anti-resonance in frequency response corresponding to the conventional subwoofer system appears at 63 Hz. Studies have shown that these rapid changes in the amplitude tend to produce a sound that is more fatiguing, less pleasing, and subjectively less accurate [43].

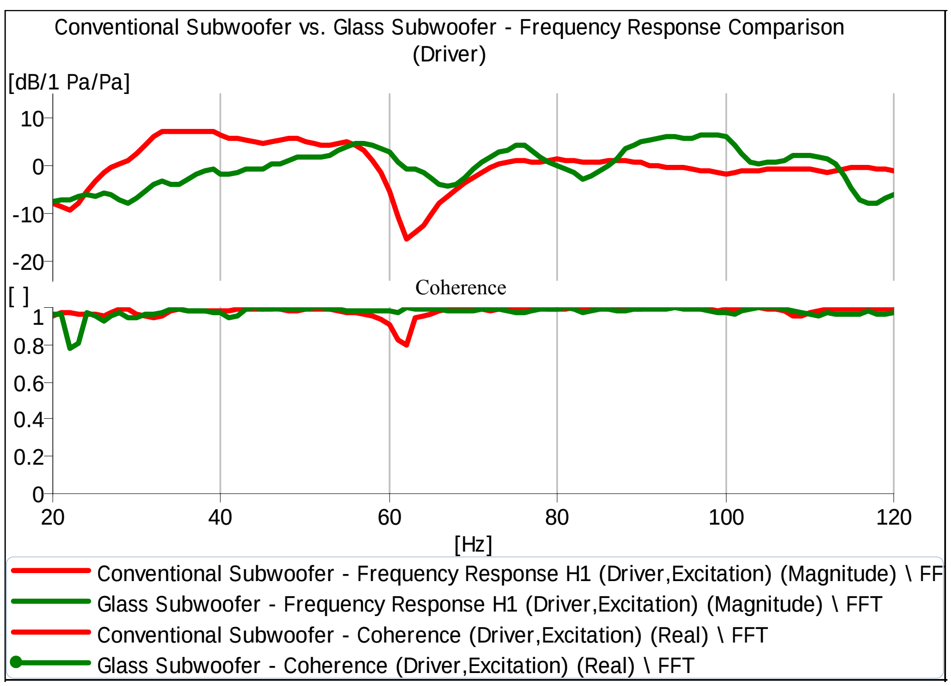


Figure 6.6 – Dual Channel Frequency Response – Pseudo Random Signal – FFT Comparison

An attempt was made to verify the response results for the two subwoofer systems by using a different excitation signal; in this case a swept sine signal instead of the pseudo random signal used in the previous analysis. Although the coherence between the swept sine input and the output signals is improved, the frequency response still lies approximately within ± 8 dB, within the range of 20 Hz to 120 Hz as shown in Figure 6.7, if the anti-node at 63 Hz is excluded. This implies that the dual channel frequency response of the two subwoofer systems is independent of the excitation signal being used, whether the signals are stationary or non-stationary.

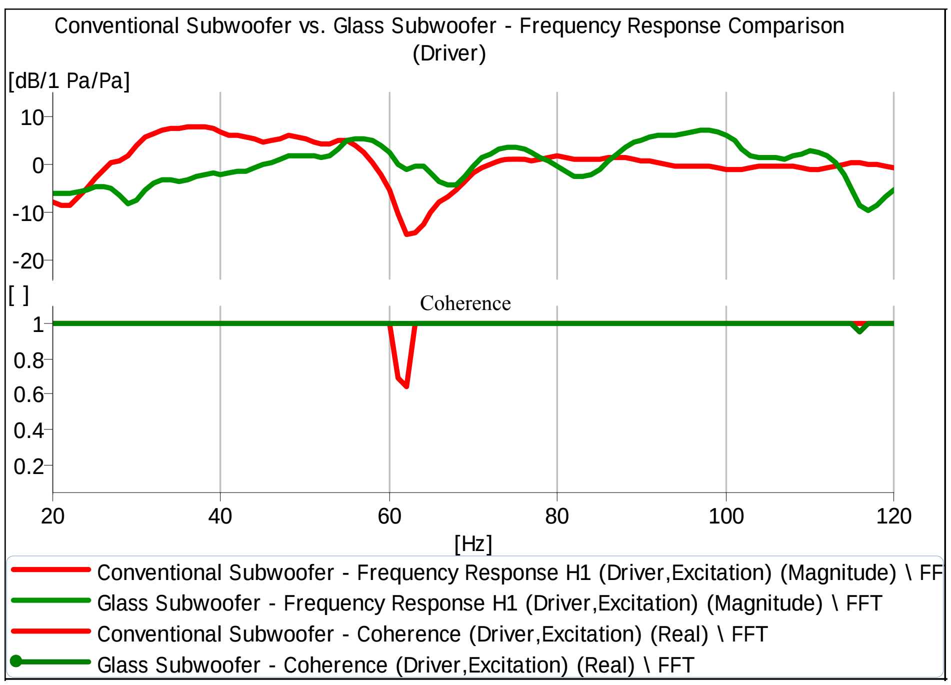


Figure 6.7 – Dual Channel Frequency Response – Swept Sine Signal – FFT Comparison

6.3 <u>Rear Glass Vibration Measurements and Results</u>

In order to investigate the glass subwoofer's differences in frequency response between the left and right side measuring locations, 12 accelerometers were mounted on the outer surface of the rear glass as shown in Figure 6.8.

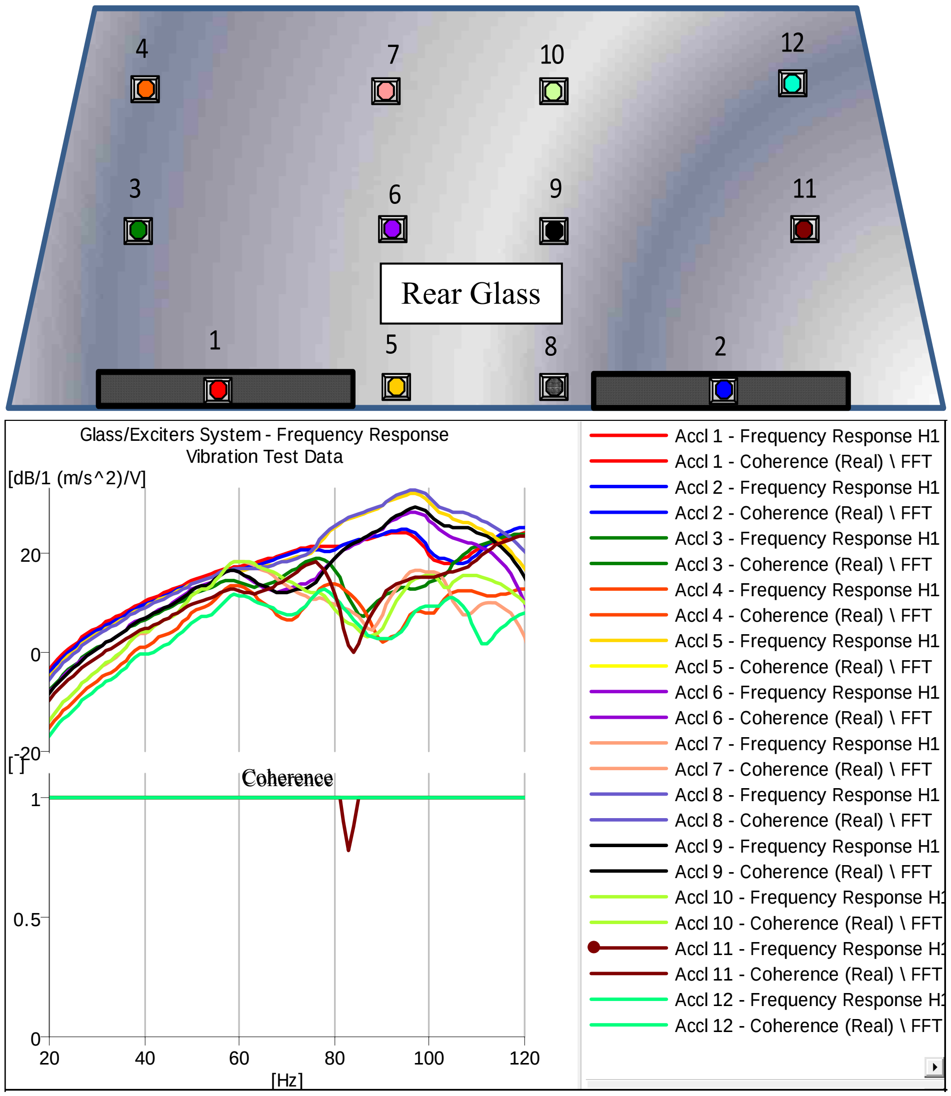


Figure 6.8 – Rear Glass Vibration Measurements – Swept Sine Signal – FFT Comparison

One can see a similar frequency response in the frequency range between 20 Hz and 60 Hz. Beyond that region, differences in the response start to become more prominent. The responses at the accelerometer locations 1 and 2 are quite similar which demonstrates that the two piezoelectric actuators are contributing equally. Upon further examination, one can notice an asymmetrical excitation of the rear glass. As the analysis moves from the bottom edge to accelerometer locations 5 and 8, toward the mid section of the rear glass, accelerometer locations 6 and 9, differences in the frequency response become quite obvious. Similar behaviour is shown from the mid section, accelerometer locations 6 and 9, towards the upper edge, accelerometer locations 7 and 10, of the rear glass. This emphasizes that in practice glass does not behave as a rigid body but rather demonstrates elastic characteristics, which are not ideally desired. If the analysis is conducted from the left edge, accelerometer locations 3 and 4, towards the mid section, accelerometer locations 6, 9, 7 and 10, of the rear glass, once again the differences in the response can be observed. This becomes more prominent as the analysis continues towards the right edge, accelerometer locations 11 and 12, of the rear glass. It can be concluded that the glass subwoofer system, although being considered as omni directional, still has a slight contribution to unsymmetrical frequency response.

6.4 Total Harmonic Distortion Results

It is well known that high level, low frequencies caused unwanted vibration in the vehicle's trim and body closures and greatly contribute to the sound distortions in the vehicle [16]. Studies have shown that a total harmonic distortion of more than 1% is audible by human hearing. In the case of the subwoofer's low frequency nonlinear distortions, the sensitivity threshold of human hearing increases to approximately 5% for real signals. However, it was previously stated that a typical automotive subwoofer system commonly produces up to 10% total harmonic distortion [20]. Similar numbers are demonstrated in this analysis. Results in Figure 6.9 and Figure 6.10 reveal the total harmonic distortion for both subwoofer systems. It is noticed that the conventional subwoofer system exhibits a higher peak at approximately 25 Hz but lower levels of distortion than the glass subwoofer system throughout the rest of the frequency range. Although this fact might be due to the challenging implementation of the glass subwoofer system design, further development is required to reduce the excessive distortion level of the glass subwoofer system.

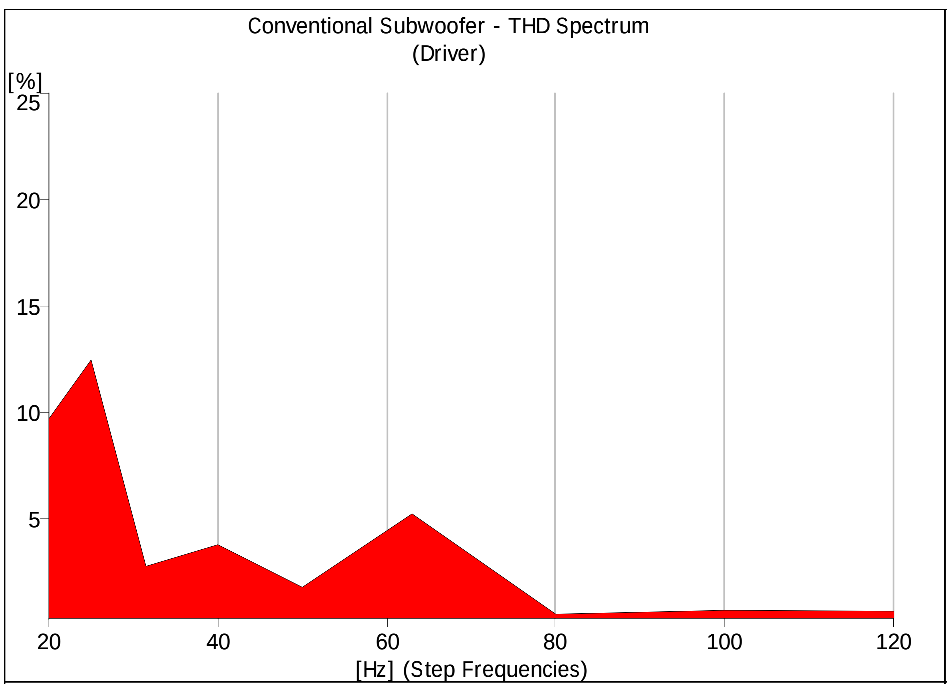


Figure 6.9 – Conventional Subwoofer System - THD

Figure 6.10 – Glass Subwoofer System - THD

6.5 <u>Binaural Loudness and Subjective Evaluation Results</u>

The outcome of the subjective analysis of the two systems is expected to determine whether the differences between the two systems observed in the objective analysis methods are significant in the perception of quality of sound. When the specific loudness values of Figure 6.11 and Figure 6.12 are calculated into total loudness, the objective results illustrate an arguable difference in loudness between the two subwoofer systems of approximately 4.6 sones. However, the paired comparison subjective analysis resulted in 15 individuals with a preference for the conventional subwoofer system, while the other 12 individuals indicated a preference for the glass subwoofer system. This demonstrates almost a split preference between the systems. Using a linear regression analysis, an attempt was made to correlate these results with loudness measurements shown in Figure 6.11 and Figure 6.12.

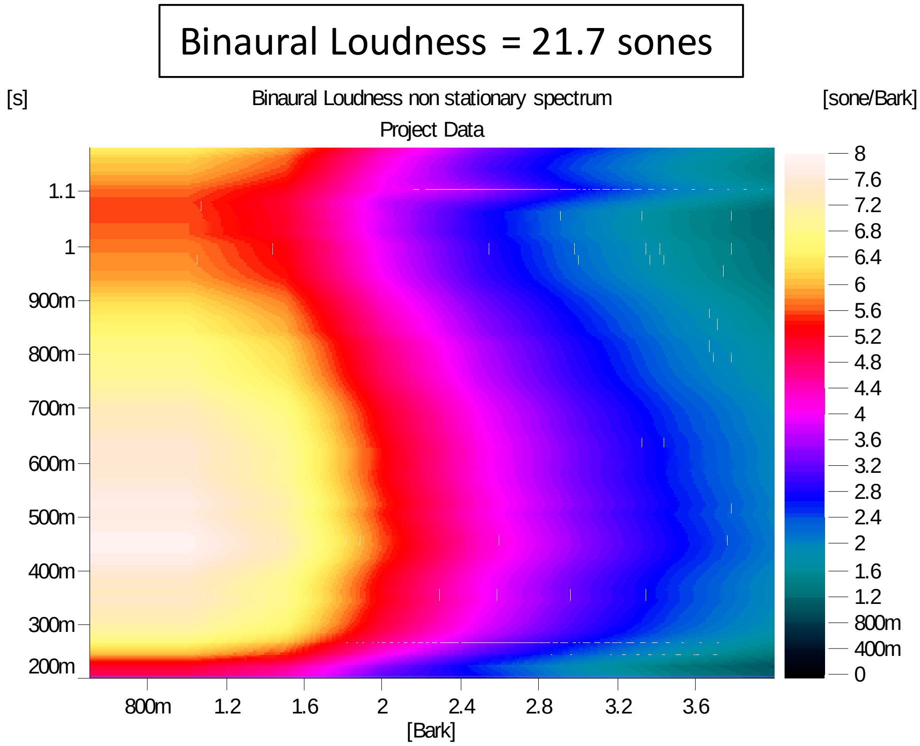


Figure 6.11 – Conventional Subwoofer System – Specific Loudness

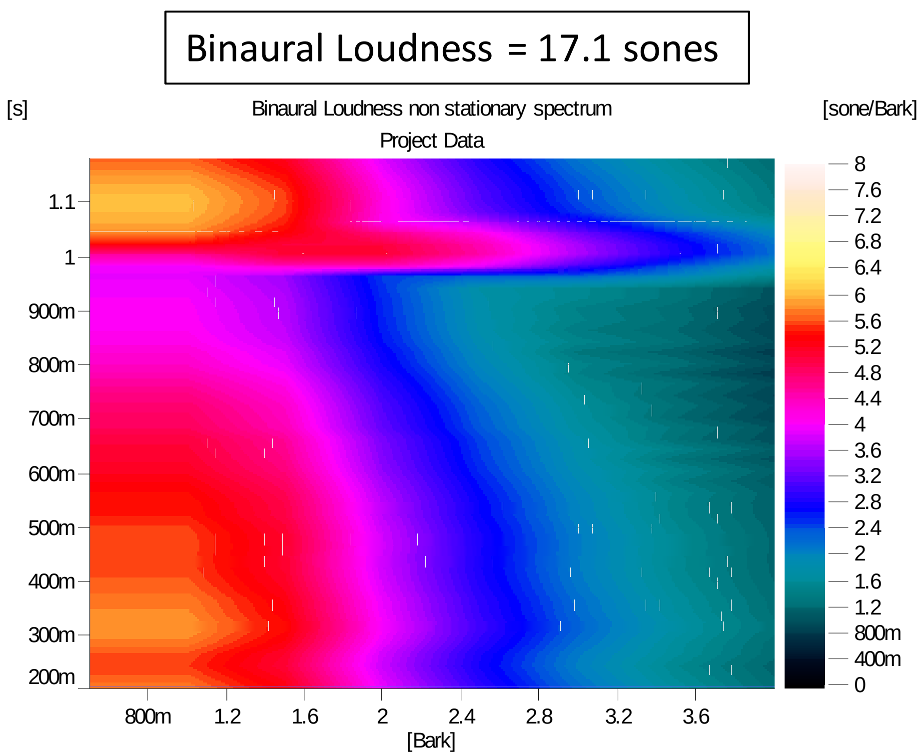


Figure 6.12 – Glass Subwoofer System – Specific Loudness

Figure 6.13 shows a plot of actual versus predicted preferences for both subwoofer systems. The difference between the estimated and measured subjective results indicates that loudness most likely does not completely illustrate all aspects of dissimilarities between the perceived sound quality of the two systems. This can be explained by knowing that the human hearing mechanism is less sensitive to sound level differences at extremes of the hearing range (i.e. 20 Hz) [44]. It might be more prudent to refer to some of the comments provided by jurors. It was stated by many that the conventional subwoofer sounded deeper and less distorted, but also less controlled and less defined relative to the sound of glass subwoofer system. One juror commented that the conventional subwoofer system was similar to buffeting noise whereas the glass subwoofer system felt more like a tingle in the back and bottom of the seat. Although it is possible that the two systems demonstrated similarities to a certain extent, the subjective preferences clearly stated that acoustic differences do exist and that they are audible to the listener.

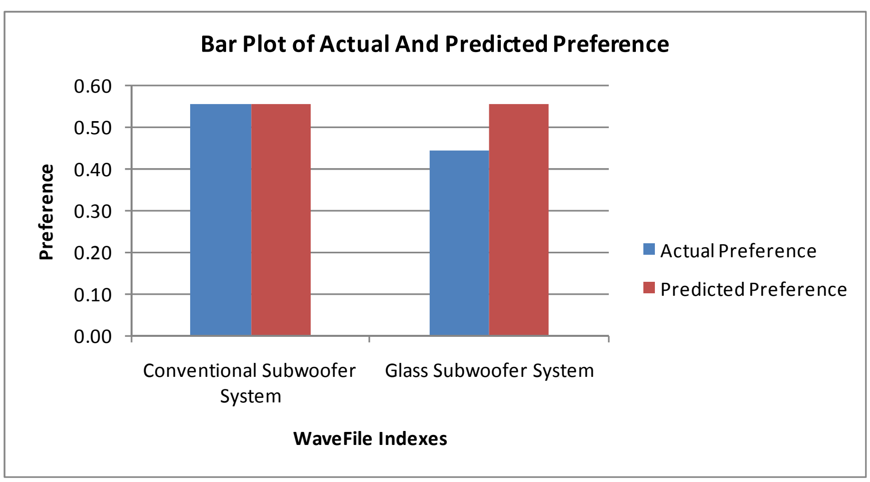


Figure 6.13 – Subjective Test Results – Actual vs. Predicated Preference

CHAPTER VII

CONCLUSIONS AND RECOMMENDATIONS

This chapter summarizes the conclusions, recommendations and contributions to the engineering science and knowledge obtained from this study.

7.1 Conclusions

After a detailed analysis of the experimental results and referring to the objectives stated in the introductory part of this work, the following conclusions are derived:

- The impact of the glass subwoofer system on the vehicle interior sound quality has been investigated. The effect and the contribution of the glass subwoofer to the vehicle's audio system sound quality are evaluated. Based on the physical properties, including the low weight, size and power consumption, as well as its acoustical characteristics, the glass subwoofer system is considered to be representative of a green technology. This technology demonstrates a great potential for a high quality audio system for hybrid and electric vehicles as well as gas vehicles where fuel consumption, interior space, and power usage are optimized for better performance and overall customer satisfaction.
- The rear glass subwoofer system is compared to the conventional subwoofer system and a relationship between objective measurements and subjective evaluations of both systems were obtained. Although the glass subwoofer system characterized in this study is a prototype, the overall results show that its acoustic characteristics are comparable to those of an upgraded conventional subwoofer system.

- Objective acoustic evaluation of the two subwoofer systems showed notable differences in performance. Both systems demonstrate reasonably flat frequency response with gentle variations in amplitude. However, frequency response graphs reveal differences in the amplitudes for the two subwoofer systems at corresponding frequencies throughout the frequency range. The total harmonic distortion performance is deteriorated for the glass subwoofer system with around 10% total harmonic distortion compared to around 5% total harmonic distortion of the conventional subwoofer system. The conventional subwoofer system exhibits higher loudness values throughout the frequency range of interest resulting in a 4.6 sones difference when compared to the glass subwoofer system.
- Subjective evaluations resulted in perceivable differences in sound quality between the two systems. In addition, out of 27 jurors, 15 individuals indicated a preference for the conventional subwoofer system while the other 12 individuals indicated a preference for the glass subwoofer system. This was an almost a split decision between the two subwoofer systems.
- A standardized testing procedure suitable for measuring and evaluation of the glass subwoofer system acoustic properties was developed and presented in Chapter 4 of this thesis.
- Appropriate testing environment, instrumentation and experimental techniques used to validate the acoustical characteristics of rear glass subwoofer system are recommended for future implementations.

7.2 Contributions to the Engineering Science and Knowledge

The contributions to the engineering science and knowledge obtained from this study include:

- The development of an objective acoustic evaluation method suitable for a rear glass subwoofer system, a new, alternative, green technology, as compared to the conventional automotive subwoofer system.
- The awareness of the significance of complementing the objective acoustic evaluation with subjective evaluation using human subjects to evaluate sound quality of a subwoofer system. Ultimately, the automotive customer's perception is the deciding factor in the final assessment of sound quality of any subwoofer system. It is not guaranteed that all aspects of this perception are necessarily captured using the currently available and common sound quality metrics as shown in this study.

7.3 Recommendations

The following recommendations provide suggestions for further investigation and improvement of the rear glass subwoofer system sound quality and its applications.

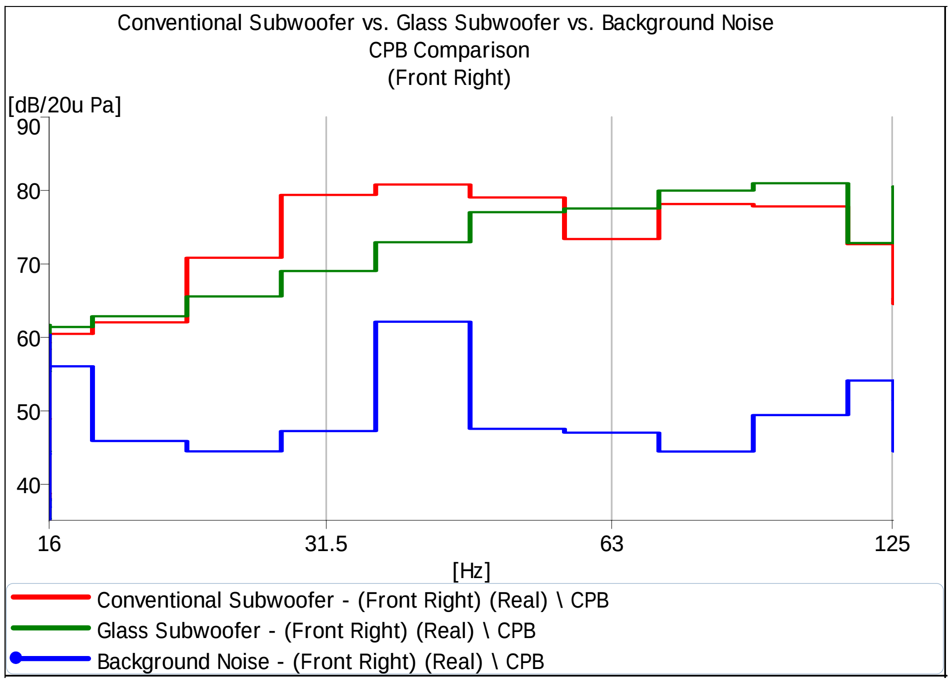
- Further investigation is necessary to determine factors influencing higher total harmonic distortion levels of the glass subwoofer system. Possibly, a modal analysis approach could be suitable for this type of investigation. This would require disengagement of the rear glass subwoofer system in order to achieve a fixed testing plane since the rear glass in the current set-up is free to move along three mounting edges on the vehicle frame.

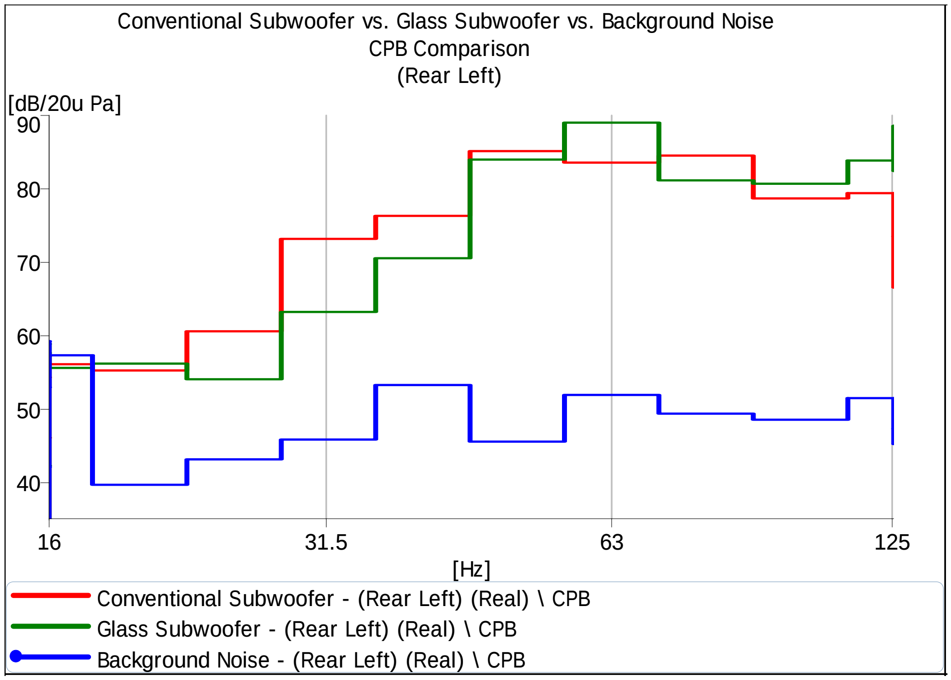
- Potential dissimilarity in the loudness values between the two systems is suspected to be due to the extended frequency range of the glass subwoofer system. Future research on this subject is necessary to make a more definitive conclusion.
- Ideally, a test fixture located in an anechoic room which allows for independent vibro-acoustic testing of rear glass subwoofer system is recommended. This would allow for traditional testing under free field conditions and true electroacoustics characteristics including frequency response and total harmonic distortion of the glass subwoofer system would be possible.
- In addition to the intended purpose of an audio subwoofer, the glass subwoofer system may be tuned to act as an active noise control mechanism to minimize or fully prevent the occurrence of automotive buffeting noise inside the vehicle.

APPENDICES

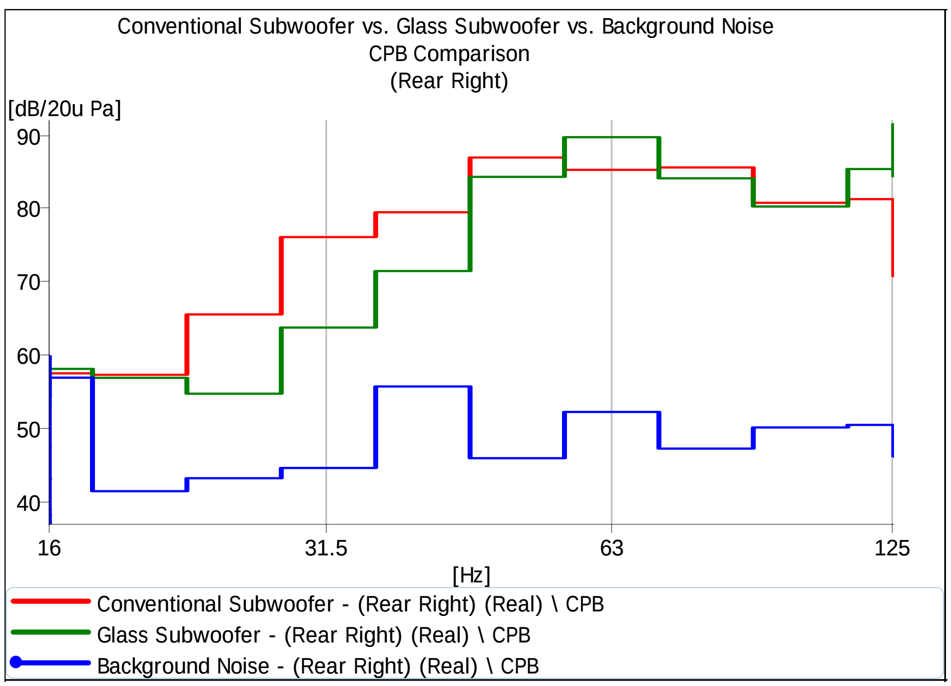
APPENDIX A – Experimental Results

EXHIBIT A1: Background Noise vs. Pseudo Random Signal



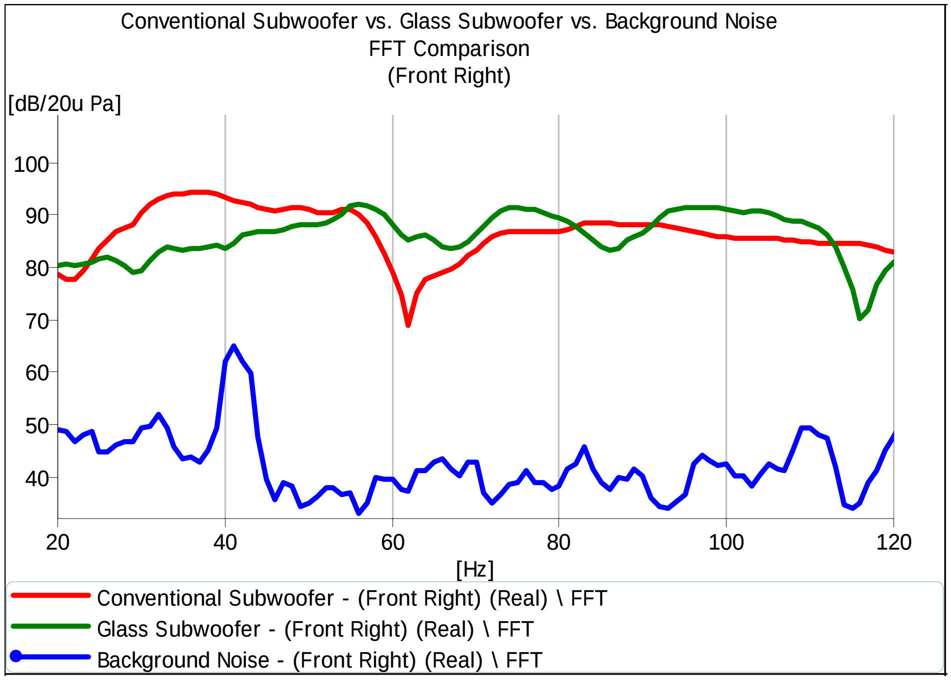


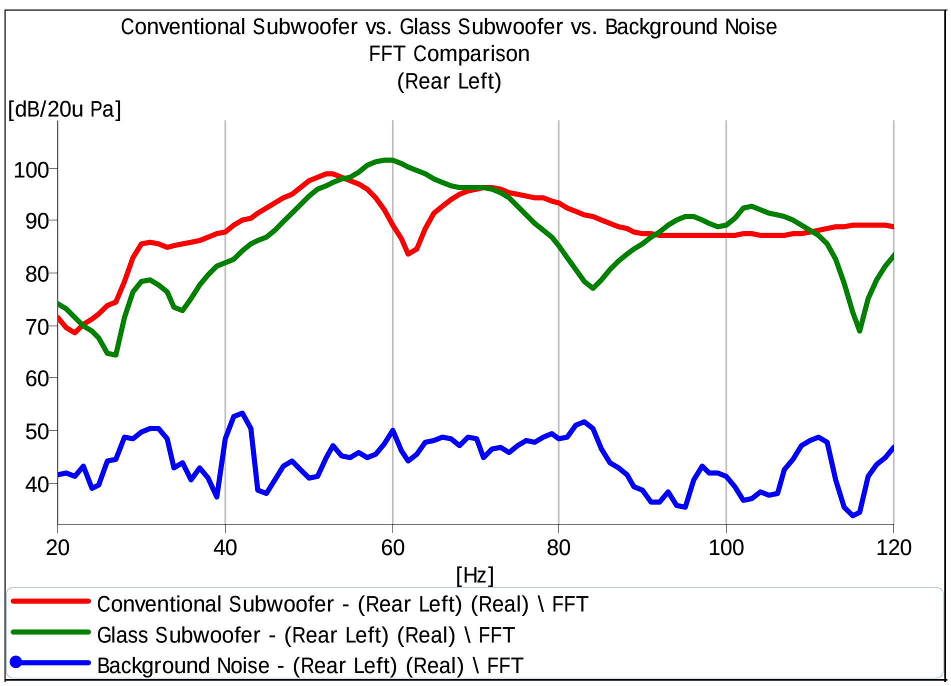
Conventional Subwoofer vs. Glass Subwoofer vs. Background Noise
CPB Comparison
(Rear Left)
[dB/20u Pa]
90
80
70
60
50
40
16
31.5
63
125
[Hz]
Conventional Subwoofer - (Rear Left) (Real) \ CPB
Glass Subwoofer - (Rear Left) (Real) \ CPB
Background Noise - (Rear Left) (Real) \ CPB



Conventional Subwoofer vs. Glass Subwoofer vs. Background Noise
CPB Comparison
(Rear Right)
[dB/20u Pa]
90
80
70
60
50
40
16
31.5
63
125
[Hz]
Conventional Subwoofer - (Rear Right) (Real) \ CPB
Glass Subwoofer - (Rear Right) (Real) \ CPB
Background Noise - (Rear Right) (Real) \ CPB

EXHIBIT A2: Background Noise vs. Swept Sine Signal





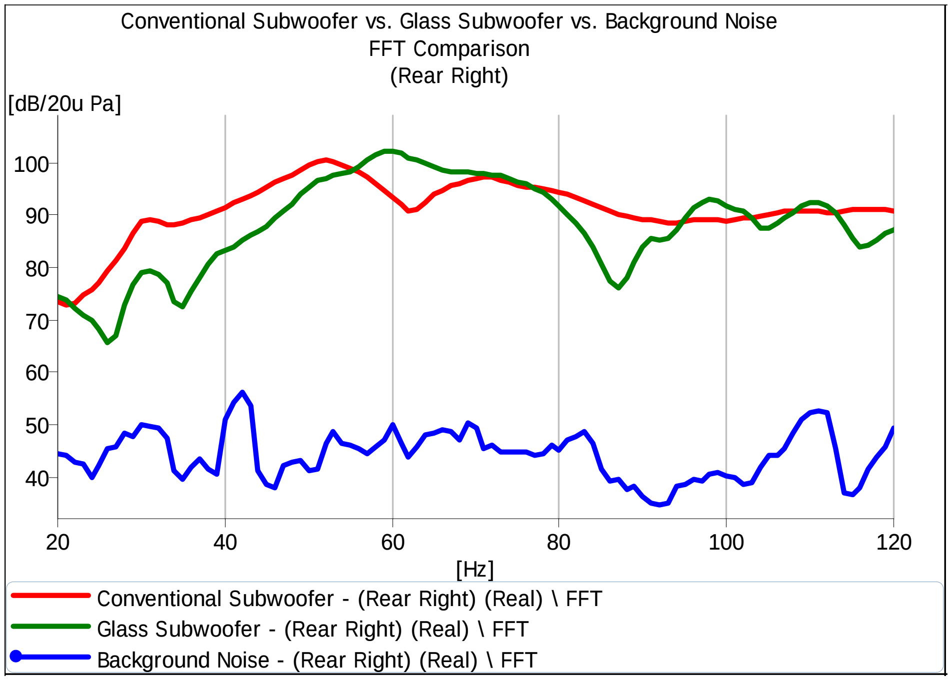
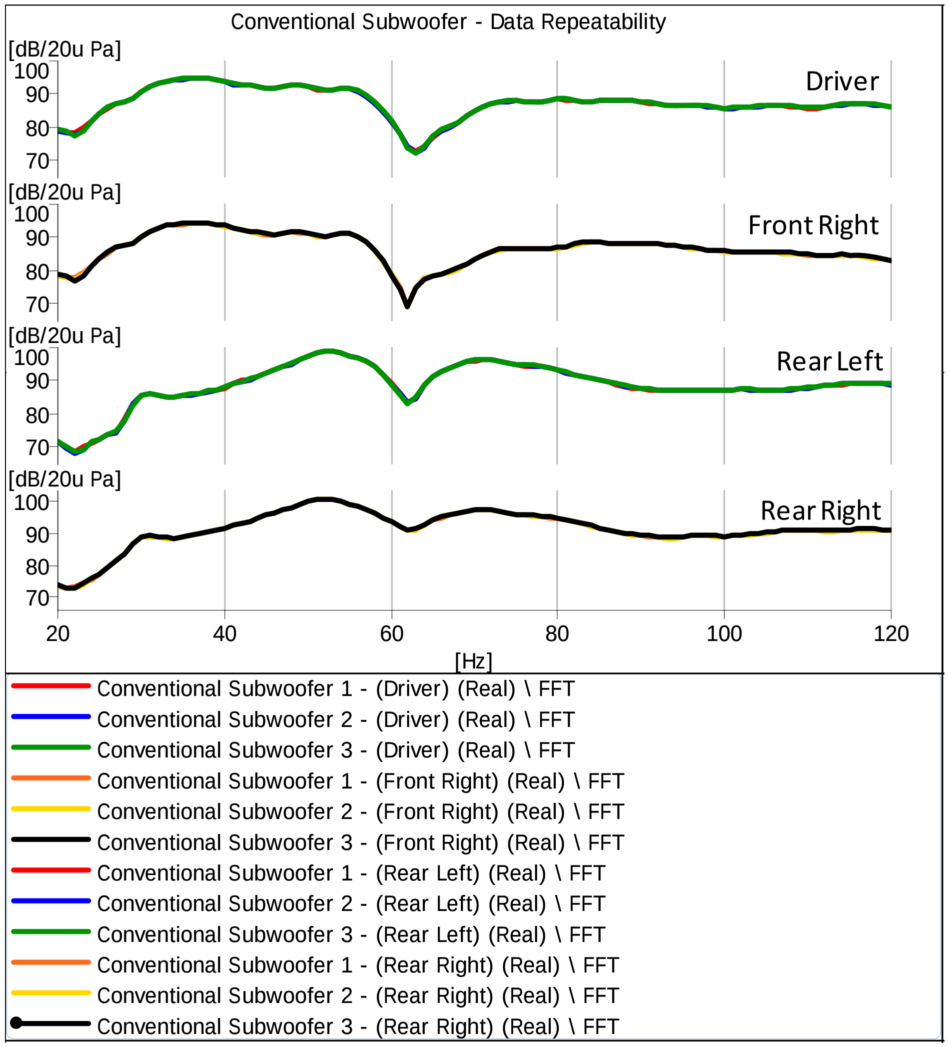
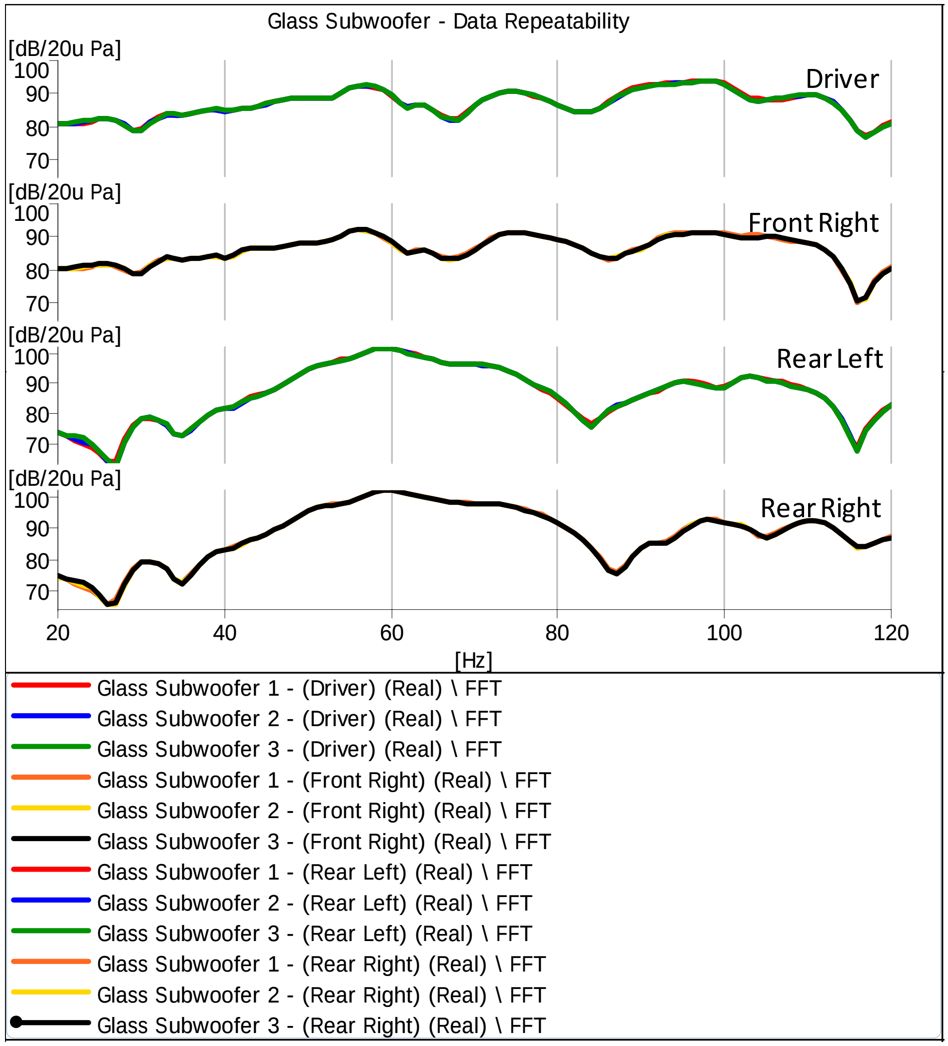

Conventional Subwoofer vs. Glass Subwoofer vs. Background Noise
FFT Comparison
(Rear Right)
[dB/20u Pa]
100
90
80
70
60
50
40
20
40
60
80
100
120
[Hz]
Conventional Subwoofer - (Rear Right) (Real) \ FFT
Glass Subwoofer - (Rear Right) (Real) \ FFT
Background Noise - (Rear Right) (Real) \ FFT

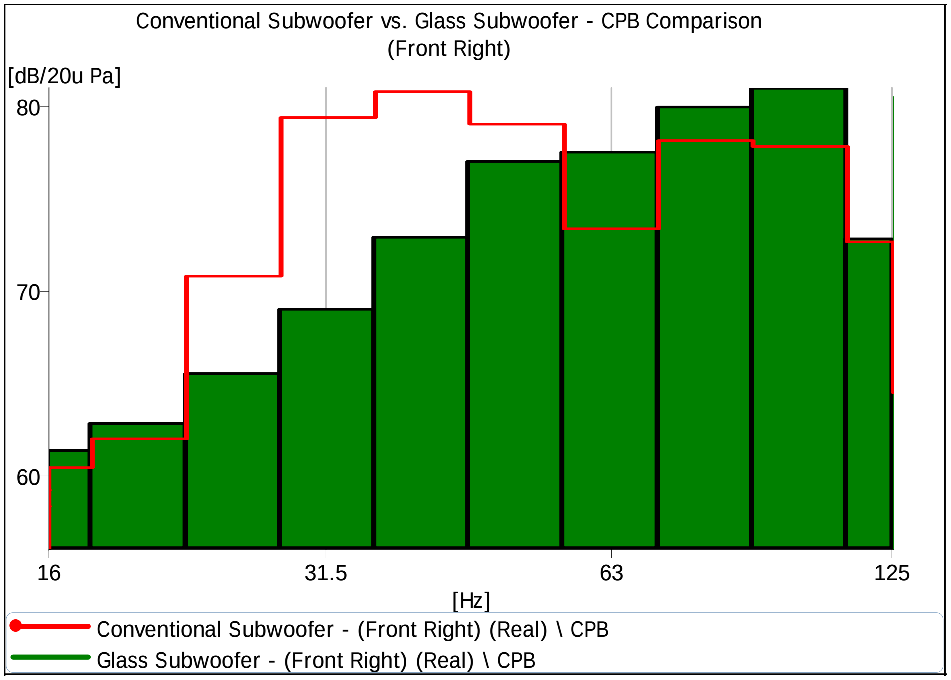
EXHIBIT A3: Data Repeatability – Swept Sine Signal

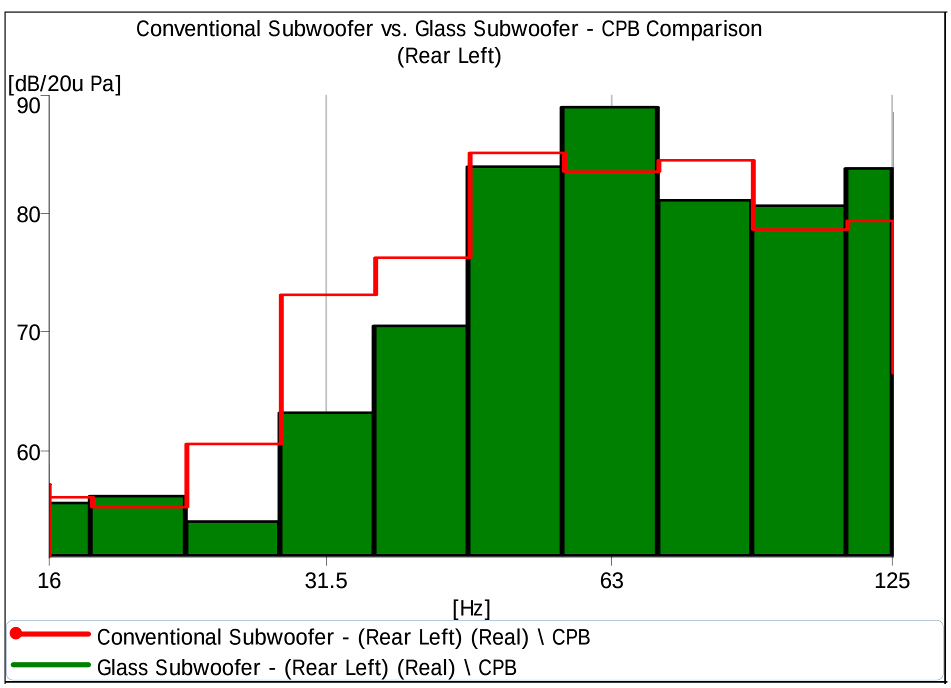




Glass Subwoofer - Data Repeatability
[dB/20u Pa]
100
90
80
70
Driver
[dB/20u Pa]
100
90
80
70
Front Right
[dB/20u Pa]
100
90
80
70
Rear Left
[dB/20u Pa]
100
90
80
70
Rear Right
20
40
60
80
100
120
[Hz]
Glass Subwoofer 1 - (Driver) (Real) \ FFT
Glass Subwoofer 2 - (Driver) (Real) \ FFT
Glass Subwoofer 3 - (Driver) (Real) \ FFT
Glass Subwoofer 1 - (Front Right) (Real) \ FFT
Glass Subwoofer 2 - (Front Right) (Real) \ FFT
Glass Subwoofer 3 - (Front Right) (Real) \ FFT
Glass Subwoofer 1 - (Rear Left) (Real) \ FFT
Glass Subwoofer 2 - (Rear Left) (Real) \ FFT
Glass Subwoofer 3 - (Rear Left) (Real) \ FFT
Glass Subwoofer 1 - (Rear Right) (Real) \ FFT
Glass Subwoofer 2 - (Rear Right) (Real) \ FFT
Glass Subwoofer 3 - (Rear Right) (Real) \ FFT

EXHIBIT A4: Single Channel Frequency Response – Pseudo Random Signal





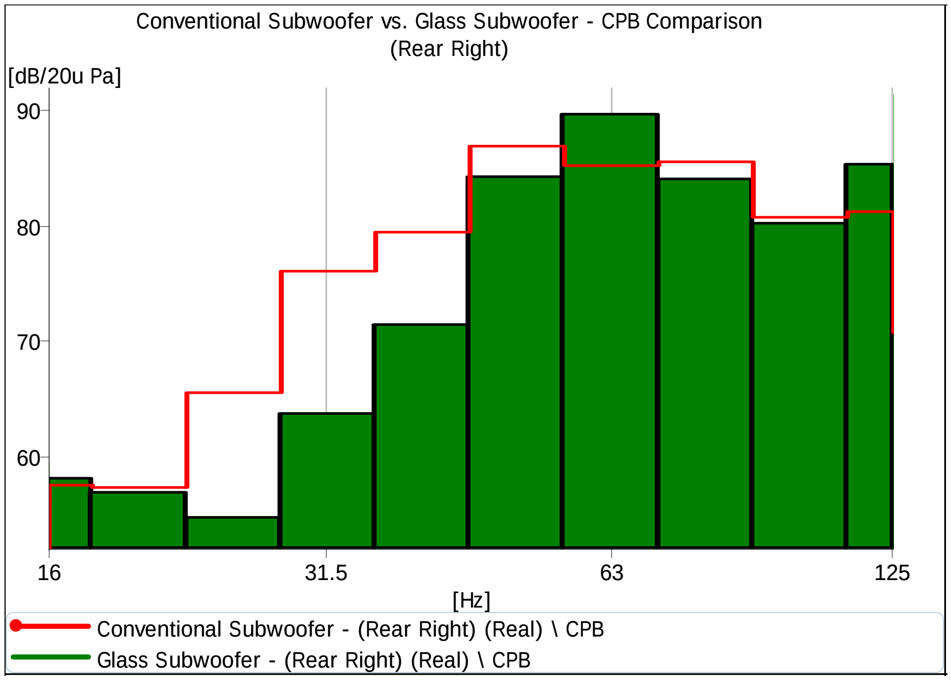
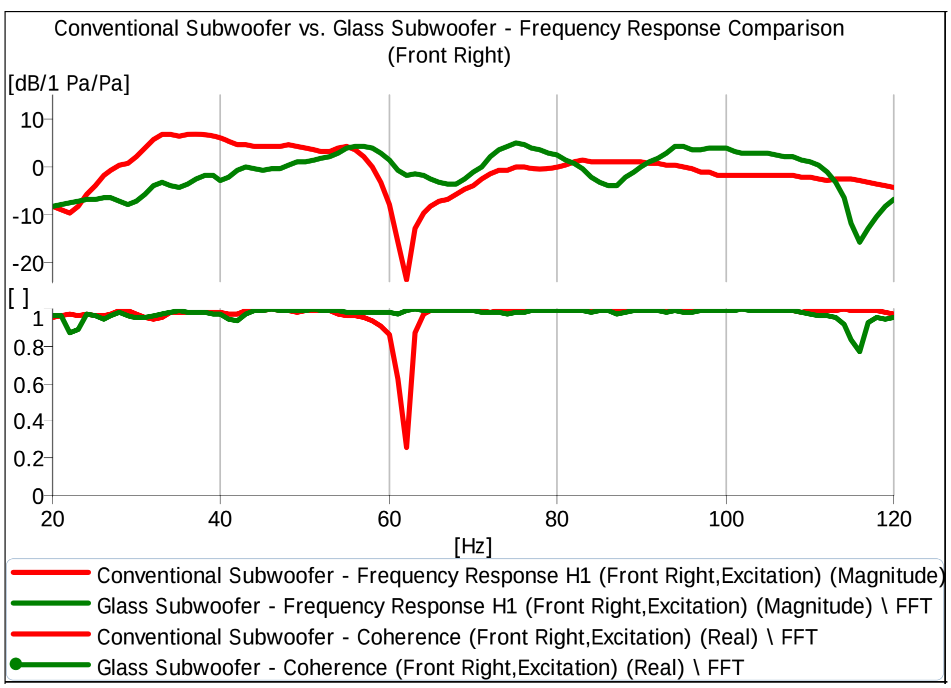
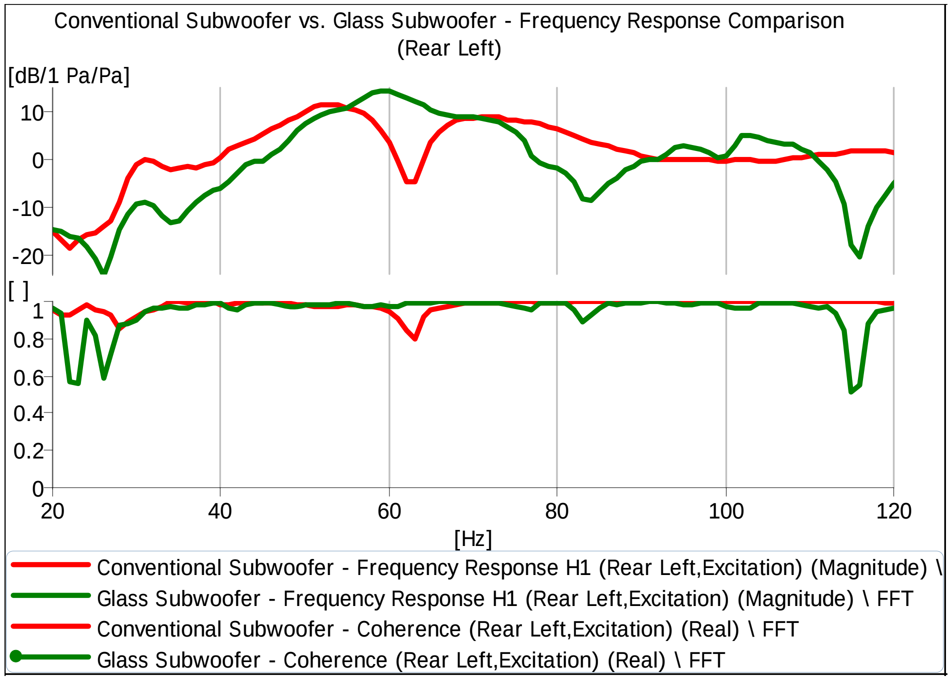


EXHIBIT A5: Dual Channel Frequency Response – Pseudo Random Signal




Conventional Subwoofer vs. Glass Subwoofer - Frequency Response Comparison
(Rear Left)
[dB/1 Pa/Pa]
10
0
-10
-20
[ ]
1
0.8
0.6
0.4
0.2
0
20
40
60
80
100
120
[Hz]
Conventional Subwoofer - Frequency Response H1 (Rear Left,Excitation) (Magnitude) \
Glass Subwoofer - Frequency Response H1 (Rear Left,Excitation) (Magnitude) \ FFT
Conventional Subwoofer - Coherence (Rear Left,Excitation) (Real) \ FFT
Glass Subwoofer - Coherence (Rear Left,Excitation) (Real) \ FFT

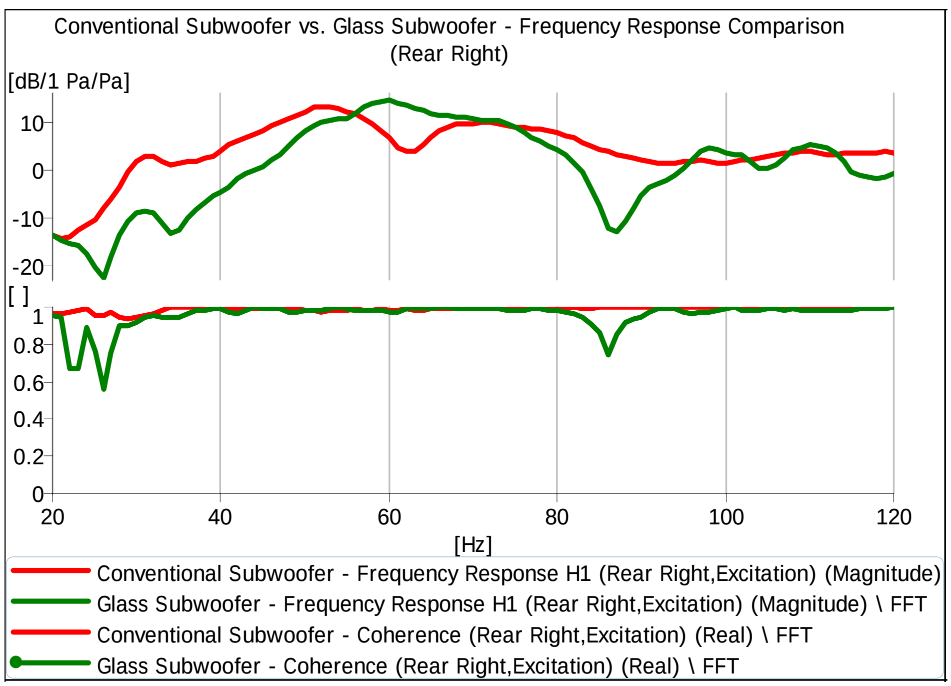
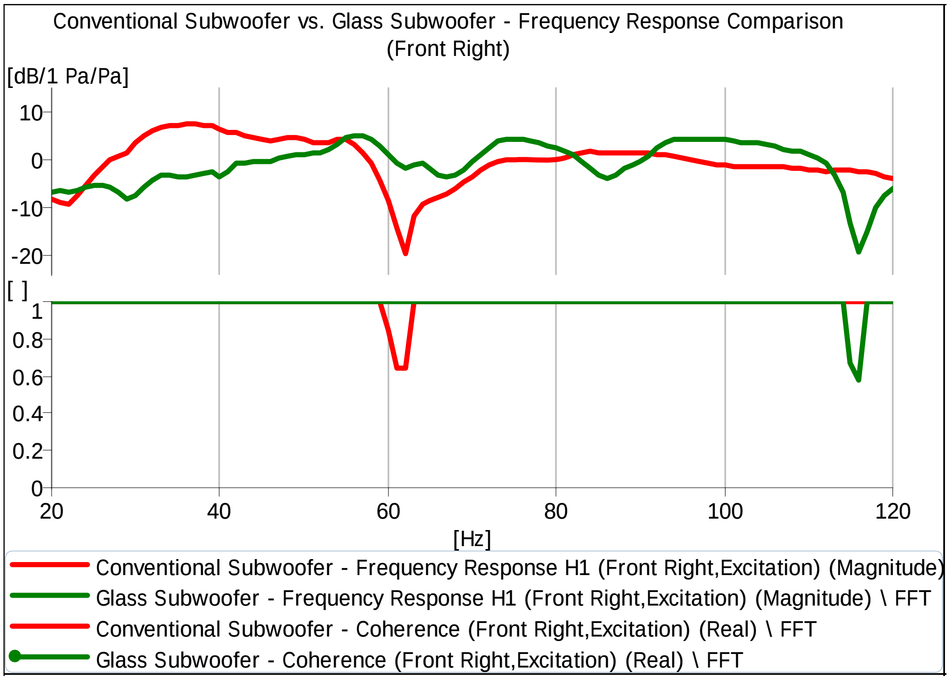
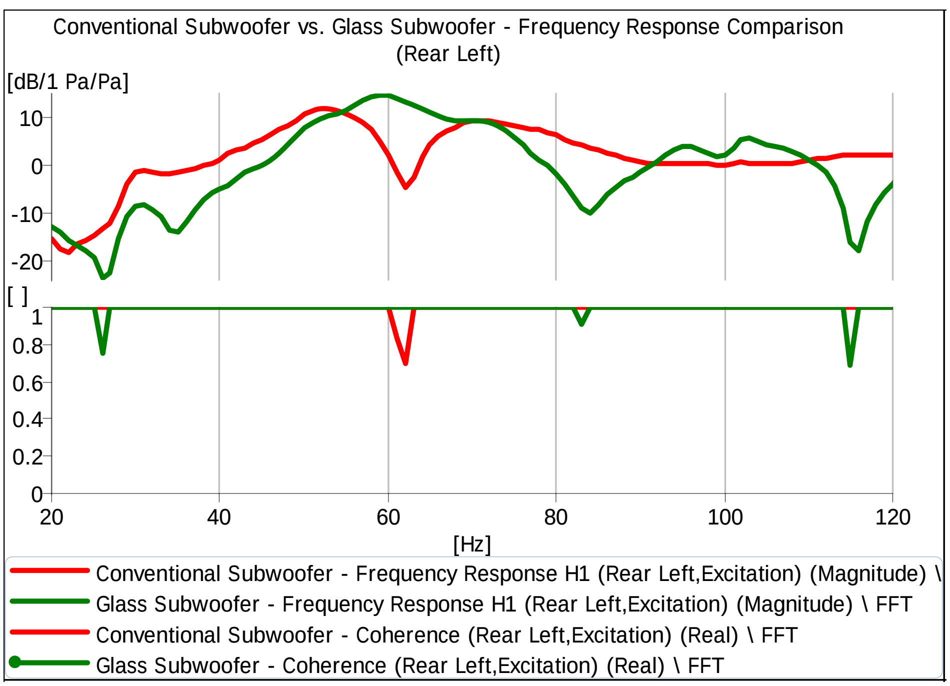

Conventional Subwoofer vs. Glass Subwoofer - Frequency Response Comparison
(Rear Right)
[dB/1 Pa/Pa]
10
0
-10
-20
[ ]
1
0.8
0.6
0.4
0.2
0
20
40
60
80
100
120
[Hz]
Conventional Subwoofer - Frequency Response H1 (Rear Right,Excitation) (Magnitude)
Glass Subwoofer - Frequency Response H1 (Rear Right,Excitation) (Magnitude) \ FFT
Conventional Subwoofer - Coherence (Rear Right,Excitation) (Real) \ FFT
Glass Subwoofer - Coherence (Rear Right,Excitation) (Real) \ FFT

EXHIBIT A6: Dual Channel Frequency Response – Swept Sine Signal





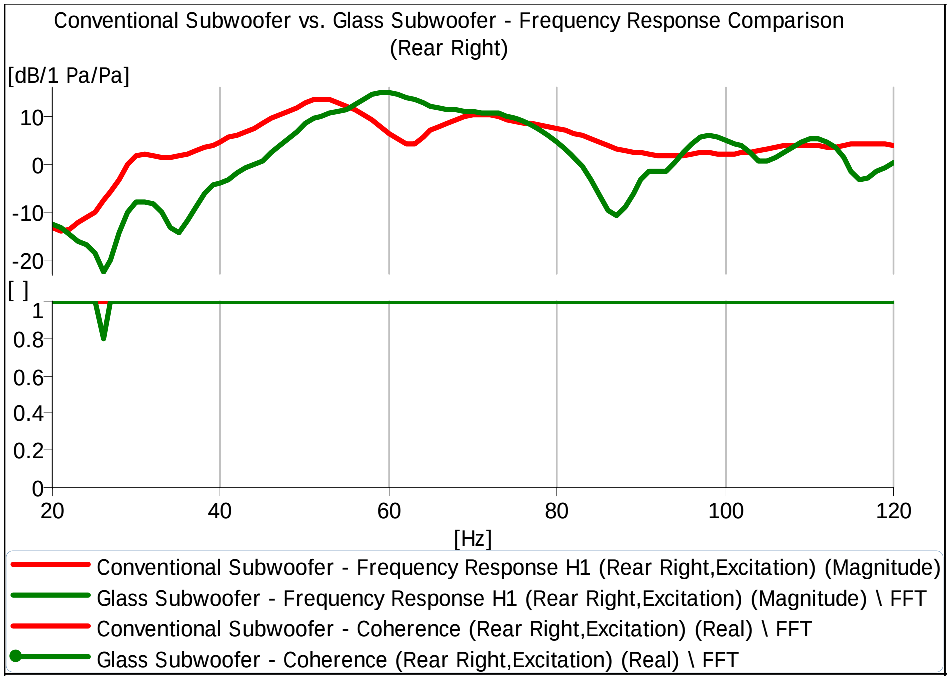
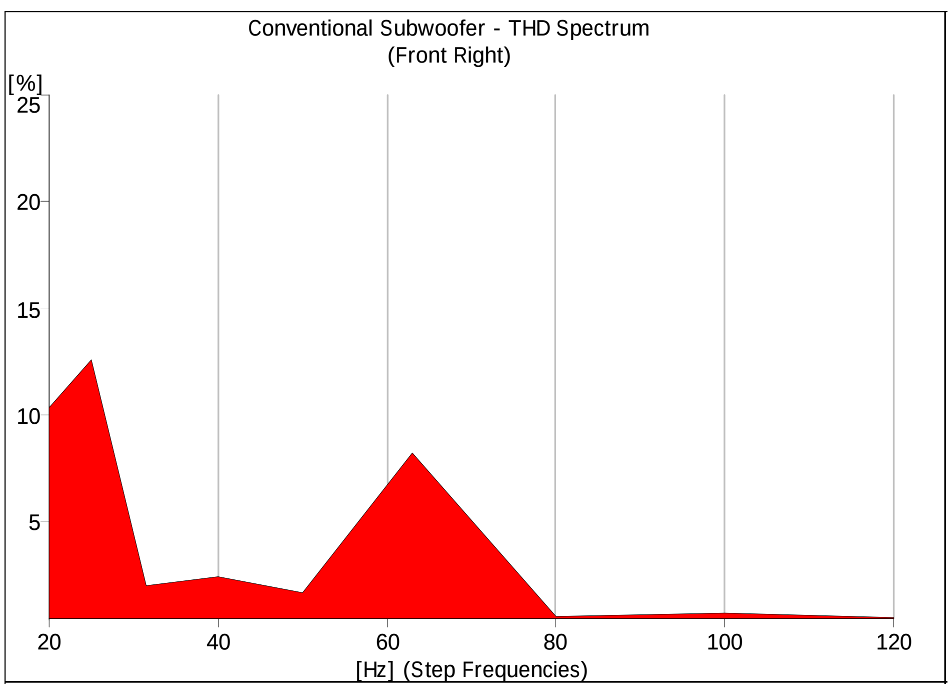
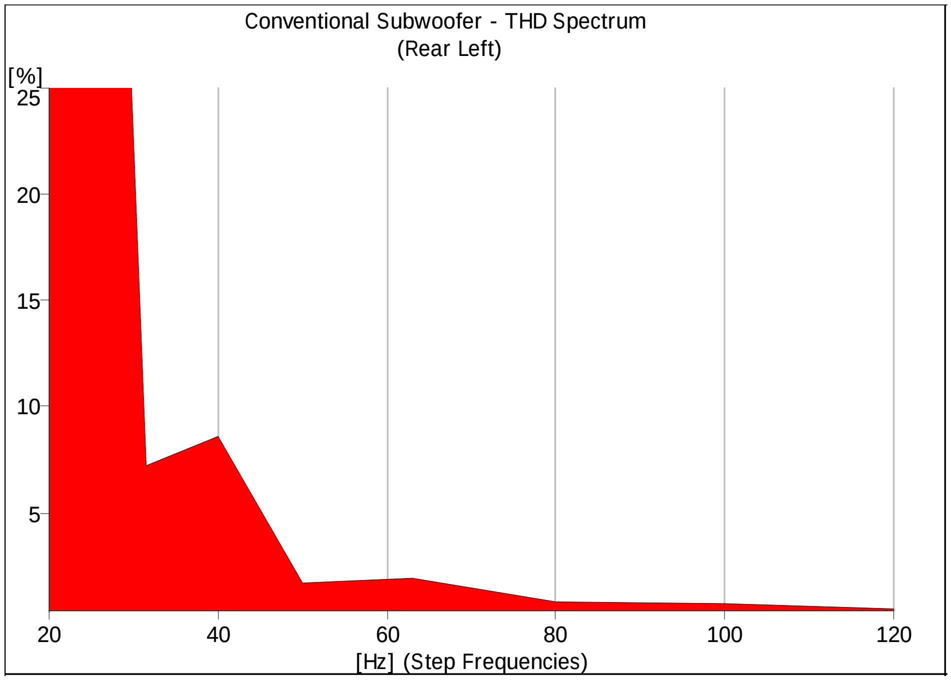
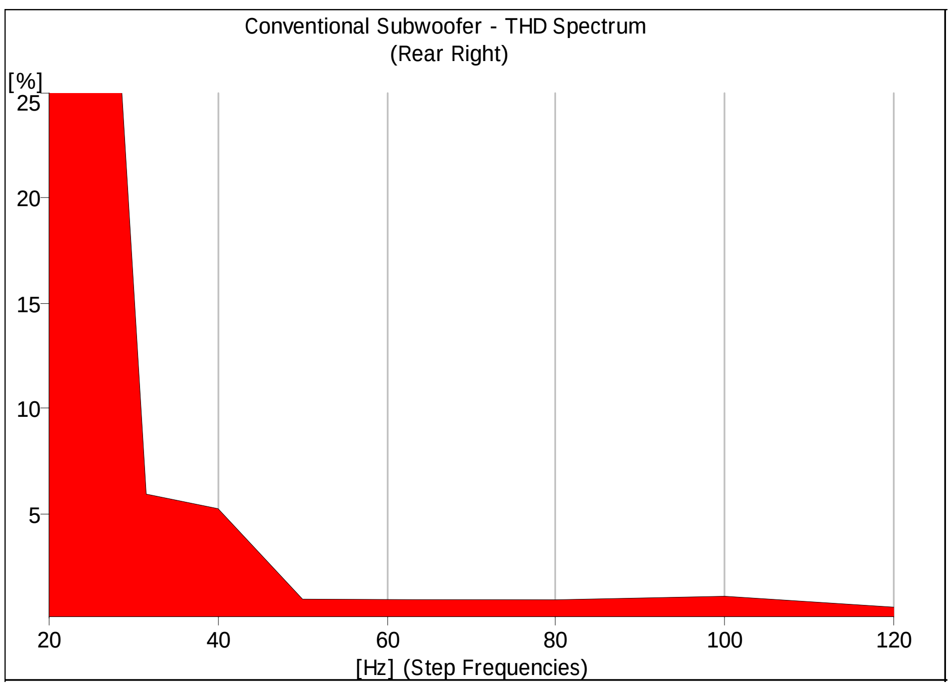


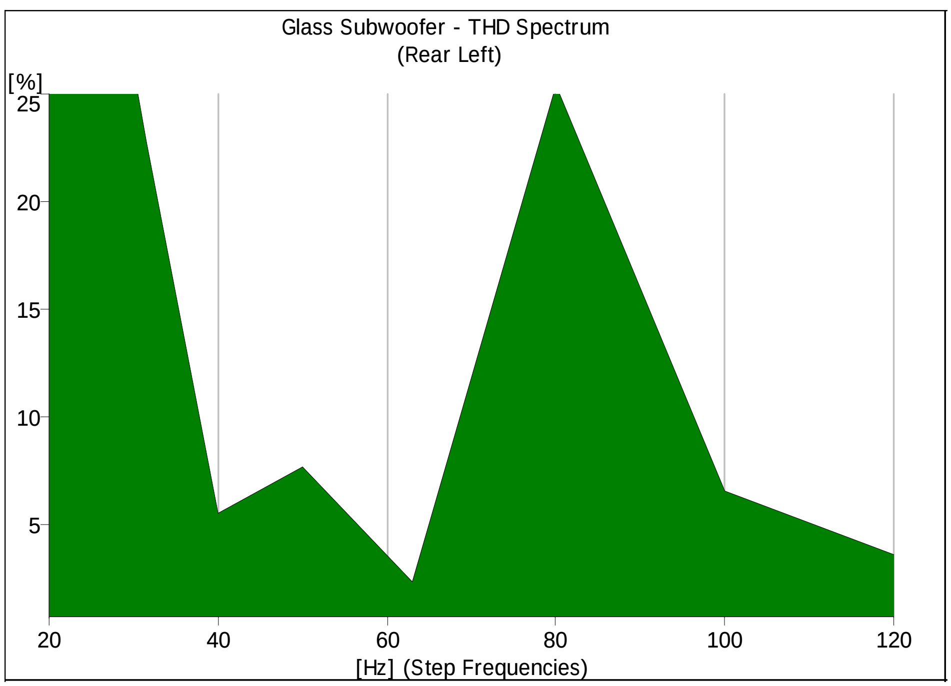
EXHIBIT A7: Total Harmonic Distortion (THD)




Conventional Subwoofer - THD Spectrum
(Rear Left)
[%]
25
20
15
10
5
20
40
60
80
100
120
[Hz] (Step Frequencies)


Conventional Subwoofer - THD Spectrum
(Rear Right)
[%]
25
20
15
10
5
20
40
60
80
100
120
[Hz] (Step Frequencies)

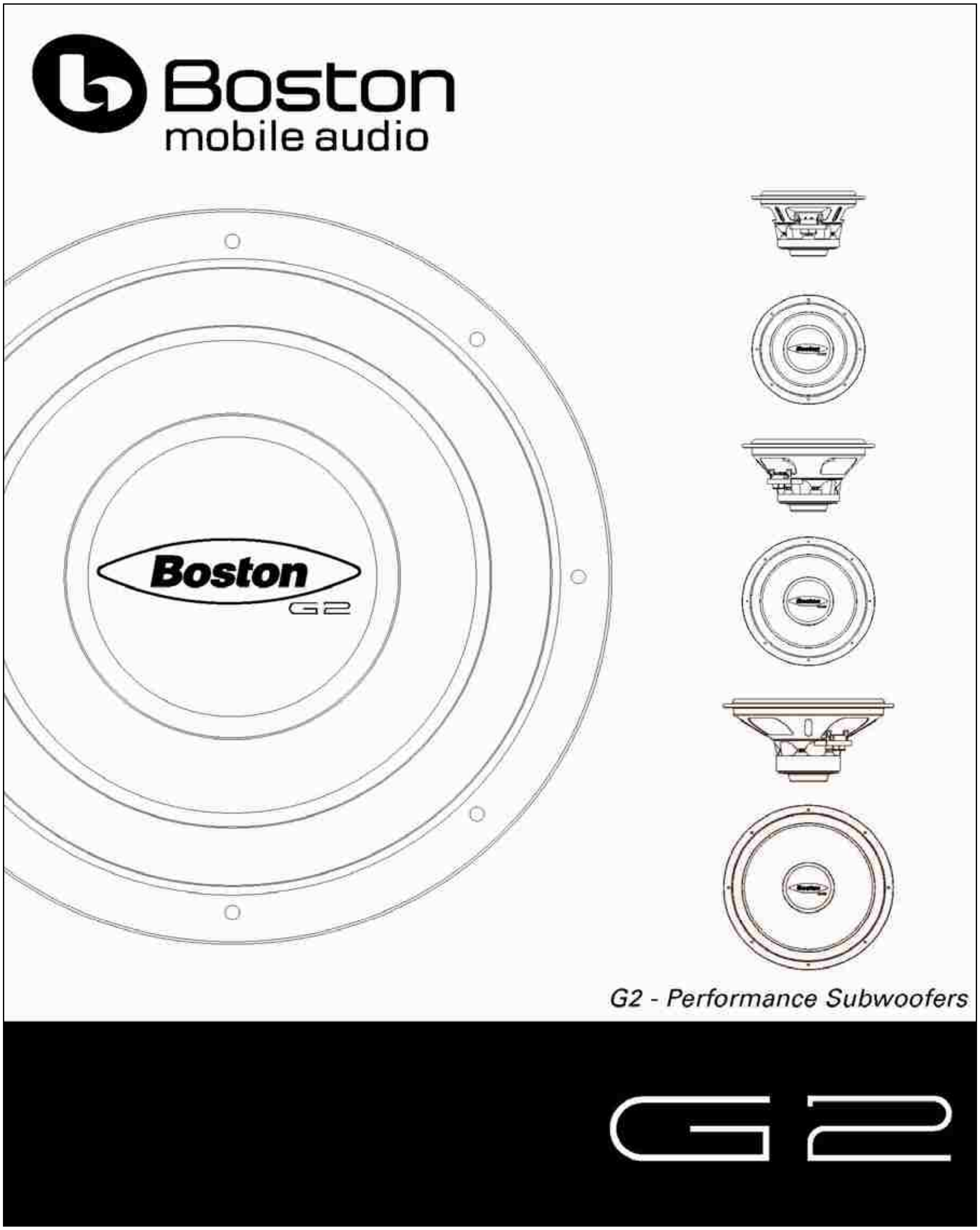

Glass Subwoofer - THD Spectrum
(Front Right)
[%]
25
20
15
10
5
20
40
60
80
100
120
[Hz] (Step Frequencies)


Glass Subwoofer - THD Spectrum
(Rear Left)
[%]
25
20
15
10
5
20
40
60
80
100
120
[Hz] (Step Frequencies)


Glass Subwoofer - THD Spectrum
(Rear Right)
[%]
25
20
15
10
5
20
40
60
80
100
120
[Hz] (Step Frequencies)

APPENDIX B – Equipment and Instrumentation Product Data Sheets

Subwoofer System - Boston Acoustics G210



Boston mobile audio

G2

G2 Specifications / Enclosure Recommendations

Model:	**G215**	**G212**	**G210**
Nominal Size:	15"	12"	10"
RMS Power Handling:	300w	300w	300w
Impedance:	4Ω or Dual 4Ω	4Ω or Dual 4Ω	4Ω or Dual 4Ω
Frequency Response: (+/-3dB in car)	18-350Hz	20-350Hz	20-350Hz
Mounting Cutout Diameter:	13 15/16" (354mm)	11 1/8" (283mm)	9 1/4" (235mm)
Mounting Depth:	7 1/16" (179mm)	7 3/16" (183mm)	6 3/8" (162mm)
Recommended Sealed Enclosure Volume*:	2.0 ft³ (56.6L)	1.0 ft³ (28.3L)	0.5 ft³ (14.2L)
Q-Tune™†: (HP & Q Setting)	26Hz@0.7	30Hz@0.7	32Hz@0.7
Recommended Ported Enclosure Volume*:	3.5 ft³ (99.1L)	2.0 ft³ (56.6L)	1.3 ft³ (36.8L)
Vent Diameter:	2 x 4" (10cm)	4" (10cm)	3" (7.6cm)
Vent Length:	14" (36cm)	13" (33cm)	10" (25cm)
Tuning Frequency:	34Hz	34Hz	36Hz
Q-Tune™†: (HP & Q Setting)	34Hz@0.9	34Hz@0.7	36Hz@0.9

*Enclosure volumes include basket and port displacement
†Q-Tune™ is a feature found on Boston GT Amplifiers

G2 Thiele-Small Parameters

Model:	**G21544**		**G2154**	**G21244**		**G2124**	**G21044**		**G2104**
	Parallel	Series		Parallel	Series		Parallel	Series	
Fs: (Hz)	26	26	26	28	28	28	34	34	37
Re: (Ohms)	1.7	6.8	3.4	1.7	6.6	3.3	1.6	6.5	3.3
Qms:	9.39	9.39	9.79	11.23	11.23	11.71	11.07	11.07	11.41
Qes:	0.85	0.85	1.06	0.61	0.61	0.68	0.62	0.62	0.74
Qts:	0.78	0.78	0.96	0.58	0.58	0.65	0.59	0.59	0.70
Vas: (Liters)	187.0	187.0	139.5	65.0	65.0	73.2	22.0	22.0	22.3
Mms: (Grams)	253.5	253.5	253.3	191.0	191.0	163.3	167.3	167.3	141.3
Cms: (μM/Newton)	162	162	147	171	171	193	132	132	134
Xmax: (Mm)	13.3	13.3	12.5	13.3	13.3	12.5	13.3	13.3	12.5
Xmech: (Mm)	30	30	30	30	30	30	30	30	30
Sd: (CM²)	907	907	907	520	520	520	345	345	345
Bl: (Tesla-M)	11.56	23.12	8.88	9.56	19.12	11.89	9.70	19.40	11.98
SPL Eff: (dB @ 1w/1m)	87.1	87.1	86.4	85.5	85.5	85.8	83.3	83.3	83.5
SPL Sen: (dB @ 2.83v)	93.9	87.9	90.2	92.3	86.3	89.6	90.2	84.2	87.4

G2 Included Hardware

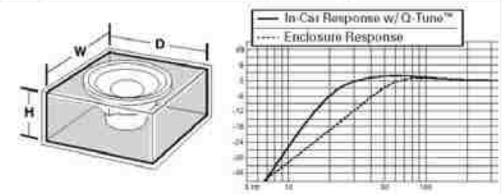
Model:	**G215**	**G212**	**G210**
5-Amp Fuse:	2	2	2
T-Nut M4:	8	8	8
Lock Washer M4:	8	8	8
Mounting Screw M4:	8	8	8
Wood Screw M4 x 38mm:	8	8	8

SureSet™ Fuse Configurations

G21544 / G21244 / G21044 | **G2154 / G2124 / G2104**

Note: The SureSet™ feature utilizes standard automotive ATC fuses to select between series or parallel voice-coil operation (DVC only). The fuses also provide voice-coil protection under transient conditions such as accidental momentary overdrive. The fuses will not protect against voice-coil failure resulting from long term high power abuse.

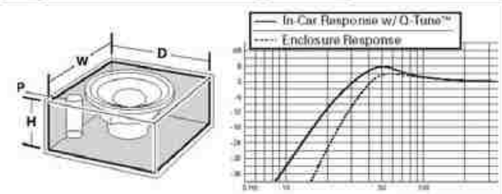
Single G2 Enclosure Design Example - Sealed



Model:	**G215***	**G212***	**G210***
H = Height:	18.0" (45.7cm)	10.5" (26.7cm)	7.5" (19.1cm)
W = Width:	18.0" (45.7cm)	13.0" (33.1cm)	11.0" (28.0cm)
D = Depth:	10.5" (26.7cm)	13.0" (33.1cm)	11.0" (28.0cm)
Gross Volume:	2.0 Ft³ (56.6L)	1.0 Ft³ (28.3L)	0.5 Ft³ (14.2L)
Q-Tune™†: (HP & Q)	26Hz@0.707	30Hz@0.707	32Hz@0.707

Enclosure dimensions are internal and include basket displacement
This enclosure size is an example and these dimensions can be modified provided that the exact internal volume is retained
¾-inch thick M.D.F. is recommended as a minimum
†Q-Tune™ is a feature found on Boston GT Amplifiers

Single G2 Enclosure Design Example - Ported



Model:	**G215**	**G212***	**G210***
H = Height:	20" (48.3cm)	15.5" (39.8cm)	12" (30.5cm)
W = Width:	18" (48.3cm)	16.5" (42.0cm)	13.25" (33.7cm)
D = Depth:	17 (36.8cm)	14" (35.6cm)	14" (35.6cm)
P = Port Clearance*:	4" (10cm)	4" (10cm)	3" (7.6cm)
Gross Volume:	3.5 Ft³ (99.1L)	2.0 Ft³ (56.6L)	1.3 Ft³ (36.8L)
Port Diameter:	2 x 4" (10cm)	4" (10cm)	3" (7.6cm)
Port Length:	14" (36cm)	13" (33cm)	10" (25cm)
Q-Tune™†: (HP & Q)	34Hz@0.9	34Hz@0.707	36Hz@0.9

Enclosure dimensions are internal and include basket and port displacement
This enclosure size is an example and these dimensions can be modified provided that the exact internal volume is retained and that there is adequate space behind the vent (diameter of port from the back and side walls)
¾-inch thick M.D.F. is recommended as a minimum
*Recommended distance from sides and back walls of enclosure
†Q-Tune™ is a feature found on Boston GT & GTA Amplifiers

142-001942-3

Audio Head Unit – Kenwood KDC-X794

KENWOOD
Listen to the Future

Kenwood Electronics Canada Inc.

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- Language

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  - eXcelon
  - Multimedia Receivers
  - Digital Media Receivers
  - CD Receivers
  - Marine
  - Amplifiers
  - Subwoofers
  - Speakers
- Communications

KDC-X794

eXcelon In-Dash CD Receiver With iPod USB Direct Control

STEERING REMOTE INPUT 50WX4 MOSFET

eXcelon

Overview General Features Specifications Accessories Support Manuals

Specifications

Tuner	
- FM	
- FM Frequency Range	87.9MHz - 107.9MHz
- FM Frequency Step	(200kHz)
- FM Usable Sensitivity	9.3dBf
- FM Quiet Sensitivity	-
- FM Frequency Response (±3.0dB)	30Hz-15kHz
- FM Signal/Noise	70dB(MONO)
- FM Selectivity	Over80dB(±400kHz)
- FM Stereo Separation	40dB(1kHz)
- AM	
- AM Frequency Range	530kHz-1700kHz
- AM Frequency Step	(10kHz)
- AM Usable Sensitivity	28dBμ (25μv)
CD	
- Digital Filter(D/A)	8 Times OverSampling
- D/A Converter	24 Bit
- Frequency Response	10-20kHz(±1dB)
- Total Harmonic Distortion	0.008%(1kHz)
- Signal/Noise Ratio (dB)	110dB(1kHz)
- Dynamic Range	93dB
- MP3 Decode	Compliant with MPEG-1/2 Audio Layer-3
- WMA Decode	Compliant with Windows Media Audio
- AAC Decode	AAC-LC ".m4a" Files
- WAV Decode	-
USB I/F	
- Compatibility	USB 1.1 & 2.0 Full Speed
- File System	FAT16/32
- Maximum Supply Current	500mA
- Decode	MP3/WMA/AAC
Preout Level(mV)/Load - Unbalanced	4000mV/10kΩ(CD/CD-CH)
Preout Impedance(Ω)	Under 600Ω

Speaker Impedance(Ω)	4-8Ω
AMP	
- Maximum Power	50Wx4
- Full Bandwidth Power	22Wx4
Bluetooth	
- Version	-
- Maximum Range	-
- Profiles	
- HFP	-
- HSP	-
- SPP	-
- A2DP	-
- AVRCP	-
- PBAP	-
- OPP	-
- SYNC	-
- MAP	-
Tone	
- Band1	60Hz ±9dB
- Band2	250Hz ±9dB
- Band3	1kHz ±9dB
- Band4	4kHz ±9dB
- Band5	16kHz ±9dB
General	
- Operating Voltage	14.4V (11V - 16V allowable)
- Maximum Current Consumption	10A
- Installation Size	
- Width	182(mm) 7-3/16(in)
- Height	53(mm) 2-1/16(in)
- Depth	158(mm) 6-1/4(in)

*Product descriptions, specifications and features are subject to change without notice. For proper fit and compatibility check with your authorized Kenwood dealer.

Amplifier Dedicated to Speakers – JL Audio XD 600/6

XD Technologies | Tutorials | Reviews

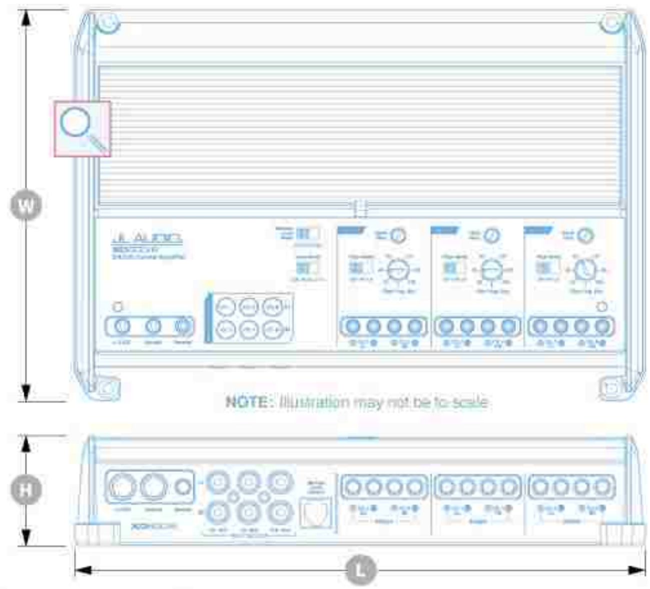
XD600/6 600W 3/4/5/6 Channel Amplifier Features:

	XD600/6
Description:	600W 3/4/5/6 Channel Amplifier
Inputs:	Three Stereo Pairs (RCA-Type Jacks), Differential-Balanced Inputs
On-board Crossover:	State-variable, 12dB/octave Butterworth with continuously variable cutoff frequency selection from 50-500 Hz. Configurable as Low-Pass or High-Pass. Defeatable.
Speaker output connections:	Accept up to 8 AWG wire
+12V and Ground connections:	Accept up to 4 AWG wire

Notes: NexD™ Ultra-High Speed Class D, Remote Level Control (2 Ch, 4 Ch. or All Channels), via Optional HD-RLC (Sold Separately).

Get ADOBE® READER®

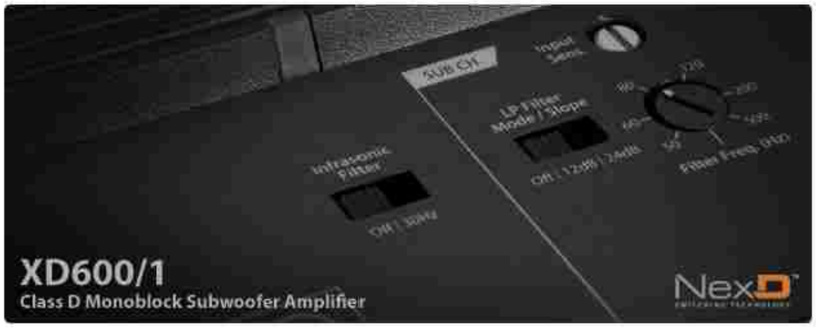
Click here to download the owner's manual (PDF).



XD600/6 600W 3/4/5/6 Channel Amplifier Specifications:

	XD600/6
Rated Power 14.4 V Stereo:	75W RMS x 6 @ 4 ohms 100W RMS x 6 @ 2 ohms
Rated Power 14.4 V Bridged :	150W RMS x 3 @ 8 ohms 200W RMS x 3 @ 4 ohms
Rated Power 12.5 V Stereo:	60W RMS x 6 @ 4 ohms 90W RMS x 6 @ 2 ohms
Rated Power 12.5 V Bridged:	120W RMS x 3 @ 8 ohms 180W RMS x 3 @ 4 ohms
THD at Rated Power:	<1% THD, 20 Hz - 20 kHz
S/N Ratio:	>104 dB referred to rated power (A-weighted, 20 Hz-20 kHz noise bandwidth) >84 dB referred to 1W (A-weighted, 20 Hz-20 kHz noise bandwidth)
Frequency Response:	12 Hz - 22 kHz (+0, -1dB)
Damping Factor:	>150 @ 4 ohms per ch./ 50 Hz >75 @ 2 ohm per ch. / 50 Hz
Input Range:	100mV - 4V RMS
Dimensions (L x W x H):	10.23 in. x 7.09 in. x 2.05 in. 260 mm x 180 mm x 52 mm

Amplifier Dedicated to Subwoofer – JL Audio XD 600/1



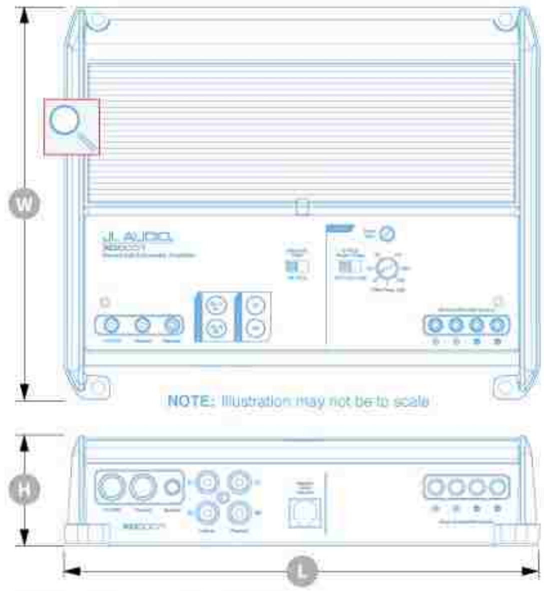
XD Technologies Tutorials Reviews

XD600/1 Monoblock Subwoofer Amplifier Features:

	XD600/1
Description:	Monoblock Subwoofer Amplifier
Inputs:	One Stereo Pair (RCA-Type Jacks), Differential-Balanced Inputs, sum to mono internally.
On-board Crossover:	State-variable, 12 dB/octave Butterworth or 24 dB/octave Linkwitz-Riley with continuously variable cutoff frequency selection from 50 - 500 Hz (defeatable).
Preamp Output:	Buffered pass-through type
Infrasonic Filter:	24 dB/octave Butterworth @ 30 Hz, defeatable

Dual mono speaker output connections:	Accept up to 12 AWG wire
+12V and Ground connections:	Accept up to 4 AWG wire
Notes: NexD™ High Speed Class D, Remote Level Control via Optional HD-RLC (Sold Separately).	

Click here to download the owner's manual (PDF).



XD600/1 Monoblock Subwoofer Amplifier Specifications:

	XD600/1
Rated Power:	600 W RMS @ 2 ohm (14.5V)
Rated Power 14.4 V:	400W RMS x 1 @ 4 ohms 600W RMS x 1 @ 2 ohms
Rated Power 12.5 V:	300W RMS x 1 @ 4 ohms 500W RMS x 1 @ 2 ohms
THD at Rated Power:	1% @ 2 ohm
S/N Ratio:	>87 dB referred to rated power (A-weighted, 20 Hz-20 kHz noise bandwidth) >67 dB referred to 1W (A-weighted, 20 Hz-20 kHz noise bandwidth)
Frequency Response:	7 Hz - 500 Hz (+0, -1dB)
Damping Factor:	>1000 @ 4 ohms / 50 Hz >500 @ 2 ohms / 50 Hz
Input Range:	100mV - 4V RMS
Dimensions (L x W x H):	8.52 in. x 7.09 in. x 2.05 in. 217 mm x 180 mm x 52 mm

Audio Control – LC8i

Eight Channel Line Output Converter with Auxiliary Input - LC8i

LC8i™

OEM Interface

Eight Channel Line Output Converter with Auxiliary Input

The new LC8i is a highly flexible ***eight-channel*** signal processor with active input circuitry that accepts speaker-level inputs, of up to 400 watts per channel, from factory-installed source units or amplifiers. It converts them to high-quality pre-amp signals that can then be directed to aftermarket processors and/or amplifiers. This means users will enjoy maximum sound quality while maintaining the use of factory volume and steering wheel controls.

Designed for maximum flexibility, the LC8i is 30% **smaller** than it's predecessor and it is equipped with an auxiliary input that allows the simple addition of secondary source units like iPods and satellite radio tuners. By tapping the dash control knob, a user can switch back and forth between the main speaker-level inputs and the auxiliary input. Additionally, the multi-function dash knob provides both volume control of the Auxiliary input ***and*** level adjustments for the subwoofer output.

Click Play to Watch Video

New Features

- Eight input and output channels – will accept **400 watt signals** ...per channel
- Compact design – **30%** smaller chassis than original AudioControl LC8
- Auxiliary input for easy interfacing of iPods or other portable audio devices
- Multi-function dash control (included) allows for Source Selection, Auxiliary Input Volume **plus** Subwoofer Level Control
- Internal channel summing
- Selectable GTO™ Signal-sensing Turn-on Circuit

All AudioControl autosound components are backed by our industry leading, **FIVE** year parts and labor warranty when installed by an authorized dealer.

View Larger Image

View Zoom Image

More . . .

LC8i Owner's Manual

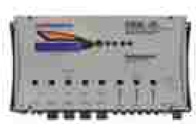
Mobile Audio Catalog

LC8i Sales Note

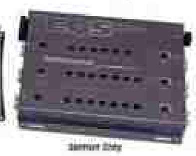
Other Items to Consider

DQXS Digital Crossover / Equalizer

DQS Six Channel Digital Car Audio Equalizer



DQL-8 Eight Channel Line Output Converter with Auxiliary Input and Digital Signal Processing



EQS Six Channel Trunk Mount Equalizer

Data Acquisition and Analysis Software - Bruel & Kjaer PULSE v15

PRODUCT DATA

Software for PULSE™ 15
incl. Types 7700, 7705, 7707, 7709, 7764, 7770, 7771, 7773, 7789 and 7797

PULSE is Brüel & Kjær's platform for noise and vibration analysis and builds on over 60 years of measurement experience and innovation.

The PULSE hardware/software family is your solid foundation upon which to build a system to suit your present needs, and which can also be extended as your requirements change. This expandability, and the continuing development of new PULSE applications and hardware, ensures the safety of your investment now and in the future.

PULSE's flexibility, combined with industry-specific solutions, has made PULSE Brüel & Kjær's best-selling analyzer platform. The PULSE system is a leader in a wide range of industries, including:

- *Automotive*
- *Electroacoustics and Telecommunications*
- *Aerospace and Defence*
- *Consumer Products*

PULSE Software and Literature Overview

The base measurement software for a PULSE system is PULSE FFT & CPB Analysis Type 7700. Separate FFT and CPB licenses are also available as FFT Analysis Type 7770 and CPB Analysis Type 7771. On this base, you can install PULSE application software such as Multichannel Data Recorder Type 7708. Table 1 illustrates the range of application software available for use with PULSE systems.

With a PULSE Software Maintenance and Support Agreement (M1) you can ensure that your PULSE installation is kept updated to the latest security updates from Microsoft® as well as having access to a global network of specialists, with experience from more than 10000 PULSE systems in a multitude of application and test configurations. Details of the PULSE Software Maintenance and Support Agreement are given in BP 1800.

We strongly recommend that you update your PULSE installation to the latest major release to ensure that the latest security updates from Microsoft® are supported by your installation.

Details on PULSE Reflex™, the post-processing suite from Brüel & Kjær that brings together a wide range of generic post-processing tools for off-line analysis and processing of time data and spectra, can be found in application-specific Product Data such as PULSE Reflex Core, which contains PULSE™ LabShop compatible FFT, CPB (1/n-octave) and Order Analysis (BP 2258); PULSE Reflex Building Acoustics (BP 2190) and PULSE Reflex Modal Analysis (BP 2257).

Details of the LAN-based hardware available for use with PULSE are given in the LAN-XI Data Acquisition Hardware Product Data (BP 2215) and the IDA^{e} Hardware Configurations for PULSE System Data (BU 0228).

Brüel & Kjær

Table 1 Overview of PULSE application software specifying support of either FFT & CPB Analysis Type 7700, FFT Analysis Type 7770 and/or CPB Analysis Type 7771. References to Brüel & Kjær source literature are also specified

	Type/Part Number	FFT and CPB Analysis Type 7700	FFT Analysis Type 7770	CPB Analysis Type 7771	Further Information	Specifications
Platform Enhancements						
PULSE Time Capture	7705	•	•	•	page 9	page 16
PULSE Analysis Engine Upgrade	7707	•	•	•	page 7	page 16
PULSE Time Data Recorder	7708	•	•	•	BP 2110	BP 2110
PULSE Viewer	7709	•	•	•	page 8	–
IDA[e] Driver for I-deas	BZ-5231	•	•	•	–	–
PULSE Reflex Base	8700				BP 2258	BP 2258
PULSE Reflex Basic Post-processing	8702	•	•		BP 2258	BP 2258
PULSE Reflex Standardised CPB Option	8706	•		•	BP 2258	BP 2258
LAN-XI Notar	BZ-7848-A	•	•	•	BP 2215	BP 2215
Acoustic Applications						
PULSE Sound Quality	7698	•	•	•	BP 1589	BP 1589
PULSE Noise Source Identification	7752	•	•	•	BP 1908	BP 1908
PULSE Material Testing	7758	•	•		BP 1870	BP 1870
PULSE Advanced Intensity Analysis	7759	•	•	•	BP 1890	BP 1890
PULSE Acoustic Test Consultant	7761	•	•	•	BP 1908	BP 1908
PULSE Pass-by Conformance Test System	7788-A	•			BP 2256	BP 2256
PULSE Vehicle Pass-by	7788-B, -C	•			BP 2011	BP 2011
PULSE Indoor Pass-by	7793	•			BP 2015	BP 2015
PULSE Sound Power	7799	•			BP 2093	BP 2093
PULSE Spherical Beamforming	8606	•	•	•	BP 2144	BP 2144
PULSE Acoustic Holography	8607	•	•	•	BP 2144	BP 2144
PULSE Beamforming	8608	•	•	•	BP 2144	BP 2144
PULSE Sound Quality Zwicker Loudness	BZ-5265	•	•	•	BP 1589	BP 1589
PULSE Sound Quality Order Analysis	BZ-5277	•	•	•	BP 1589	BP 1589
PULSE Pyschoacoustic Test Bench	BZ-5301	•	•	•	BP 1589	BP 1589
Robot Option for ATC	BZ-5370	•	•	•	BP 1908	BP 1908
PULSE Position Detection Option	BZ-5611	•	•	•	BP 1908	BP 1908
PULSE Quasi-stationary Calculations	BZ-5635	•	•	•	BP 2144	BP 2144
PULSE Transient Calculations	BZ-5636	•	•	•	BP 2144	BP 2144
PULSE Conformal Calculations	BZ-5637	•	•	•	BP 2144	BP 2144
Electroacoustics						
PULSE Basic Electroacoustics	7797	•	•		page 12	page 17
PULSE Electroacoustics	7907	•	•	•	BP 2085	BP 2085
PULSE Voice Testing for Hands-free Equipment	7909-S 1	•	•		BP 2116	BP 2116
Telephone Test on PULSE	BZ-5137	•			BP 1684	BP 1684
PULSE SSR Analysis – Harmonic Distortion	BZ-5548	•	•		BP 2085	BP 2085
PULSE SSR Analysis – Intermodulation Distortion	BZ-5549	•	•		BP 2085	BP 2085
PULSE SSR Analysis – Difference Frequency Distortion	BZ-5550	•	•		BP 2085	BP 2085
PULSE Directivity and Polar Plot	BZ-5551	•	•		BP 2085	BP 2085
PULSE Sequencer	BZ-5600	•	•		BP 2085	BP 2085
PDM for Electroacoustics	BZ-5601	•	•		BP 2085	BP 2085
PULSE Receiver Test Applications	BZ-5602	•	•		BP 2085	BP 2085
PULSE Loudspeaker Test Applications	BZ-5603	•	•		BP 2085	BP 2085
PULSE Thiele Small Parameter Calculation	BZ-5604	•	•		BP 2085	BP 2085
PULSE TSR Analysis – Harmonic Distortion	BZ-5742	•	•		BP 2085	BP 2085
PULSE Microphone Test Application	BZ-5743	•	•		BP 2085	BP 2085
PULSE Headset Test Application	BZ-5744	•	•		BP 2085	BP 2085
Machine Diagnostics						
PULSE Order Analysis	7702	•	•		BP 1634	BP 1634
PULSE Vold-Kalman Order Tracking Filter	7703	•	•		BP 1760	BP 1760
PULSE Envelope Analysis	7773	•	•		page 11	page 17
PULSE Two-plane and Multi-plane Balancing Consultants	7790-A/B	•	•		BP 2010	BP 2010
PULSE Vibration Check for Aircraft Engines	7795	•	•		BP 2059	BP 2059
PULSE Vibration Analysis for Aircraft Engines	7906-S 1	•	•		BP 2059	BP 2059
Orbit and Polar Plots for PULSE	WT-9695	•	•		–	–
PULSE Reflex Basic Order Analysis	8704	•	•		BP 2258	BP 2258

	Type/Part Number	FFT and CPB Analysis Type 7700	FFT Analysis Type 7770	CPB Analysis Type 7771	Further Information	Specifications
Structural Dynamics						
PULSE Structural Dynamic Test Consultants	7753/7765	•	•		BP 1850	BP 1850
ME'scopeVES™ Modal and Structural Analysis, incl. PULSE Bridge to ME'scopeVES	7754/7755-A	•	•		BP 1843	BP 1843
PULSE Operational Modal Analysis	7760	•	•		BP 1889	BP 1889
PULSE Multiple-Input Multiple-Output Analysis	7764	•	•		page 10	page 17
PULSE Run-up/down ODS Option	BZ-5612	•	•		BP 1850	BP 1850
PULSE Animation Option	BZ-5613	•	•		BP 1850	BP 1850
PULSE Reflex Geometry	8719	•	•		BP 2257	BP 2257
PULSE Reflex Basic Modal Analysis	8720	•	•		BP 2257	BP 2257
PULSE Reflex Advanced Modal Analysis	8720	•	•		BP 2257	BP 2257
PULSE Reflex Shock Response Analysis	8730	•	•		BP 2339	BP 2339
Vibroacoustics						
PULSE Source Path Contribution	7798	•	•		BP 2086	BP 2086
PULSE DTS Software for NVH Simulator	8601	•	•		BP 2109	BP 2109
Test and Data Management						
PULSE Data Manager	7767	•	•	•	BP 1961	BP 1961
PULSE Time	7789	•	•	•	page 8	page 17
PULSE Automotive Test Manager	7796	•	•	•	BP 2061	BP 2061
PULSE ASAM ODS Connectivity	8605	•	•	•	BP 2187	BP 2187
PULSE CAN Bus Option	BZ-5610	•	•	•	BP 2150	BP 2150
Environmental Noise & Vibration						
PULSE Reflex Building Acoustics	8780				BP 2190	BP 2190

Pre-configured PULSE bundles (measurement software and hardware) are available for many common noise and vibration measurements. See the PULSE Analyzers and Solutions catalogue BF 0209 for a complete description of available PULSE bundles.

Data Acquisition and Analysis Software - Bruel & Kjaer PULSE Sound Quality

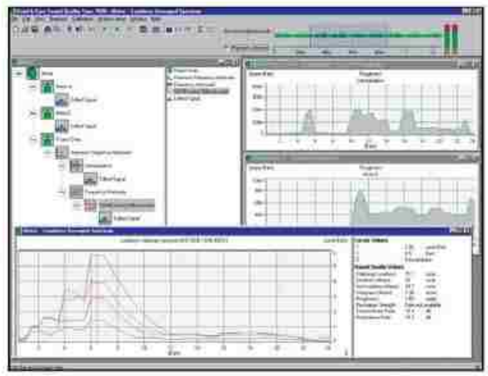
PRODUCT DATA

PULSE Sound Quality Software — Type 7698
The PULSE Application for Analysing and Improving Sound Quality

PULSE Sound Quality Software Type 7698 is advanced, stand-alone software which can record, analyse, edit and play back binaural or monaural product sounds or other audio signals. With the benefit of OLE automation, the software can be controlled from other applications and provides a direct interface with the new Psychoacoustic Test Bench software. This organises subjective and objective tests and correlates the results into a combination metric.

Sound Quality software can record up to four channels using a four channel sound card. It can also import signals recorded, for example, with Portable PULSE, for sound quality evaluations in addition to other available PULSE analysis methods.

PULSE Sound Quality software is the core of a complete sound quality system. Add the necessary hardware and you have a complete sound quality solution.



USES AND FEATURES

USES
- Analysis of product sound
- Editing recorded sounds to simulate product improvement
- Preparing listening tests and play lists for product evaluation
- Determining sound quality parameters: loudness, non-stationary loudness, binaural loudness, sharpness, fluctuation strength, roughness and related parameters
- Visualising and editing orders on rotating machinery

FEATURES
- Runs under Microsoft® Windows® 2000 or XP with Microsoft® sound system compatible sound card (up to 4 channels)
- Reads PULSE Data Recorder files
- Controls PULSE for measurements with Data Recorder or Time Capture options
- Powerful Zwicker Loudness analysis option BZ 5265
- Order Analysis capabilities with option BZ 5277
- Subjective/Objective correlation tool with Psychoacoustic Test Bench option BZ 5301
- Jury Test tool for designing and executing sound quality listening tests
- Psychoacoustic correction for improved realism during playback
- Frequency and time domain editing of multiple signals with real-time capability
- Displays multispectra as waterfalls, contour plots, envelopes and slices
- Fully OLE programmable for automating routine tasks such as analysis and reporting
- User-definable edits using Visual Basic® or Visual C++®
- User-definable macros using VBScript or JavaScript
- Performs regression analysis and creates a combination metric with Psychoacoustic Test Bench option BZ 5301

BENEFIT
- A complete sound quality solution

Brüel & Kjær

Specifications – PULSE Sound Quality Software Type 7698

SIGNAL INPUT/OUTPUT
- 2 channels (+ 2 tacho channels with BZ 5277)
- Analog or Digital
- Tacho can be read from a 16th-bit encoding or normal encoded
- Direct recording with PULSE and Time Capture or Data Recorder options
- Loading of time data from existing PULSE projects and recorder files

ANALYSIS
- Overall Levels: RMS, Statistics, Metric Statistics
- FFT
- Statistical Regression Analysis (with BZ 5301)
- Envelope

ZWICKER LOUDNESS ANALYSIS (WITH BZ 5265)
- Stationary Loudness (according to DIN 45631/ISO 532B)
- Non-stationary Loudness (according to Zwicker Loudness)
- Binaural Louness

ZWICKER LOUDNESS METRICS (WITH BZ 5265)
- Loudness
- Sharpness
- Roughness
- Fluctuation Strength

OTHER METRICS
- Tone-to-noise ratio (according to ANSI S1.13–1995)
- Prominence ratio (according to ANSI S1.13–1995)
- A, B, C, D weighting
- User-defined cursor readings

EDITS
- Peak Limit
- Time Attenuate
- Level Edit, +/– 20 dB
- Demodulation
- Frequency Attenuate
- Frequency Shift
- Passband
- Peak Limit Frequency
- Harmonic Frequency Attenuate
- Harmonic Frequency Shift
- Harmonic Passband
- Generator
- Mixer
- Real-time Filter
- Frequency Response Filter
- User-defined Filter

DISPLAYS
- Real-time playback monitor
- Time (Lin/Lin, Log/Lin, Lin/Log, Log/Log, Waterfall, Contour)
- Single Signal or Multiple Signal Graphs

ORDER ANALYSIS (WITH BZ 5277)
- Schmitt trigger of tacho signals
- RPM extraction from tacho signals
- FFT analysis of sound signal using RPM signal to determine orders

ORDER-RELATED EDITS (WITH BZ 5277)
- Order Attenuate
  position specified by order start and width attenuation in dB or absolute
- Order Passband
  position specified by order start and width stop-band gain in dB or absolute

ORDER-RELATED DISPLAYS (WITH BZ 5277)
- Time graphs:
  Tacho
  Schmitt triggered tacho
  RPM
- FFT Contour versus RPM
- FFT Waterfall versus RPM

with Zwicker Loudness Metrics (with BZ 5265)
- Loudness versus RPM
- Roughness versus RPM
- Fluctuation Strength versus RPM

PSYCHOACOUSTIC TEST BENCH (WITH BZ 5301)
- Controls Type 7698 from an Excel spreadsheet
- Set up of calculation of objective metrics for wave signals
- Set up of subjective tests and evaluation of results from jury members
- Remote jury tests using Internet
- Regression analysis for calculation of combination metrics

WEIGHTING CURVE CORRECTIONS
- Input (Flat, 4100 diffuse-field, 4100 free-field or user-defined)
- Output (Flat, HT 0012, HT 0017 or user-defined)

CALIBRATION
- Input level
- Output level
- Charge Injection Calibration

AUTOMATION
- OLE 2 interface
- Programmable from e.g., Visual Basic®, Visual C++®, Microsoft® Excel, Microsoft® Word
- VBScript, JavaScript

DATA IMPORT AND EXPORT
Import and export of multichannel external file formats: HDF/DAT – HEAD Acoustics ArtemiS format. Support for both 16- and 24-bit data (Note: In 16-bit format, the tachometer information is embedded in the LSB of the left channel). MATLAB time format – contains no calibration information

Data Export
- .WAV (wave file)
- .TXT (ASCII text file)
- Universal File Format (UFF)
- HDF/DAT
- MATLAB
- TEAC GX-1

Data Import
- .WAV (wave file)
- Universal File Format (UFF)
- HDF/DAT
- MATLAB
- TEAC GX-1
- Import from base PULSE software – requires that either Data Recorder Type 7701 or Time Capture Type 7705 is installed on PULSE computer. Support of 24-bit DAT files from Type 7701

PLAYBACK
- Single or repeated
- Calibrated or uncalibrated
- Synchronised (original vs. edited)

JURY TEST
- Test methods supported: Paired Comparison, Semantic Differential
- Voice annotation
- User-defined delays
- Play List

ON-LINE DOCUMENTATION
- On-line, context-sensitive Help and User Manual

Data Acquisition Hardware - Bruel & Kjaer B-Frame Type 3560 B

SYSTEM DATA

IDA^e Hardware Configurations for PULSE
— Types 3560-B, 3560-C, 3560-D and 3560-E

PULSE™ is a versatile, task-oriented sound and vibration analysis system. It provides the platform for a range of PC-based measurement solutions from Brüel & Kjær. A PULSE system consists of a PC with LAN interface, PULSE software, Microsoft® Windows® operating system, Microsoft® Office, and data acquisition front-end hardware. Up to 10 front-ends can be combined into one measurement system with more than 300 input channels.

This System Data describes the hardware available for Data Acquisition Front-ends Types 3560-B, C, D and E.

PULSE Software as well as PULSE Pocket Analyzer Type 3560-L and the PULSE Lite software are described separately.

050149

USES AND FEATURES

USES

- Multiframe systems comprising up to 10 front-ends with synchronous sampling between front-ends for real-time measurements on more than 300 channels:
  - Type 3560-B: 5 input and 1 output channel
  - Type 3560-C: 2 modules. Up to 17 input and/or 3 generator output channels
  - Type 3560-D: 7 modules. Up to 65 input and/or 10 generator output channels
  - Type 3560-E: 10 modules. Up to 96 input and/or 16 generator output channels)
- Signal and system analysis using all PULSE application packages for, or example:
  - Time data acquisition
  - General noise and vibration measurements
  - Basic and advanced acoustics
  - Structural Analysis
  - Machine Diagnostics
  - Electroacoustic testing

FEATURES

- Dyn-X input modules with single, 160 dB input range
- Automatic detection of front-end hardware and transducers – supports IEEE 1451.4-capable transducers with TEDS (Transducer Electronic Data Sheet)
- Fully conditioned input and output channels for microphones and accelerometers, charge transducers, CCLD transducers and other transducers acting as voltage sources
- Full overload detection including out-of-band overload and indication of incorrect conditioning
- LAN interface allows the front-end to be placed close to the test object and reduces transducer cable length
- Rugged design for industrial use
- Battery (3560-B, C only)/external DC operated acquisition unit for field use
- Low-noise operation

Brüel & Kjær

Introduction

PULSE is a versatile, task-oriented system for noise and vibration analysis. It provides the platform for a range of PC-based measurement solutions from Brüel & Kjær.

A PULSE system consists of a PC with LAN interface, PULSE software, Windows® 2000, XP or Windows Vista®, Microsoft® Office and IDA^e-based data acquisition front-end hardware. A system can contain more than 300 input channels located in up to 10 front-ends. The input/output conditioning modules perform signal conditioning and digitise the transducer signals. The IDA^e modules available for use in PULSE systems are shown in Fig. 1 and listed in the Ordering Information on page 23. Modules can be freely mixed in a single front-end or in a multiframe system. Further information on the controller and input/output modules is given in Table 1.

Fig. 1
Overview of the components available for use in a PULSE System with LAN Interface

Standard configurations for a wide variety of applications are described in the "PULSE Analyzers & Solutions" Catalogue both printed (BF 0209) and on www.bksv.com

For information on PULSE Pocket Analyzer Type 3560-L, see the separate Product Data (BP 1967)

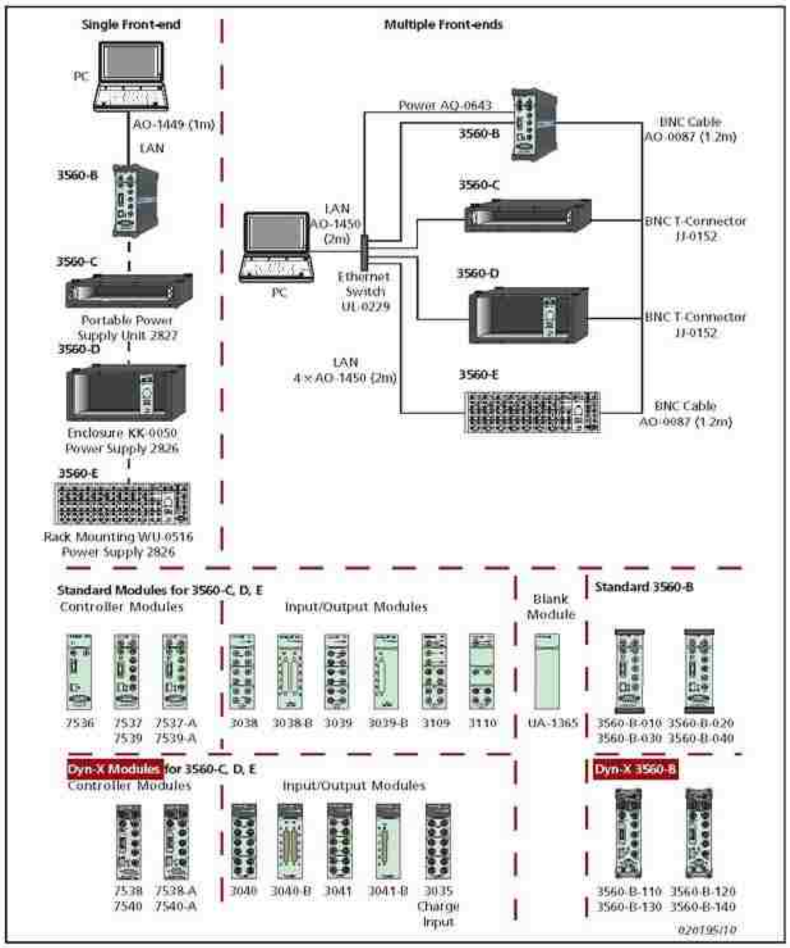


Table 1 Types and modules comprising PULSE front-ends

<table>
<tr><th>Type</th><th>Product Name</th><th>Frequency Range</th><th>Aux. Channels</th><th>Simultaneous Channels</th><th>Connectors</th><th>Input Type</th><th colspan="3"></th></tr>
<tr><td colspan="10">Type 3560-B</td></tr>
<tr><td>3560-B-010</td><td rowspan="8">5-channel PULSE Data Acquisition Unit</td><td rowspan="8">0 Hz to 25.6 kHz</td><td rowspan="8">16 Aux Input[d] (10 samples/s) 2 Digital Output</td><td rowspan="4">5 Input 1 Sine Output</td><td rowspan="2">LEMO</td><td rowspan="2">Direct/CCLD[a, b]/Mic. Preamp. 1 Tacho Conditioning[c]</td><td rowspan="8" colspan="3"></td></tr>
<tr><td>3560-B-110</td></tr>
<tr><td>3560-B-020</td><td rowspan="2">BNC</td><td rowspan="2">Direct/CCLD[a] 1 Tacho Conditioning[c]</td></tr>
<tr><td>3560-B-120</td></tr>
<tr><td>3560-B-030</td><td rowspan="4">5 Input 1 Generator Output</td><td rowspan="2">LEMO</td><td rowspan="2">Direct/CCLD[a, b]/Mic. Preamp. 1 Tacho Conditioning[c]</td></tr>
<tr><td>3560-B-130</td></tr>
<tr><td>3560-B-040</td><td rowspan="2">BNC</td><td rowspan="2">Direct/CCLD[a] 1 Tacho Conditioning[c]</td></tr>
<tr><td>3560-B-140</td></tr>
<tr><td colspan="7">Types 3560-C, D, E</td><td>C</td><td>D</td><td>E</td></tr>
<tr><td>3109</td><td>Generator, 4/2-ch. Input/ Output Module</td><td>0 Hz to 25.6 kHz</td><td rowspan="10">–</td><td>4 Input 2 Generator Output</td><td rowspan="2">BNC and LEMO</td><td rowspan="2">Direct/CCLD[a]/Mic. Preamp. 1 Tacho Conditioning[c]</td><td rowspan="12">1 of these modules Up to 17 input channels</td><td rowspan="12">Up to 5 of these modules Up to 65 input channels</td><td rowspan="12">Up to 8 of these modules Up to 96 input channels</td></tr>
<tr><td>3110</td><td>Generator, 2/1-ch. Input/ Output Module</td><td>0 Hz to 204.8 kHz</td><td>2 Input 1 Generator Output[e]</td></tr>
<tr><td>3038</td><td rowspan="4">12-ch. Input Module</td><td rowspan="8">0 Hz to 25.6 kHz</td><td rowspan="4">12 Input</td><td rowspan="2">BNC</td><td rowspan="2">Direct/CCLD[a] 2 Tacho Conditioning[b]</td></tr>
<tr><td>3040</td></tr>
<tr><td>3038-B</td><td rowspan="2">2 × Sub-D</td><td rowspan="2">Direct/CCLD[a]/Mic. Preamp.[b]</td></tr>
<tr><td>3040-B</td></tr>
<tr><td>3039</td><td rowspan="4">6-ch. Input Module</td><td rowspan="4">6 Input</td><td rowspan="2">BNC and LEMO</td><td rowspan="2">Direct/CCLD[a]/Mic. Preamp 1 Tacho Conditioning[c]</td></tr>
<tr><td>3041</td></tr>
<tr><td>3039-B</td><td rowspan="2">Sub-D</td><td rowspan="2">Direct/CCLD[a]/Mic. Preamp.[b]</td></tr>
<tr><td>3041-B</td></tr>
<tr><td>3035</td><td>6-ch. Charge & CCLD Input Module</td><td>0 Hz to 25.6 kHz</td><td>–</td><td>6 Input</td><td>BNT/BNC and TNC</td><td>Charge/Direct/CCLD[a] Tacho Conditioning on BNT Connector</td></tr>
<tr><td>UA-1365</td><td>Blank Module</td><td colspan="5">–</td></tr>
<tr><td>7536</td><td>Controller Module</td><td>–</td><td rowspan="9">16 Aux Input[d] (10 samples/s) 2 Digital Output</td><td colspan="3">–</td><td rowspan="9">1 of these modules</td><td rowspan="9">1 of these modules</td><td rowspan="9">1 of these modules</td></tr>
<tr><td>7537</td><td rowspan="8">5/1-ch. Input/Output Controller Module</td><td rowspan="8">0 Hz to 25.6 kHz</td><td rowspan="2">5 Input 1 Sine Output</td><td rowspan="4">LEMO</td><td rowspan="4">Direct/CCLD[a, b]/Mic. Preamp. 1 Tacho Conditioning[c]</td></tr>
<tr><td>7538</td></tr>
<tr><td>7539</td><td rowspan="2">5 Input 1 Generator Output</td></tr>
<tr><td>7540</td></tr>
<tr><td>7537-A</td><td rowspan="2">5 Input 1 Sine Output</td><td rowspan="4">BNC</td><td rowspan="4">Direct/CCLD[a] 1 Tacho Conditioning[c]</td></tr>
<tr><td>7538-A</td></tr>
<tr><td>7539-A</td><td rowspan="2">5 Input 1 Generator Output</td></tr>
<tr><td>7540-A</td></tr>
</table>

a. Constant Current Line Drive for DeltaTron® and ICP® Accelerometers or Microphone Preamplifier
b. Using adaptor cables
c. All input channels can be used for tachometer operation
d. Only 12-channel currently supported in PULSE software
e. Upper frequency @ 102.4 kHz

Dyn-X modules – See "Dyn-X Modules – Types 7538/38-A, 7540/40-A, 3035, 3040/40-B, 3041/41-B, 3560-B-110/120/130/140" on page 6.

PULSE Type 3560-B – Compact Data Acquisition Unit, up to 5 Input Channels

FEATURES

- Compact, robust casing for industrial and hard everyday use
- Battery operated (5 hours continuous) or DC powered (10 – 32 V)
- Silent operation to 35°C
- Cooling fans can be turned off for silent operation (will automatically restart if too hot)
- Synchronous sampling with other PULSE front-ends

030150

Type 3560-B is a compact data acquisition system for battery/DC powered operation. The unit handles communication with the PC, measurement input and provides a sample clock. Eight versions are available, four standard and four Dyn-X – see the upper portion of Table 1.

A handle, UA-1689, is available for mounting on top of Type 3560-B, making it easier to carry.

Compliance with Standards

(For environmental specifications and compliance with standards for PCs, see the specifications given by their respective manufacturers)

TYPES 3560-B-010, -020, -030, -040, -110, -120, -130, -140,
TYPES 3560-C, 3560-D AND 3560-E WITH CONTROLLER MODULE TYPE 7536,
INPUT/OUTPUT CONTROLLER MODULE TYPE 7537, 7537-A, 7538, 7538-A, 7539, 7539-A, 7540 OR 7540-A
INPUT/OUTPUT MODULE TYPE 3035, 3038, 3038-B, 3039, 3039-B, 3040, 3040-B, 3041, 3041-B, 3109 OR 3110

CE, C-Tick	CE-mark indicates compliance with: EMC Directive and Low Voltage Directive. C-Tick mark indicates compliance with the EMC requirements of Australia and New Zealand.
Safety	EN/IEC 61010–1: Safety requirements for electrical equipment for measurement, control and laboratory use. UL 61010B–1: Standard for Safety – Electrical measuring and test equipment.
EMC Emission	EN/IEC 61000–6–3: Generic emission standard for residential, commercial and light industrial environments. EN/IEC 61000–6–4: Generic emission standard for industrial environments. CISPR 22: Radio disturbance characteristics of information technology equipment. Class B Limits. FCC Rules, Part 15: Complies with the limits for a Class B digital device.
EMC Immunity	EN/IEC 61000–6–1: Generic standards – Immunity for residential, commercial and light industrial environments. EN/IEC 61000–6–2: Generic standards – Immunity for industrial environments. EN/IEC 61326: Electrical equipment for measurement, control and laboratory use – EMC requirements. **Note:** The above is only guaranteed using accessories listed in this System Data.
Temperature	IEC 60068–2–1 & IEC 60068–2–2: Environmental Testing. Cold and Dry Heat. Operating Temperature: –10 to +50°C (14 to 122°F) Storage Temperature: –25 to +70°C (–13 to 158°F)
Humidity	IEC 60068–2–78: Damp Heat: 93% RH (non-condensing at 40°C (104°F))
Mechanical	Operating (peak values) MIL–STD–810C: Vibration: 12.7 mm, 15 ms^{-2}, 5 – 500 Hz Non-operating: IEC 60068–2–6: Vibration: 0.3 mm, 20 ms^{-2}, 10 – 500 Hz IEC 60068–2–27: Shock: 1000 ms^{-2} IEC 60068–2–29: Bump: 1000 bumps at: 250 ms^{-2}
Enclosure	IEC 60529: Protection provided by enclosures: **3560-B:** IP 40; **3560-C:** IP 32; **3560-D:** IP 40; **3560-E:** IP 20

EFFECT OF RADIATED/CONDUCTED RF, MAGNETIC FIELD AND VIBRATION
Radiated RF: 80 – 1000 MHz, 80% AM 1 kHz, 10 V/m
Conducted RF: 0.15 – 80 MHz, 80% AM 1 kHz, 10 V
Magnetic Field: 30 A/m, 50 Hz
Vibration: 5 – 500 Hz, 12.7 mm, 15 m/s^2
Input measured in 7.071 mV range with shorted input. All values are RMS. Conducted RF immunity on all channels is only guaranteed using an external connection from measuring ground to chassis terminal on Types 2826 or 2827

Input/Output	Radiated RF	Conducted RF	Magnetic Field	Vibration
Direct/CCLD	<10 μV	<130 μV	<4 μV	<80 μV
Preamplifier	<10 μV	< 25 μV	<8 μV	<80 μV
Generator	<60 μV	< 25 μV	<4 μV	< 5 μV
Charge	<130 fC	<130 fC	<10 fC	<80 fC

Specifications – PULSE Types 3560-B/C/D/E

Multi-analyzer Systems Type 3560-B, 3560-C, 3560-D and 3560-E with LAN interface are modular, expandable, multi-analysis systems that include the following components:

- Pentium® PC
- PULSE software
- Microsoft® Windows® 2000 or Windows® XP or Windows Vista® operating system
- Microsoft® Office 2000, 2003, 2007 or XP
- Front-end comprising: Power Supply/Frame, Controller Module and a number of Input/Output Modules (see below)

Specifications – Portable PULSE Type 3560-B

POWER REQUIREMENTS
Fulfils the requirements of ISO 7637–1 and 7637–2 with batteries
Voltage: 10 – 32 V DC
Power Consumption:
Nominal: 14 W
Max.: 26 W (while charging battery)
Ext. Power Connector: LEMO coax., FFA.00.113, ground on shield

BATTERIES
Optional Accessories: 2 × DR35 NiMH or NI 1030, 10.8 V (nominal)
Working Time (Continuous): 5 hours
Charging Time: 5 hours/battery

ACOUSTIC NOISE EMISSION (at 1 m)
Silent operation to 35°C (95°F) when not charging batteries. When charging batteries, fan operation may start at a lower ambient temperature

Sound Calibrator - Bruel & Kjaer Type 4231

PRODUCT DATA

Sound Calibrator — Type 4231

Sound Calibrator Type 4231 is a handy, portable sound source for calibration of sound level meters and other sound measurement equipment. The calibrator is very robust and stable, and conforms to EN/IEC 60942 Class LS and Class 1, and ANSI S1.40-1984.

USES AND FEATURES

USES

- Calibration of sound level meters and other sound measurement equipment

FEATURES

- Conforms to EN/IEC 60942 (2003) Class LS and Class 1, and ANSI S1.40 – 1984
- Robust, pocket-sized design with highly stable level and frequency
- Calibration accuracy ± 0.2 dB
- 94 dB SPL, or 114 dB SPL for calibration in noisy environments
- Extremely small influence of static pressure and temperature
- Sound pressure independent of microphone equivalent volume
- 1 kHz calibration frequency for correct calibration level independent of weighting networks
- Fits Brüel & Kjær 1″ and 1/2″ microphones (1/4″ and 1/8″ microphones with adaptor)
- Switches off automatically when removed from the microphone

Brüel & Kjær

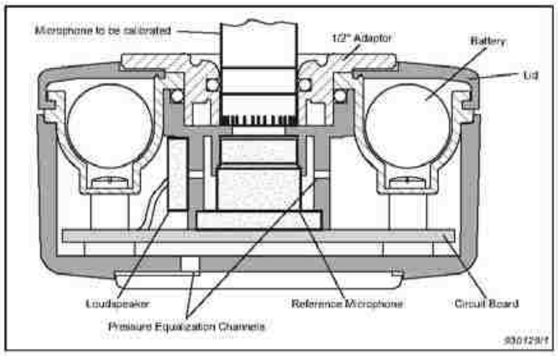
Sound Calibrator Type 4231

Sound Calibrator Type 4231 is a pocket-sized, battery operated sound source for quick and direct calibration of sound level meters and other sound measuring systems. It fits Brüel & Kjær 1″ microphones and using the removable adaptor, 1/2″ microphones. With optional adaptors, it can be used for 1/4″ and 1/8″ microphones as well.

The calibration frequency is 1000 Hz (the reference frequency for the standardised international weighting networks), so the same calibration value is obtained for all weighting networks (A, B, C, D and Linear). The calibration pressure of 94 ±0.2 dB re 20 µPa is equal to 1 Pa or 1 N/m^2. The +20 dB level step gives 114 dB SPL, which is convenient for calibration in noisy environments, or for checking linearity.

The design of Type 4231 is based on a feed-back arrangement to ensure a highly stable sound pressure level and ease of use. The feed-back loop uses a condenser microphone (see Fig. 1), which is specially developed for this purpose.

Fig. 1
Cross-sectional view of Sound Calibrator Type 4231. The feed-back loop is based on a high-quality condenser microphone to ensure a very stable sound pressure level



This microphone is optimised to have extremely high stability and independence of variations in static pressure and temperature around the 1 kHz calibration frequency. The result of this is a user-friendly calibrator where exact fitting of the microphone is non critical and the effects of changes in temperature and static pressure are negligible.

Fig. 2
Type 4231 fitted to Hand-held Analyzer Type 2250. The calibrator's centre of gravity is positioned very close to the microphone, giving a stable set-up

The calibrator gives a continuous sound pressure level when fitted on a microphone (see Fig. 2) and activated.

The sensitivity of the sound measuring equipment can then be adjusted until it indicates the correct sound pressure level.

The calibrator is automatically switched off when removed from the microphone.

A leather protecting case, which does not need to be removed to use the calibrator, is supplied.

Compliance with Standards

CE C-Tick	CE-mark indicates compliance with: EMC Directive and Low Voltage Directive. C-Tick mark indicates compliance with the EMC requirements of Australia and New Zealand.
Safety	EN/IEC 61010–1: Safety requirements for electrical equipment for measurement, control and laboratory use. ANSI/UL 61010–1: Safety requirements for electrical equipment for measurement, control and laboratory use.
EMC Emission	EN/IEC 61000–6–3: Generic emission standard for residential, commercial and light industrial environments. EN/IEC 61000–6–4: Generic emission standard for industrial environments. CISPR 22: Radio disturbance characteristics of information technology equipment. Class B Limits. FCC Rules, Part 15: Complies with the limits for a Class B digital device. EN/IEC 60942: Instrumentation Standard – Electroacoustics – Sound Calibrators.
EMC Immunity	EN/IEC 61000–6–1: Generic standards – Immunity for residential, commercial and light industrial environments. EN/IEC 61000–6–2: Generic standards – Immunity for industrial environments. EN/IEC 61326: Electrical equipment for measurement, control and laboratory use – EMC requirements. EN/IEC 60942: Instrumentation Standard – Electroacoustics – Sound Calibrators. **Note:** The above is only guaranteed using accessories listed in this Product Data sheet.
Temperature	IEC 60068–2–1 & IEC 60068–2–2: Environmental Testing. Cold and Dry Heat. Operating Temperature: –10 to +50°C (14 to 122°F) Storage Temperature: –25 to +70°C (–13 to +158°F)
Humidity	IEC 60068–2–78: Damp Heat: 90% RH (non-condensing at 40°C (104°F)).
Mechanical	Non-operating: IEC 60068–2–6: Vibration: 0.3 mm (10 to 58 Hz), 20 m/s^2 (58–500 Hz) IEC 60068–2–27: Shock: 1000 m/s^2 IEC 60068–2–29: Bump: 3000 bumps at 400 m/s^2
Enclosure	IEC 60529: Protection provided by enclosures: IP 50 with leather protection case.

Specifications – Sound Calibrator Type 4231

STANDARDS SATISFIED
EN/IEC 60942 (2003), Class LS and Class 1, Sound Calibrators
ANSI S1.40–1984, Specification for Acoustic Calibrators

SOUND PRESSURE LEVELS
94.0 dB ±0.2 dB (Principal SPL) or
114.0 dB ±0.2 dB re 20 μPa at reference conditions

FREQUENCY
1 kHz ±0.1%

SPECIFIED MICROPHONE
Size according to IEC 61094-4:
- 1″ without adaptor
- 1/2″ with adaptor UC-0210 (supplied)
- 1/4″ with adaptor DP-0775 (optional)
- 1/8″ with adaptor DP-0774 (optional)

EQUIVALENT FREE-FIELD LEVEL
(0° incidence, re Nominal Sound Pressure Level)
–0.15 dB for 1/2″ Brüel & Kjær Microphones. See Type 4231 User Manual for other microphones

EQUIVALENT RANDOM INCIDENCE LEVEL
(re Nominal Sound Pressure Level)
+0.0 dB for 1″, 1/2″, 1/4″ and 1/8″ Brüel & Kjær Microphones

NOMINAL EFFECTIVE COUPLER VOLUME
> 200 cm^3 at reference conditions

DISTORTION
< 1%

LEVEL STABILITY
Short-term: Better than 0.02 dB (as specified in IEC 60942)
One Year: Better than 0.05 dB (σ = 96%)
Stabilization Time: <5 s

REFERENCE CONDITIONS
Temperature: 23°C ±3°C (73° ±5°F)
Pressure: 101 ±4 kPa
Humidity: 50%, –10% +15% RH
Effective Load Volume: 0.25 cm^3

ENVIRONMENTAL CONDITIONS
Pressure: 65 to 108 kPa
Humidity: 10 to 90% RH (non-condensing)
Effective Load Volume: 0 to 1.5 cm^3

INFLUENCE OF ENVIRONMENTAL CONDITIONS (Typical)
Temperature Coefficient: ±0.0015 dB/°C
Pressure Coefficient: $+8 \times 10^{-4}$ dB/kPa
Humidity Coefficient: 0.001 dB/% RH

POWER SUPPLY
Batteries: 2 × 1.5 V IEC Type LR6 ("AA" size)
Lifetime: Typically 200 hours continuous operation with alkaline batteries at 23°C (73°F)
Battery Check: When Type 4231 stops working continuously, and only operates when the On/Off button is held in, the batteries should be replaced

DIMENSIONS AND WEIGHT
(Without case)
Height: 40 mm (1.5″)
Width: 72 mm (2.8″)
Depth: 72 mm (2.8″)
Weight: 150 g (0.33 lb.), including batteries

Note: All values are typical at 25°C (77°F), unless measurement uncertainty or tolerance field is specified. All uncertainty values are specified at 2σ (i.e., expanded uncertainty using a coverage factor of 2)

Calibrator Exciter - Bruel & Kjaer Type 4294

PRODUCT DATA

Calibration Exciter — Type 4294 and Type 4294-002

USES

- Precise field calibration of vibration transducers
- Rapid calibration and checking
- Quick, easy field calibration of vibration measurement and recording systems

FEATURES

- Small, lightweight, and battery-driven
- Leather case for impact protection
- Designed for everyday use in harsh environments
- Acceleration, velocity and displacement calibration
- High-precision, crystal-controlled servo operating at 159.15 Hz (1000 rad s^{-1})
- Drop- and environment-tested according to IEC 60068
- Splash-proof according to IP 54 (IEC 60529)

Description

Type 4294 permits accurate adjustment of measuring instrumentation at a standard acceleration level of 10 ms^{-2} (0 – 70 g load). The reference signal may additionally be used for velocity and displacement calibration, at 10 mms^{-1} and 10 µm respectively.

Type 4294-002 permits accurate adjustment of measuring instrumentation at a standard acceleration level of 3.16 ms^{-2} (0 – 200 g load). The reference signal may additionally be used for velocity and displacement calibration, at 3.16 mms^{-1} and 3.16 µm respectively.

The calibrator embodies an electromagnetic exciter driven by a crystal oscillator at a frequency of 159.15 Hz (1000 rad s^{-1}). Servo feedback via a small accelerometer on the underside of the vibration table is used to maintain a constant and accurate vibration level independent of the mass of the transducer under test (70 g for Type 4294 and 200 g for Type 4294-002).

To prevent overload, power for the calibrator is automatically disconnected if a transducer mass above a certain level is mounted on the table (70 g for Type 4294 and 200 g for Type 4294-002).

Use of the calibrator is very straightforward. The transducer is conveniently attached to the calibrator table using a 10–32 UNF Steel Stud (YQ 2963). Alternatively, the 10 g Mounting Disc (DB 2996) supplied, provides a convenient means of attaching transducers manufactured with 3 mm threads or those fitted with Mounting Magnet UA 0642. The mounting disc also permits the attachment of transducers with either beeswax or cyanoacrylate adhesive.

The calibrator is actuated by pressing the small button on the side of its housing. Following system adjustment, the calibrator is switched off by pressing the button a second time. To prolong the useful life of its built-in battery, Type 4294 automatically switches off after approximately 100 seconds.

Brüel & Kjær

Compliance with Standards

CE, C-Tick	CE-mark indicates compliance with: EMC Directive and Low Voltage Directive. C-Tick mark indicates compliance with the EMC requirements of Australia and New Zealand
Safety	EN/IEC 61010–1: Safety requirements for electrical equipment for measurement, control and laboratory use. UL 61010B–1: Standard for Safety – Electrical measuring and test equipment
EMC Emission	EN/IEC 61000–6–3: Generic emission standard for residential, commercial and light industrial environments. EN/IEC 61000–6–4: Generic emission standard for industrial environments. CISPR 22: Radio disturbance characteristics of information technology equipment. Class B Limits. FCC Rules, Part 15: Complies with the limits for a Class B digital device.
EMC Immunity	EN/IEC61000–6–1: Generic standards – Immunity for residential, commercial and light industrial environments. EN/IEC 61000–6–2: Generic standards – Immunity for industrial environments. EN/IEC 61326: Electrical equipment for measurement, control and laboratory use – EMC requirements. **Note:** The above is only guaranteed using accessories listed in this Product Data sheet.
Temperature	IEC 60068–2–1 & IEC 60068–2–2: Environmental Testing. Cold and Dry Heat. Operating Temperature: +10 to +40°C (50 to 104°F) for 10 ms^{-2} reference within ± 3% and 3.16 ms^{-2} reference within ± 3% –10 to +55°C (14 to 131°F) for 10 ms^{-2} reference within ± 5% and 3.16 ms^{-2} reference within ± 5% Storage Temperature: –25 to +70°C (–13 to 158°F) IEC 60068–2–14: Change of temperature: –10 to +55°C (2 cycles, 1°C/min)
Humidity	IEC 60068–2–78: Damp Heat: 90% RH (non-condensing at 30°C (86°F))
Mechanical	Non-operating: IEC 60068–2–6: Vibration: 0.3 mm, 20 m/s^2, 10–500 Hz IEC 60068–2–27: Shock: 1000 m/s^2 IEC 60068–2–29: Bump: 1000 bumps at 400 m/s^2
Enclosure	IEC 60529: Protection provided by enclosures: IP 54

Specifications – Calibration Exciters Types 4294 and 4294-002

	4294	4294-002
Dynamic Characteristics		
Frequency (Hz)	159.15 ± 0.02%	
Acceleration (ms^{-2} (RMS))	10 ± 3%	3.16 ± 3%
Velocity (mms^{-1} (RMS))	10 ± 3%	3.16 ± 3%
Displacement (μm)	10 ± 3%	3.16 ± 3%
Transverse Amplitude	< 5% of main axis amplitude	
Distortion	4294:< 2% for 10 to 70 g 4294-002: < 2% for 10 to 200 g load 4294 & 4294-002: typical < 7% for 0 to 10 g. Use DB 2699 (10 g) with very light accelerometers to achieve 2% distortion	
Power Requirements		
Built-in Battery	One 9 V Alkaline Battery QB 0016 (IEC type 6LR61)	
Battery Life	Approx. 200 calibrations, each lasting 100 s with automatic switching off at the end of each calibration	
Warm-up Time (Seconds)	< 5	
Signal Duration (Seconds)	103 ± 1 s with automatic stop	
Long-term Stability	Better than 1% per year for acceleration, velocity and displacement; better than 10 ppm per year for frequency	
Physical Characteristics		
Length	155 mm (6.1 in)	
Diameter	52 mm (2.05 in)	
Weight	500 g (17.6 oz) including battery and leather case	
Transducer Mounting		
Maximum Load (g)	70	200
Mounting Torque (Nm)	max. 0.5	
Mounting Thread	10–32 UNF	

Ordering Information

Types 4294 and 4294-002 include the following accessories:

- Leather Case
- 9 V Battery
- 10–32 UNF Steel Stud
- Mounting Disc Adaptor
- Calibration Chart

OPTIONAL ACCESSORIES

4294-CAI	Accredited Initial Calibration
4294-CAF	Accredited Calibration
4294-EW1	4294 Calibration Exciter Extended Warranty, one year extension
4294-002-CAI	Accredited Initial Calibration
4294-002-CAF	Accredited Calibration
4294-002-EW1	4294 Calibration Exciter Extended Warranty, one year extension

RE-CALIBRATION

Periodic re-calibration of Type 4294 is recommended in order to maintain the high accuracy of the vibration unit, and in order to have proof of traceability. Depending on the application, a re-calibration every 1 – 3 years is recommended.

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Local representatives and service organisations worldwide

Brüel & Kjær

BP 2101–11 0506 Rosendahls Bogtrykkeri

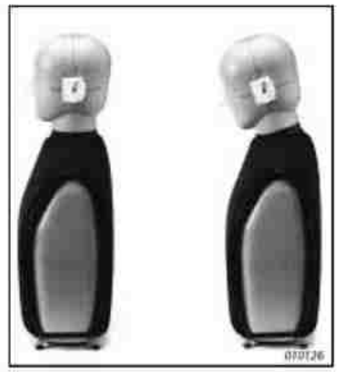
HATS - Bruel & Kjaer Type 4100

PRODUCT DATA

Sound Quality Head and Torso Simulator — Types 4100 and 4100 D

USES

- Recording vehicle noise for sound quality evaluation and testing
- Recording noise from domestic appliances, office equipment, etc., for sound quality optimisation
- Recording noise from sub-suppliers' products and components to evaluate and optimise their sound quality
- Evaluation of headphones, and hearing protectors where a blocked ear canal is desired
- Binaural sound and music recording



FEATURES

- Directivity optimised for sound-image localisation
- Type 4100 includes Falcon Range® Preamplifiers Type 2669 L with CIC facility
- Type 4100 D includes DeltaTron® Preamplifiers Type 2671
- High sensitivity, low noise, ½" Falcon microphones
- IEEE P1451.4-capable transducers with TEDS (Transducer Electronic Data Sheet)
- Manikin with surfaces and pinnae modelling the geometry of the average adult head and torso
- ITU–T compliance with the acoustic requirements of ITU–T Rec. P.58, IEC 959 and ANSI S3 36–1985, except for exclusion of the ear canal
- Adjustable neck angle
- Light and robust
- Accredited calibration available

General

Sound Quality Head and Torso Simulators Types 4100 and 4100 D are manikins designed for sound quality testing.

Two microphones, positioned at the entrances to the ear canals on the manikin's head, simulate the spatial separation from ear to ear of a human head and ensure a signal that includes the interference patterns caused by the head and upper body. This gives an extremely accurate three-dimensional recording.

Two moulded-silicone pinna simulators sit around the microphones to provide directivity patterns similar to the human ear.

The simulator has a sound-dampening fabric cover which slips easily over the manikin's neck. This assists in changing the reflections from the body and shoulders to obtain the correct directivity.

The position of the head can be adjusted by turning the neck ring so that the head looks straight forward or slightly down at an angle of 17°.

Microphones are easily installed or removed by screwing or unscrewing them from the ear cavities.

Types 4100 and 4100 D contain IEEE P1451.4-capable transducers with standardised Transducer Electronic Data Sheets (TEDS). This feature allows automatic front-end and analyzer setup, based on information stored in the transducer. This information includes, for example, sensitivity, serial number, manufacturer and calibration date.

Sound Quality

The sound quality of the noise from a product, as perceived by a human being, is an increasingly important factor when assessing the total quality of the product.

This applies to all forms of transport: vehicles, aircraft, trains and ships. Household and office machinery products

4100, 4100 D

Brüel & Kjær

are also increasingly subject to the optimisation of their sound quality.

Sub-suppliers of products and components to the above-mentioned industries are often required to include an acceptable sound quality as a part of the product specifications.

Subjective Listening Tests

The final evaluation of the sound quality of a product is normally made using a selected group of people – a jury in a listening test.

To have the jury listen to the sound in reality, for example each jury member driving a car and then reporting on the sound quality, is very time consuming and costly. To overcome this, Type 4100 can be used to make a high-quality binaural recording of the product's noise on the hard disk of a portable PC with high-quality sound card. This can then be simultaneously presented to all members of the jury off-site.

However, to avoid bias errors in this process, it is important that the acoustic properties of the recording and playback are as accurate as possible. Types 4100 and 4100 D have therefore been designed to have a frequency response to sounds coming from all directions which closely approximates the direction-dependent human response, and to have inter-aural time differences very close to those of the average person.

System for Sound Quality Optimisation

Quite often, the first evaluation of the sound quality of a product, as perceived by the jury, is not satisfactory. Therefore, the recorded signals from Types 4100 and 4100 D can be modified using a wide range of time/frequency domain editing techniques using a sound quality software program and a PC. The modified signals can then be compared with the original, by the jury, in a listening test. If the modified signal is preferred, information on the changes in the noise can be used by the product designer to obtain – by physical changes – improved sound quality.

Specifications – Type 4100

MICROPHONES AND PREAMPLIFIERS
Two Type 4190–L–002 microphone/preamplifier assemblies with built-in TEDS, each comprising a ½" Falcon Range Microphone Type 4190* placed in the bottom of the concha, and Falcon series Preamplifier Type 2669 L* with charge injection calibration (CIC) facility and LEMO connector
Microphone Sensitivity: 50 mV/Pa. Individually calibrated
Upper Limit of Dynamic Range: 148 dB SPL at 3% distortion
Max. Sound Pressure Level:
159 dB peak with Preamplifier Type 2669 and mains driven power supplies
138 dB peak with Preamplifier Type 2669 and battery power supplies
Preamp. Lower Limiting Frequency: <2 Hz (–3 dB)

Specifications – Type 4100 D

MICROPHONES AND PREAMPLIFIERS
Two Type 4189–A–002 microphone/preamplifier assemblies with built-in TEDS, each comprising a ½" Falcon Range Microphone Type 4189* placed in the bottom of the concha and a DeltaTron Preamplifier Type 2671* with BNC connector
Microphone Sensitivity: 50 mV/Pa. Individually calibrated
Upper Limit of Dynamic Range: 146 dB SPL at 3% distortion
Max. Sound Pressure Level: 138 dB peak with DeltaTron Preamplifier Type 2671
Preamp. Lower Limiting Frequency: <12 Hz (–3 dB)

* See separate Product Data for details

Common Specifications – Types 4100, 4100 D

PINNA SIMULATOR
Dimensions similar to those specified in ITU–T Rec. P.58, IEC 959 and ANSI S3, 36–1985, except for the ear canal extensions

HEAD AND TORSO SHAPES
The main dimensions comply with the dimensional requirements of ITU–T Rec. P.58 and the reports from IEC 959 and ANSI S3 36–1985

SHOULDER DAMPING FABRIC
The shoulders, chest and back are covered with a damping fabric to adjust diffraction. The fabric has a minimum of 10% absorption in the range of 100 Hz to 20 kHz

LEFT/RIGHT EAR TRACKING
±1 dB up to 5 kHz
±3 dB up to 8 kHz

CALIBRATION
Sensitivity calibration can be made using a calibrator or pistonphone with Calibration Adaptor DP 0887

DIMENSIONS AND WEIGHT
Head Height: 700 mm (27.6")
Torso: 480×440×210 mm (18.9×17.3×8.3")
Weight: 7.9 kg (17.4 lb.)

CE CE-mark indicates compliance with EMC Directive and Low Voltage Directive. (See also Microphone and Preamplifier Product Data)

Ordering Information

Types 4100 and 4100 D Sound Quality Head and Torso Simulator
Include the following accessories:

BC 0200:	Calibration Chart
DP 0887:	Calibration Adaptor
UA 1043:	Support Leg
UA 1052:	Handle
UC 5290:	Tripod Mounting Adaptor

CALIBRATION OPTIONS

CAI 4100	Accredited Initial Calibration
CAI 4100 D	Accredited Initial Calibration
CAF 4100	Accredited Calibration
CAF 4100 D	Accredited Calibration
TCF 4100	Conformance Test
TCF 4100 D	Conformance Test

BP 1436–13 01/08

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Ireland (01)803 7600 · Italy 02 57 68061 · Japan 03-3779-8671 · Republic of Korea (02)3473-0605 · Netherlands (31)318 559290 · Norway 66771155
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PRODUCT DATA

½″ Prepolarized Free-field Microphone — Type 4189

Type 4189 is designed for high-precision, free-field measurements where a microphone with high sensitivity is required. Being prepolarized, Type 4189 can be used with both DeltaTron® and classical preamplifiers.

080143

USES
- Precision sound measurement
- Premium class sound level meters
- Equipment complying with IEC 61672 class 1

FEATURES
- Sensitivity: 50 mV/Pa
- Frequency: 6.3 Hz – 20 kHz
- Dynamic Range: 14.6 – 146 dB
- Temperature: –30 to +150°C (–22 to +302°F)
- Polarization: Prepolarized

Use of Free-field Microphones

At higher frequencies, reflections and diffractions causes a pressure increase in front of the diaphragm of a microphone. If not corrected, for this would result in an increased output voltage. A free-field optimisation means that the frequency response of the microphone has been designed in such a way that the free-field response at 0 degrees incidence is flat. This microphone is optimised for use with the protection grid in place.

Free-field microphones are commonly used for sound measurement in an anechoic chamber or far away from reflecting buildings, etc. Another area for free-field microphones is for general electroacoustic measurements purposes like loudspeaker and microphone measurements.

Type 4189 is suited for use in class 1 Sound Level Meters and for all high-precision acoustic measurements where a robust and stable free-field microphone with an upper frequency of 20 kHz is required.

Manufacturing and Stability

A press-fitted, stainless-steel diaphragm ensures superior long-term stability and mechanical robustness – Type 4189 will withstand the 1 m drop test of IEC 60068–2–32.

All Brüel & Kjær Measuring Microphones are assembled in a clean room. This ensures that the microphones maintain their inherent low noise floor and high stability, even when used in environments with a combination of high humidity and high temperature.

Polarization Voltage

Being prepolarized, Type 4189 is especially well suited for battery operated equipment and operation in environments with high humidity.

TEDS Microphones

Type 4189 is available in TEDS combinations with either classical or DeltaTron type preamplifier. The TEDS microphone is considered one unit and has been sealed in a clean environment. The TEDS is programmed with the loaded sensitivity of the actual cartridge and the data is therefore readily available. The default TEDS template is to IEEE P1451.4 but TEDS to IEEE 1451.4 is available on request.

Individual Calibration Data

Each Type 4189 comes with an individual calibration chart including information about the open-circuit sensitivity, the frequency response in a free field as well as the electrostatic actuator response.

An enclosed mini-CD contains the individual calibration data at 1/12-octave frequencies plus a wealth of technical information, such as the influence of different accessories, response in different sound fields and much more. Using the CD data and the REq-X feature of PULSE™, a real-

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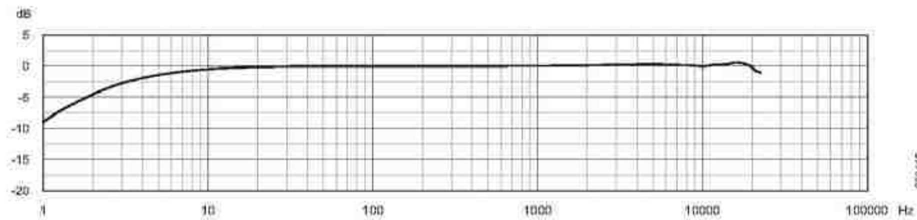
time correction for different measurement situations, can increase measurement accuracy.

***Fig. 1** Typical free-field response of the microphone with protection grid. The low-frequency response is valid when the vent is exposed to the sound field*



Specifications – ½″ Free-field Microphone Type 4189 (valid from serial number 2495387)

IEC 61094–4 Type Designation: WS2F
Polarization Voltage: 0 V (prepolarized)
Open-circuit Sensitivity (250 Hz)[a]: 50 mV/Pa, −26 dB ±1.5 dB re 1 V/Pa
0° Incidence Free-field Response[a]:
10 Hz to 8 kHz: ±1 dB
6.3 Hz to 20 kHz ±2 dB
Lower Limiting Frequency (–3 dB)[a]: 2 to 4 Hz
Pressure Equalization Vent: Rear vented
Diaphragm Resonance Frequency: 14 kHz (90° phase shift)
Cartridge Capacitance[a]: 14 pF at 250 Hz
Equivalent Air Volume: 46 mm^3 (250 Hz)
Pistonphone Correction (Type 4228 with DP-0776) 0.00 dB
Cartridge Thermal Noise: 14.6 dB(A), 15.3 dB(Lin)
Upper Limit of Dynamic Range (3% Distortion): >146 dB SPL[b]

a. Individually calibrated

Max. Sound Pressure Level: 158 dB (peak)

ENVIRONMENTAL
Operating Temperature Range: –30 to +150°C (–22 to +302°F)
Storage Temperature:
In Microphone Box: –30 to +70°C (–22 to +158°F)
With Mini-CD: 5 to 50°C (41 to 122°F)
Temperature Coefficient (250 Hz): −0.006 dB/K (–10 to +50°C, 14 to 122°F)
Pressure Coefficient: –0.01 dB/kPa
Operating Humidity Range: 0 to 100% RH (without condensation)
Influence of Humidity: <0.1 dB in the absence of condensation
Vibration Sensitivity (<1000 Hz): 62.5 dB equivalent SPL for 1 m/s^2 axial vibration

b. 137 dB (peak) with DeltaTron preamplifier and 24 V supply and 140 dB (peak) with ±15 V supply

Magnetic Field Sensitivity: 6 dB SPL for 80 A/m, 50 Hz field
Estimated Long-term Stability:
>1000 years/dB in dry air at 20°C (68°F)
>2 hours/dB in dry air at 150°C (302°F))
>40 years/dB in air at 20°C (68°F), 90% RH)
>1 year/dB in air at 50°C (122°F), 90% RH)

DIMENSIONS
Diameter with Grid: 13.2 mm (0.52″)
Diameter without Grid: 12.7 mm (0.50″)
Height with Grid: 17.6 mm (0.69″)
Height without Grid: 16.3 mm (0.64″)

Thread for Preamplifier Mounting: 11.7 mm–60 UNS

Note: All values are typical at 23°C (73.4°F), 101.3 kPa and 50% RH unless otherwise specified

CE Compliance with EMC Directive

Ordering Information

Type 4189	½″ Prepolarized Free-field Microphone

Includes the following accessories:
- BC-0224: Calibration Chart[c]
- BC-5002: Microphone Mini-CD[c]

TEDS COMBINATIONS

4189-A-021	½″ Free-field Microphone with Preamplifier Type 2671
4189-A-031	½″ Free-field Microphone with Preamplifier Type 2699
4189-B-001	½″ Free-field Microphone with Preamplifier Type 2669-B
4189-C-001	½″ Free-field Microphone with Preamplifier Type 2669-C
4189-L-001	½″ Free-field Microphone with Preamplifier Type 2669-L
4189-W-003	½″ Free-field Microphone with Preamplifier Type 2671-W-001

c. State microphone serial number if re-ordering calibration data

OPTIONAL ACCESSORIES

Type 2669	½″ Microphone Preamplifier
Type 2671	½″ DeltaTron Preamplifier
2671-W-001	½″ DeltaTron Preamplifier (version with LLF < 1.2 Hz)
Type 2699	½″ DeltaTron Preamplifier, A-weighted
Type 4231	Sound Calibrator
Type 4228	Pistonphone
Type 4226	Multifunction Acoustic Calibrator
DP-0776	Calibration Adaptor for ½″ Microphones
UA-0033	Electrostatic Actuator
UA-1260	½″ Angle Adaptor (approx. 80°)
UA-0386	Nose Cone for ½″ Microphone
UA-0237	Windscreen for ½″ microphone, 90 mm diameter
UA-0459	Windscreen for ½″ microphone, 65 mm diameter
BA 5105	The Microphone Handbook

CALIBRATION SERVICES

4189-CAI	Accredited Initial Calibration
4189-CAF	Accredited Calibration
4189-CFF	Factory Standard Calibration

Brüel & Kjær reserves the right to change specifications and accessories without notice

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BP 2210–11 2008-06

PRODUCT DATA

Miniature DeltaTron Accelerometers — Types 4507 and 4508
Miniature DeltaTron TEDS Accelerometers — Types 4507 B and 4508 B
Miniature Charge Accelerometers — Types 4507 C and 4508 C

Miniature DeltaTron® Accelerometers Types 4507 and 4508 consist of a ThetaShear® accelerometer and a DeltaTron preamplifier in a lightweight titanium housing with integrated 10 – 32 UNF connectors. Types 4507 C and 4508 C are similar to the DeltaTron accelerometers but come without the preamplifier.

040248

USES AND FEATURES

USES
- Modal measurements for automotive body and power-train applications
- Multichannel modal analysis measurements
- Structural analysis measurements

FEATURES
- Robust titanium housing with integrated titanium connector
- Easily fitted to different test objects using a selection of mounting clips
- Low-weight ThetaShear design giving high sensitivity/weight ratio and very low sensitivity to environmental factors
- Triaxial mounting facility

DeltaTron Accelerometers
- Connect directly to DeltaTron power supply (ICP® compatible). The DeltaTron principle allows the use of inexpensive cables. Low output impedance so that long cables can be used
- Built-in, low-noise preamplifiers with ASICs give more than 100 dB dynamic range
- Choice of sensitivities from 10 mV/g to 1 V/g
- ID (TEDS) "Smart Transducer Interface" IEEE – P1451.4 (Types 4507 B and 4508 B)

Charge Accelerometers (4507 C and 4508 C)
- Sensitivity 5 pC/g
- Operating temperature up to 250°C (482°F)

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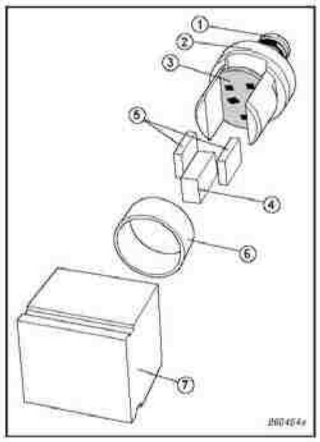
Description

Miniature DeltaTron Accelerometers Types 4507 and 4508 are specifically designed to withstand the rough environment of the automotive industry. A combination of high sensitivity, low mass and small physical dimensions make them ideal for modal measurements, such as automotive body and power-train measurements, as well as for modal analysis on aircraft, trains and satellites. The main difference between the two Types is the position of the coaxial connector which is on the top surface perpendicular to the main axis for Type 4508 (top-mounted connector), and on the side surface parallel to the main axis for Type 4507 (side-mounted connector).

Design

Fig. 1
Exploded view of Miniature DeltaTron Accelerometer Type 4508 (top mounted connector) showing the ThetaShear design and the built-in DeltaTron preamplifier



The 10 – 32 UNF connector (1) is an integrated part of the top piece (2), which also contains the preamplifier (3) (not 4507 C or 4508 C). The slotted cylindrical stanchion holds a central seismic mass (4) flanked by two piezoelectric plates (5). This assembly is clamped rigidly by a ring (6). The parts are firmly held together without the use of any bonding agent other than friction, a principle which has proved extremely reliable in Brüel & Kjær DeltaShear® accelerometers. This assembly is hermetically welded to the titanium housing (7).

Mounting

Special effort has been put into making mounting as flexible as possible. The accelerometer housing has slots that allow the use of mounting clips so that the accelerometers can be easily fitted to a number of different test objects, or removed, for example, for calibration. UA 1407, UA 1475 and UA 1478 are sets of one hundred plastic mounting clips. UA 1564 is a set of five high-temperature mounting clips.

Fig. 2
High-temperature Mounting Clip UA 1564

Specifications:

Temperature range:		–55° to +175°C (–67° to +347°F)
If discolouring can be accepted:		–55° to +250°C (–67° to +482°F)
Weight:		5.7 gram
Maximum acceleration (with a 5 gram accelerometer):		50 g peak
(Perpendicular to mounting surface):		250 g peak
Material:	Base – Anodized aluminium	Spring – Stainless spring steel

Compliance with Standards

	CE-mark indicates compliance with: EMC Directive and Low Voltage Directive. C-Tick mark indicates compliance with the EMC requirements of Australia and New Zealand
Safety	EN 61010–1 and IEC 61010–1: Safety requirements for electrical equipment for measurement, control and laboratory use. UL 3111–1: Standard for Safety – Electrical measuring and test equipment
EMC Emission	EN/IEC 61000–6–3: Generic emission standard for residential, commercial and light industrial environments. EN/IEC 61000–6–4: Generic emission standard for industrial environments. CISPR 22: Radio disturbance characteristics of information technology equipment. Class B Limits. FCC Rules, Part 15: Complies with the limits for a Class B digital device.
EMC Immunity	EN 50082–1: Generic immunity standard. Part 1: Residential, commercial and light industry. EN 50082–2: Generic immunity standard. Part 2: Industrial environment. **Note 1:** The above is guaranteed using Cable AO 1382 only. **Note 2:** Sensitivity to RF (in accordance with EN 50082–2) 4507, 4507 B, 4507 B003, 4507 B004, 4508, 4508 B and 4508 B003: <60 µV 4507 001, 4507 B001, 4508 001 and 4508 B001: <10 µV 4507 002, 4507 B002, 4507 B005, 4507 B006, 4508 002, 4508 B002 and 4508 B004: <100 µV
Temperature	IEC 68–2–1 & IEC 68–2–2: Environmental Testing. Cold and Dry Heat. Operating Temperature: 4507, 4507 001, 4507 B, 4507 B001, 4507 B003, 4507 B004, 4508, 4508 001, 4508 B, 4508 B001 and 4508 B003: –54° to +121°C (–65° to +250°F) 4507 002, 4507 B002, 4507 B005, 4507 B006, 4508 002, 4508 B002 and 4508 B004: –54° to +100°C (–65° to +212°F) 4507 C and 4508 C: –74° to +250°C (–101° to +482°F)

Specifications – Miniature DeltaTron Accelerometers Types 4507

	Sensitivity	Sensitivity Tolerance	Measuring Range	Frequency Range, 10%	Phase Response, ± 5°	Built-in ID (TEDS)	Output Impedance	Bias Voltage	Start-up Time (± 10% of final bias)	Inherent Noise (broadband) Equivalent Vibration Level		Temperature Coefficient of Sensitivity	Sensing Element	Sealing	Humidity	Mounting Slots (pairs)
Units	mV/ms⁻²	%	ms⁻²	Hz	Hz		Ω	V	s	µV	µg	%/°C			%	
4507	10	±5	700	0.3–6k	2–5k	No	<2	12±1	5	<35	<350	0.09	PZ23	Welded	90	1
4507-001	1	±5	7000	0.1–6k	0.5–5k	No	<2	12±1	50	<8	<800	0.09	PZ23	Welded	90	1
4507-002	100	±10	70	0.4–6k	2–5k	No	<2	12±2	5	<150	<150	0.18	PZ27	Hermetic	100	1
4507 B	10	±5	700	0.3–6k	2–5k	Yes	<30	13±1	5	<35	<350	0.09	PZ23	Welded	90	1
4507 B001	1	±5	7000	0.1–6k	0.5–5k	Yes	<30	13±1	50	<8	<800	0.09	PZ23	Welded	90	1
4507 B002	100	±10	70	0.4–6k	2–5k	Yes	<30	13±2	5	<150	<150	0.18	PZ27	Hermetic	100	1
4507 B003	10	±5	700	0.3–6k	2–5k	Yes	<30	13±1	5	<35	<350	0.09	PZ23	Welded	90	None
4507 B004	10	±5	700	0.3–6k	2–5k	Yes	<30	13±1	5	<35	<350	0.09	PZ23	Welded	90	3
4507 B005	100	±10	70	0.4–6k	2–5k	Yes	<30	13±2	5	<150	<150	0.18	PZ27	Hermetic	100	3
4507 B006	50	±5	140	0.2–6k	1–5k	Yes	<30	13±2	10	<30	<160	0.18	PZ27	Hermetic	100	3

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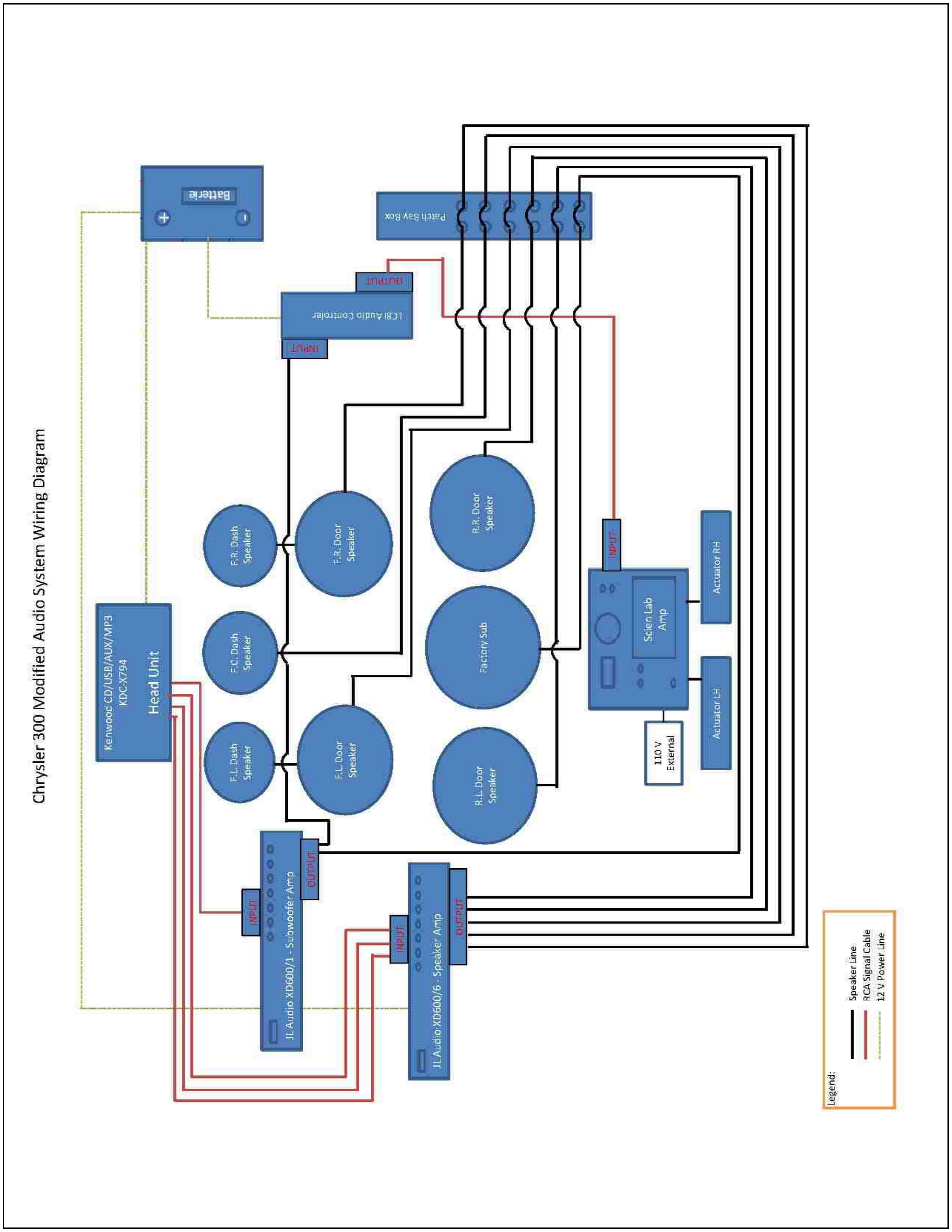
APPENDIX C – Test Vehicle's Modified Audio System Wiring Diagram



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