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SIMULATION, VISUALIZATION AND PERCEPTION OF SOUND IN A REAL AND VIRTUAL VEHICLE INTERIOR USING BEAMFORMING

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ABSTRACT

This paper describes a method to visualize and localize the sound that is measured in real and virtual vehicle interior acoustic environments. Real world measured results using a sound level meter, with dual-channel input and beamforming are used as an input toward simulation of the same conditions within a virtual environment. The use of an acoustic camera along with noise image software as a short introduction to beamforming method for such application is demonstrated. Furthermore, the transition from the three-dimensional sound recording inside a vehicle to the three-dimensional virtual acoustic mapping, visualization and sound perception for its directionality by real subjects within the virtual environment is described along with application of the head related transfer function. The findings will contribute toward sound source localization and reducing the noise annoyance to the drivers and passengers in cars.

Key words: Interior Vehicle Sound Mapping, Visualization, Sound Localization, Auditory Navigation, Virtual Acoustics, Spatial Hearing, Dynamic Auralization, HRTF

1 INTRODUCTION

Mapping of sound fields is a major area of interest in the car industry. Localizing the noise sources is a key path toward identifying source location, direction and reducing the noise source as part of increasing the comfort of drivers of all ages. Different methods are used to quantify the sound intensity and spectrum and map the sound field within an interior of a car. Some of the methods used for this purpose are direct measurements as well as nearfield acoustic holography and beamforming techniques using spherical arrays.

The problem of noise produced inside vehicles has been studied and investigated by many researchers **[1, 2].** A major objective has been reducing the noise annoyance to the drivers and passengers inside the car. The oldest subjective index, the “Salford Criterion”, is shown in **Table 1** and is used to meet such expectations or objectives. Most sounds come from the engine, transmission system and accessories, road excitation, aerodynamic (not a dominant source type), and most drivers perceive these sources as background noise. However, outside vehicle noise reaching the driver’s ears, such as a blowing horn, sirens from police cars, emergency vehicles or fire engines, train crossings, buses, pedestrian crossing sound alarms during daytime (e.g., Japan, Fukuyama and Tokyo), do impact the driver’s sound localization and their perception of sources’ locations, direction and intensity**[5].**

Table 1: Sound pressure level and associated subjective rating of based on Salford Criterion

Subjective rating of Noise inside vehicles	Noise level not exceeding dB(A) Level
Quiet	67
Noticeable	73
Intrusive	79
Annoying	85
Very Annoying	91

Measurements were made inside and outside of a vehicle within typical city rush hours using a Brüel & Kjær sound level meter, type 2770 with dual-channel input (microphone, sound intensity probe, accelerometer or direct signal) with 4.2 Hz to 22.4 kHz broadband linear frequency range and 16.6 to 140 dB at A-weighted dynamic range with supplied microphone, type 4189. The sound level measured data for simultaneously inside and outside of a car , show peaks of Lzeq (dBA) as high as 120 and the spectral variability and noticeable changes at low and high frequencies while having the largest reduction or difference in transmission loss within middle frequencies.

"Loudness" is subjective judgment on intensity of a sound by humans and depends on the pressure and frequency of the stimulus and if the sound field is diffuse or free filed. The unit is the sone. The "loudness level" is equal to 10*log 2(loudness) + 40. The unit is Phone. The Zwicker method of calculation of stationary loudness is based on measurement as described in ISO 532-1975, Method B. "LAeq" is the A – weighted equivalent continuous noise level. Lzeq is a noise parameter calculated based on linear or without any frequency weighting, broadband or flat. Through use of Brüel & Kjær Sound level meter, type 2770, the above indexes were measured within a car interior for a selected length of time. **See Figure 1.**

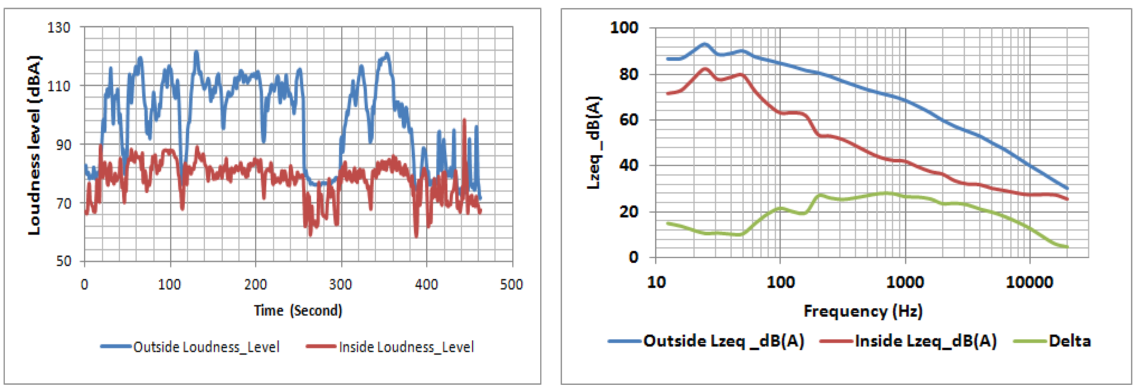


Figure 1: Sound pressure level readings taken inside and outside of Ford Focus 2007 car and the measured results are presented as loudness level (dBA) and L_{zeq} vs. un-weighted frequency.

Noise inside the car depends on the spectral power of noise source from engine and dynamic or elastic properties of the body structure and interior space and the natural frequency of the interior body materials and the resonances in the air inside the interior cabin. Past and recent studies show the occurrence of resonances of 63 to 200 Hz at the driver's seat and the passenger side is close to 100 to 150 Hz. While the outside sound pressure level is within 60 to 120 dB (A) **[3]**. Good correlations have been observed between subjective assessment of Salford Criterion and other indicators such as Zwicker Phones. The results indicate that the subjective response to internal car noise along with their comparison between the ANSIS3.4:2007 and DIN-45631:1991, that it is possible to adequately measure subjective assessment using A weighting network **[4, 5]. See Figure 2.**

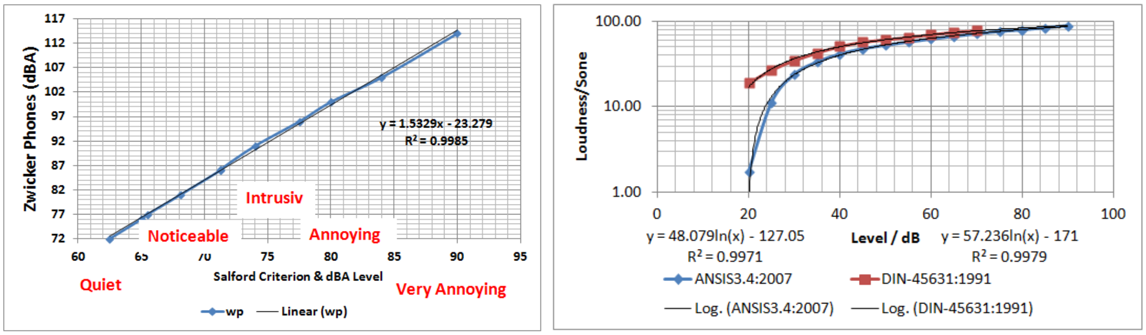


Figure 2: Sound pressure level readings and associated subjective rating of the noise conditions are presented for Salford Criterion as compared with as Zwicker Phones scale. The fitted functions for loudness show good correlation with and between the ANSIS3.4:2007 and DIN-45631:1991 as presented.

Beamforming systems for imaging analysis of complex sound sources have been made possible by Noise Image software in industrial applications for number of years. Real architectural spaces have complex deep-structured surfaces and often times require distributed sound sources within the space. These conditions require high demand in computing time during simulations and or detailed measurements during space auditing for the sound system, but are not free of errors in plane approximation in the form of falsification of levels, distorted localization and aliasing effects in simulated and or measured results as a consequence. The sound auralization based on these results will not be as realistic as the measured data within these spaces. One of the objectives in this study was to conduct an experiment in such a way that the perceived sound changes according to the movements of the subject (e.g. distance & orientation) in a given simulated space in this case the interior of a vehicle for its acoustic characteristics to be evaluated. The results could be used as part of an auditory navigation experiment. Many auditory navigation tests have been conducted in the past, and an overview of these different techniques needed in auditory navigation shown in research work by **[6, 7, and 8]** is directly related examples of such experimental work.

In our experiment we applied the capability of EASE, simulation software, and FMOD auralization systems within a VR Laboratory to obtain close approximation of the simulated interior space architectural characteristics of an actual 3D-virtual reality of a car. The locations of the loud speakers or in this case noise sources (e.g., horn, train, police car etc…) and their directionality are modeled based on the actual measurements using the acoustic camera recording system **[9 to 13].** Given the dynamic changes within today's audio system technologies; the meaning of "3D Sound" to the public at large is characterized more in terms of processing, however this characterization is not monophonic or stereo. Most human sensations involve electrochemical or chemical reaction within the brain and the "3D sound" is best understood or perceived through its localization within a given space for its volume and surface absorption characteristic.

This perception is individual and totally subjective and not objective. Based on many technical papers published related to sound perception and sound localization in humans; the externalization and localization (directional) effect, the environmental reflections, scattering, and reverberation cues are required to produce the best 3D sound effect. In passive "3D sound" processing, the source position as being localized does not change if the listener moves the head since the dynamic or static sound source is independent of the head position and its direction; however; in the interactive "3D sound", dynamic cues such as listener position and movement plus sound movement (if any) within the audio scene must be added. These sound localizations are achieved by the application of the Head Related Transfer Function (HRTF) through the inclusion of the cues exclusive of differences between the signals reaching each ear as Inter-aural Time Delay (ITD) and Inter-Aural Level Differences (ILD). "Relying on a variety of cues, including intensity, timing, and spectrum, our brains recreate a three-dimensional image of the acoustic landscape from the sounds we hear "**[14].**

2 METHODOLOGY

Nearfield Acoustic Holography (NAH) measurement plane has to be very close to the source plane in order to catch the evanescent waves of the sound field and the spatial resolution is decreased with the distance between the planes. Beamforming techniques have the advantage for the measurements to be made further away from the source covering the entire sound field in a single measurement. However, the resolution is very poor at low frequencies even though it is improved with spherical harmonics beamforming instead of delay-and-sum beamforming. Spherical beamforming uses a far-field beamforming technique designed for free-field conditions in an interior space, which is a reflective sound field. Spherical beamforming does not make use of a flat, two-dimensional array but employs a spherical array. Therefore, the spherical beamforming technique helps identify the exact position of a sound source in the surrounding space. We combined this method with the HRTF to visualize the location of the source with respect the driver's head. The advantages and disadvantages of these methods are discussed in some detail within Lanslots' article and others **[15, 19 and 20].** A series of interior car cabin noise measurements are made in a stationary position while being exposed to pink noise source located outside using spherical beamforming array Noise source was placed 5 meter outside while the array is located inside the car's cabin. This setup enables us to localize outside noise transmission inside the car. The car windows were closed all the time. The additional channel called Mic, using the B & K sensor (BK4189W003) as a reference microphone was placed directly in front of the membrane of the speaker. The speaker was 5m away from the car at a height of approximately 80cm (membrane). The position was changed from front to back and left to right sides of the vehicle at the same 5 meter distance for parametric testing.

Figure 3 – Experimental set up for acoustic mapping (SPL) of the pink noise inside a car

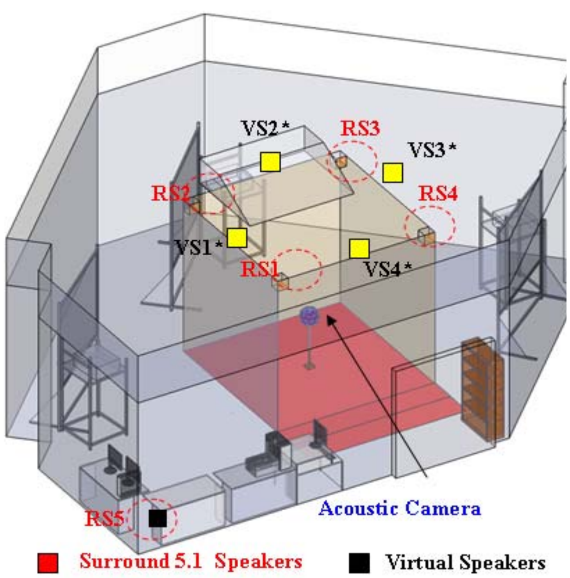
Spherical Beamforming Array's specifications given this particular application are: 48 microphones, 35 cm diameter, Carbon fiber structure, Dynamic of the microphones: 35 dB -130 dB, Recommended mapping frequencies: 1 kHz - 20 kHz, Typical measurement distance: 0.4 - 2 m, Data Recorder, 92 kHz, Sampling frequency, 48 to 144 channels per 10 inch rack (24 channels per card), Ethernet Interface > high transfer rate > 20 MByte/s, network-compatible, Digital card with 12 extra channels for recordings of RPM, rotation angle, reversal point, etc., Integrated PC with Windows XP (embedded), Software, Noise Image, Power Supply, Mobile power supply/battery pack. **See Figure 3**.

2.1 The 3D Virtual Reality Lab

The 3D virtual reality (VR) laboratory includes an immersive virtual-reality like environment, measuring 10 ft (3.048 m) in width, depth, and height. It runs on a cluster of six workstations, with one control computer, one motion-tracking computer, and four rendering computers. The renderers are Box Tech Workstations, with quad-core CPUs at 2.6 GHz, 8 GB RAM, and NVIDIA Quadro FX 5600 + GSync graphics cards. Four Christie Mirage S+4K projectors produce 3D images on the left, front, right, and floor surfaces. The resolution per surface is 1024 x 1024 pixels. The stereo mode is frame-sequential (alternating left-right) at 96 frames per second. Infrared emitters synchronize Stereo Graphics Crystal Eyes® liquid crystal shutter glasses with the projectors. A Vicon MX13 system with eight 1.3 megapixel cameras provides wireless (near infrared) motion-tracking of the shutter glasses and a Logitech Rumble Pad game controller. The sound system comprises four Klipsch speakers mounted in the upper corners, a Klipsch subwoofer on the floor a short distance away, and two amplifiers at 100 watts per channel. The software is an ongoing in-house development, named Jugula,that integrates several open-sources, proprietary, and custom-developed subsystems for graphics, sound, animation, physics, motion-tracking, data management, and networking [16, 17].

2.2 The Virtual Reality Modeling Language (VRML) & Interactive Audio

The VRML is representative of simulation capabilities in the Virtual Environment. VRML version 2, also known as VRML97, was adopted as an International Standard ISO/IEC 14772 in 1997. The standards specify a file format, a content model, and algorithms for its interpretation. The model is a directed acyclic graph that includes nodes for geometry, color, texture, and light, as well as sound. However, it provides for only one texture per shape and one pair of texture coordinates per vertex. VRML version 2 was amended in 2002 to add geospatial and NURBS support, but the shape, material, lighting, and sound specifications remain unchanged. FMOD is a programming library and toolkit for the creation and playback of interactive audio. It supports all leading operating systems and game platforms. FMOD is a proprietary audio library made by Firelight Technologies that plays music files of diverse formats on many different operating system platforms, used in games and software applications to provide audio functionality within the 3D virtual Laboratory. The FMOD sound system has an advanced plug-in architecture, that can be used to extend the support of audio formats or to develop new output types, e.g. for streaming. FMOD sound system now contains three main parts: 1) FMOD Ex, the low-level sound engine, 2) FMOD Event System, more abstract, higher level application layer to simplify playback of content created with FMOD Designer, 3) FMOD Designer, the sound designer tool used for authoring complex sound events and music for playback [17-18]. Figure 4 shows sound mapping representation for a selected speaker and schematic, computer model of the CAVE space, speakers' locations and real inside view of the VR Laboratory.



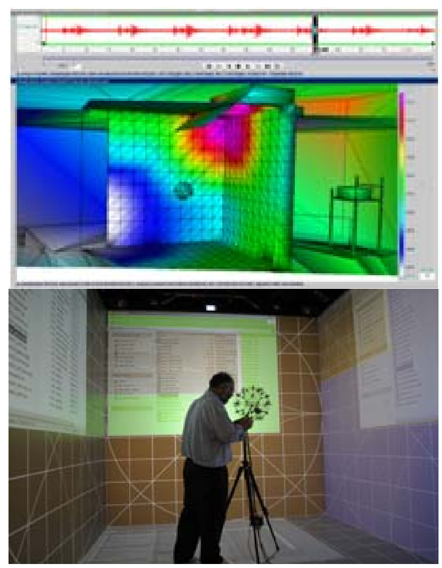


Figure 4: Schematic computer model and real views of the virtual lab's speakers and their locations.

2.3 Principle of the Delay-and-Sum Beamformer

The time domain calculation of a delay-and sum beamformer within the Noise image software is achieved by the use of the equations 1 and 2 for the reconstruction of the time function at every location seen by the acoustic camera microphones' field of view.

$$\hat{f}(\mathrm{x},t) = \frac{1}{M}\sum_{i=1}^{M} w_i\, f_i\left(\mathrm{x},(t-\Delta_i)\right) \qquad \textbf{(1)}$$

Effective value of p at location **x**:

$$\hat{p}_{eff}(\mathrm{x}) \approx \hat{p}_{eff}(\mathrm{x},n) = \sqrt{\frac{1}{n}\sum_{k=0}^{n-1} \hat{f}^{\,2}(\mathrm{x},t_k)} \qquad \textbf{(2)}$$

Where x is the location of a point and t denotes the time and M is the number of the microphones in the sensor array. The **fi (t)** are the recorded time functions of the individual microphones, and the **Δi** are the appropriate relative time delays, which are calculated from the absolute run times **Δi=** by subtracting the minimum over all **Ti** . The symbol c denotes the speed of sound in air and **|ri|=|Xi - X|** is the geometrical distance between the spatial position of microphone number **i** and the actually calculated focus point **x**. Despite its extreme simplicity, the delay-and-sum method in the time domain is quite robust and powerful and has shown its practical usability in an extraordinary wide range of acoustic localization and troubleshooting applications for several years **[19-23].**

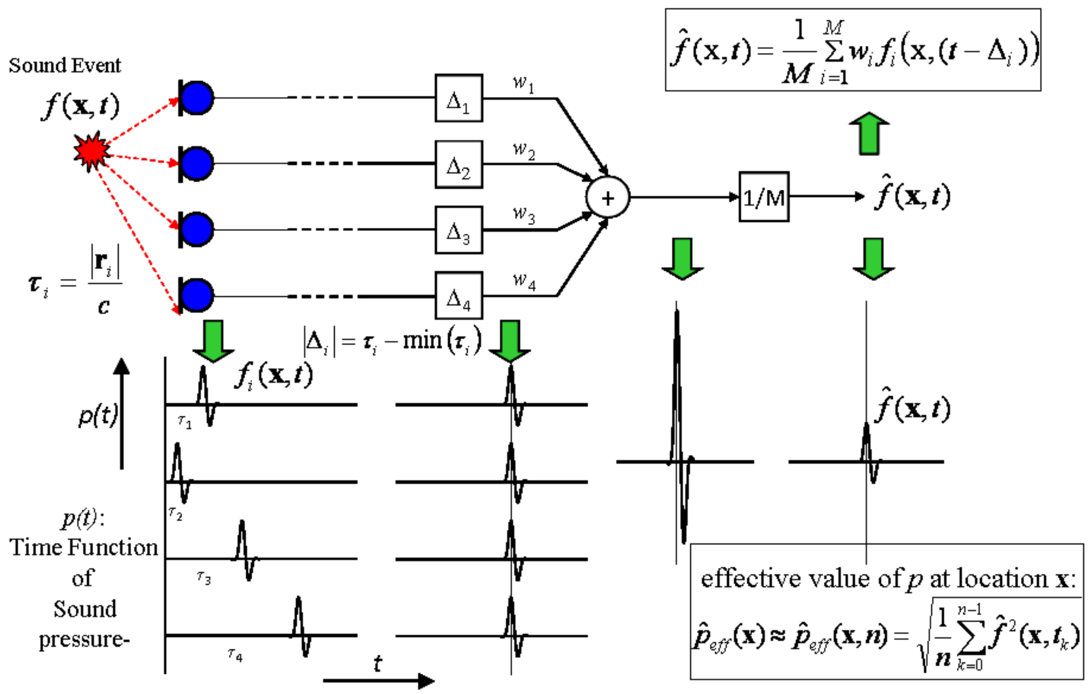


Figure 5: Basic principle of the delay-and-sum beamformer in the time domain.

The output devices such as speakers at a distance from the listener or headphones impact the user experience with "3D Sound". Passive "3D Sound" where the results always sound the same, the sound spatialization information can be used within a sound playback file, and heard through a stereo amplifier with speakers or headphones. Interactive "3D Sound" situations where the listener's apparent position within the audio scene is required, the "3D Sound" rendering algorithms should be done on listener equipment in real time. The Computer Aided Virtual Environment **"CAVE"**'s sound system utilizes the FMOD. It is known that the system in its speaker mode will never be as good as the headphone mode for "3D sound" rendering since the use of HRTF rendering algorithm must interpolate for locations in between the given HRTF locations, it requires high real time computing **[9, 17].** See **Figure 5.**

3 HUMAN SUBJECT EXPERIMENTAL SET UP

The subjects were free to move and turn in a virtual space to analyze the effect of various noise factors (e.g., computers, projectors and HVAC system within the space) which impact the sound stimulus and hence their directional cues. This takes place within a simulated acoustics environment while the recorded sound of outside vehicle noise such as horn, siren or music is played through 4 real speakers. The FMOD system provided or simulated the additional 4 virtual speakers between the real speakers. The time spent from starting to end position, and trajectory of the subject's motion, were recorded digitally using a digital camera equipped with an equal distance projection fisheye lens and a video recorder. Specific observations were made on each subject's ability to locate or localize the real or virtual source by looking and aiming the "crosshair" indicator closest to each speaker's location. A Head-Tracking Calibration Feature (HTCF) that projects the transverse, sagittal, and coronal planes of the user's head (centered between the eyes) was used during the experiment. The intersections of the transverse and sagittal planes with the display planes produce a "crosshair" that indicates the user's center of vision. This was used to determine the best estimate of the locations of the real or virtual speakers as identified by the subjects (e.g., **-** Left and + Right of the speakers in **centimeter**). Additionally, HTCF projects "diamond" quadrilaterals from the head center perpendicularly onto each display plane. When the user looks directly at a display, without head rotation, the crosshairs should center in the diamonds. The head-tracking system is calibrated accordingly. See **Figure 6** for the interior view of the VR lab while the simulated scene is displayed and **Figure 9** for subject field of view and direction of gaze for locating the real or virtual speakers.

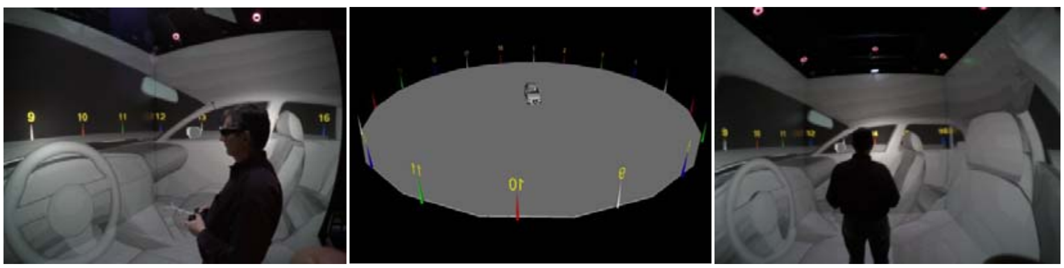


Figure 6 – Experimental set up for virtual acoustic localization of noise inside a car.

In the experiment, instructions were given aurally. In the beginning of the experiment there were intentionally no rehearsal tests to help the objectives of the experiment, so that the subjects would not notice or recognize the locations of the real loudspeakers within the virtual simulated environment. A complete test set included 8 speakers (4 real and 4 virtual) and each source played an identical WAVE file within the virtual space. These 8 positions for all speakers are shown in **Figure 4** as virtual speakers (VS1, VS2, VS3 and VS4) and the real loudspeakers as (RS1, RS2, RS3 and RS4) in the real space.

3.1 Directing the Subjects

Moving in the virtual space was controlled with the use of Jugular software and a motion tracking system. The subject was able to move his head forward, backward, and to turn left and right at normal human speed within the interior of the vehicle. When subject assumed that he/she has identified the sound source direction for a given sequence (e.g. clockwise at 45 degree intervals), he/she indicated that by stopping at the final location. This experiment was done in the horizontal and vertical plane given the 45 degree downward tilt of angle by the speakers aimed toward the center of the interior of the car at 5 feet height from ground. The sound source was a point source. Starting positions were in order from speaker 1 to 8 and the approximate or exact locations were identified by all subjects. The entire experiment did not exceed 5 minutes per subject. WAVE files were displayed or simulated acoustically within the virtual environment using the FMOD sound system **[17]**.

4 RESULTS

Objective Measures: A series of interior car cabin noise measurements were made in a stationary position using spherical beamforming array while being exposed to pink noise source generated outside of the vehicle. Acoustic map (SPL) of the pink noise as a sound source from four directions reaching inside the car with doors and windows closed were measured and are presented in **Figure 7.** The measured results were mapped over an sphere representing drivers head position and utilizing the Head Related Transfer Function option to view the source localization over the head as well as the car surface as viewed from outside for a given directionally. **Figure 8** shows acoustic map (SPL) of the pink noise as a sound source reaching inside the car through HRTF filter given the space allocated for the driver's head position within the domain of measured frequencies.

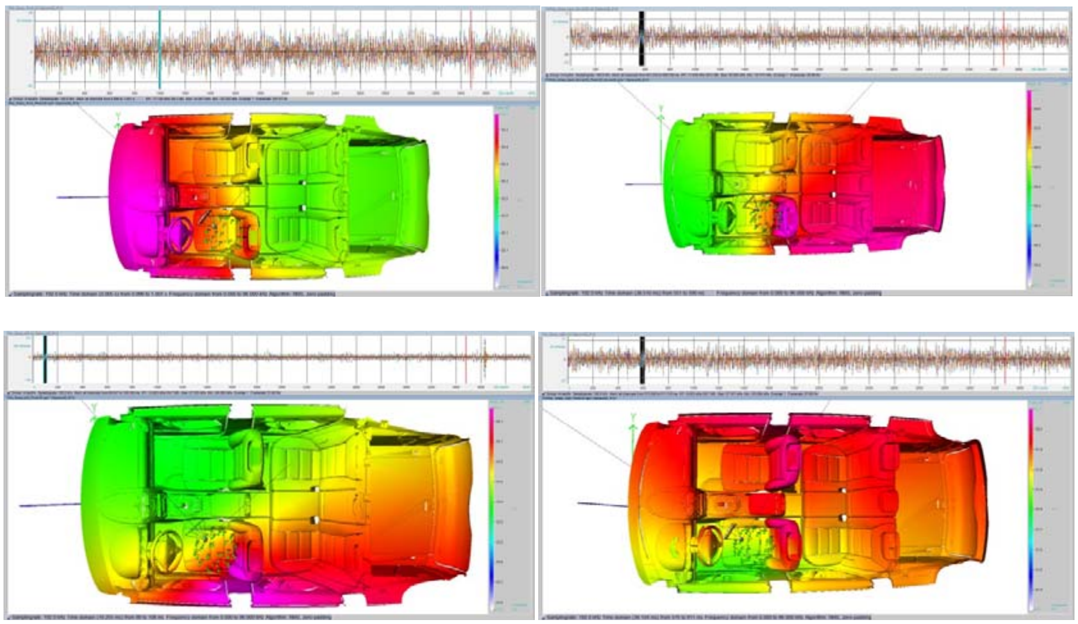


Figure 7: Acoustic map (SPL) of the pink noise as a sound source reaching inside the car with doors and windows closed and from four directions (clockwise; front, back, right and left) within the domain of frequency measured by the spherical array.

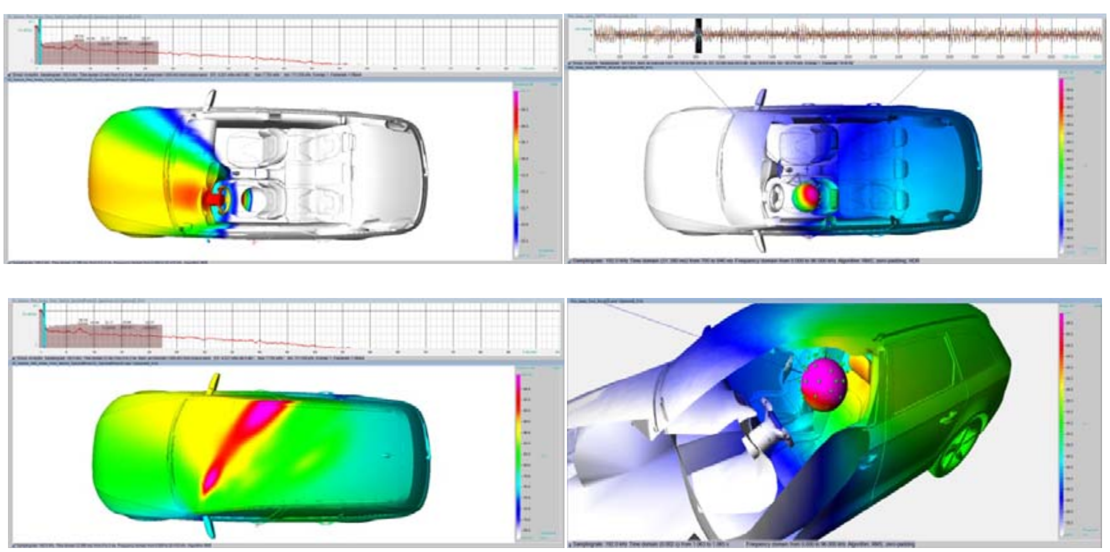


Figure 8: Acoustic map (SPL) of the pink noise as a sound source reaching inside the car through HRTF given the space allocated for the drivers head within the domain of measured frequencies.

Subjective Measures: Stimulus, panning method for localizing the sound within the acoustic environment was used to conduct this experiment. Each WAVE file had equal loudness. Each speaker cycle of time play was about 8 seconds long including one second dead band as played in a sequential loop for all 8 speaker positions. The sound source had an Omni-directional pattern. The synthesized or aurulized sections of the sound were produced by a physical-based model **[22, 23].** Panning or seeking for the sound source methods had no limitation on subject movements. The inter-aural time difference (ITD), was included as an auditory cue to all test conditions. The ITD was calculated from a spherical head model within EASE program **[13]** and implemented with a short delay line. When the subject asked to locate the sound, they started at the center of the space facing the center wall with full degree of freedom to move within the virtual space. Once the source of noise such as horn, siren music started to play, subjects were asked to seek and locate the sound and its direction, Subject movement and recognition of the source location were recorded using a camera and a video recorder. The **Figures 6** and **9** show the experimental set up for virtual acoustic localization of noise inside a simulated car. The color ID numbers at a distance shown in the image within figure 6 were used as visual cue to help identifying the source direction by the subjects at 20 degree azimuthal intervals. The subjective evaluation results within the simulated environment are summarized in **Table 2.**

Table 2: The subjective evaluation results within the simulated environment.

Subject Performance	**1SPK-RS1**	**2SPK-VS1**	**3SPK-RS2**	**4SPK-VS2**	**5SPK-RS3**	**6SPK-VS3**	**7SPK-RS4**	**8SPK-VS4**
Avgerage time to locate the source (sec)	2	4	2	3	3	4	5	5
Location within centimeter of speaker + or -	**-5**	10	**-5**	**-10**	20	**-25**	10	25
# of Subjects from 61 (total) located the Spk	43	55	59	55	58	51	44	41
% Number of subjects located SPK	0.70	**0.90**	0.97	**0.90**	0.95	**0.83**	0.71	**0.66**
Speakers Position								
Speaker within two wall	NO	YES	YES	YES	YES	YES	NO	NO
SPK Field of projection (Horizontal/Vertical)	90/90	90/90	90/90	90/90	90/90	90/90	90/90	90/90
Aiming direction downward center	45/45	45/45	45/45	45/45	45/45	45/45	45/45	45/45

The best results in virtual sound source localization are achieved if and only if the binaural difference-based cues and the monaural and binaural cues that arise from the scattering process of the Head Related Transfer Function (HRTF) for the particular individual are included within the virtual scene. If the primary output device is a pair of headphones or speakers, the "3D Sound" rendering process will result in a stereo signal, (e.g. one signal for the "Left ear" and one signal for the "Right ear"), but the signal components tend to cancel cross talk effects between Left and Right channels. The effectiveness to which this can be done varies both with distance from the listener, as well as distance (or angle) between the speakers. The effect is best with speakers with the angle range of 27-45 degrees between the listener and speaker on either side of the listener when facing forward. Measured results for different viewing conditions of the surfaces within the virtual laboratory provide some clue to the incoming sound and its directions. **Figure 4** shows the mapping of the sound filed for real speaker #2 within the CAVE. The sequence of the speakers and the sound intensity distribution in terms of calculated Delta dB within each viewing scene for each real and virtual speaker provide an insight into the sound projected from each speaker. These results show the impact of the VR screen on sound distribution within the space and as to how it is perceived by the human subjects with respect to each speaker (real and or virtual) as indicated. The measured data were viewed and examined using the Noise Image software with and without the HRTF as a filter for the sound source directionality as it was perceived by the subject at the center of the space. The measurement and simulation path within real and virtual environment for all experimental procedures are shown in a flow chart diagram including the steps for HRTF filtering Noise Image software within **Figure 9.**

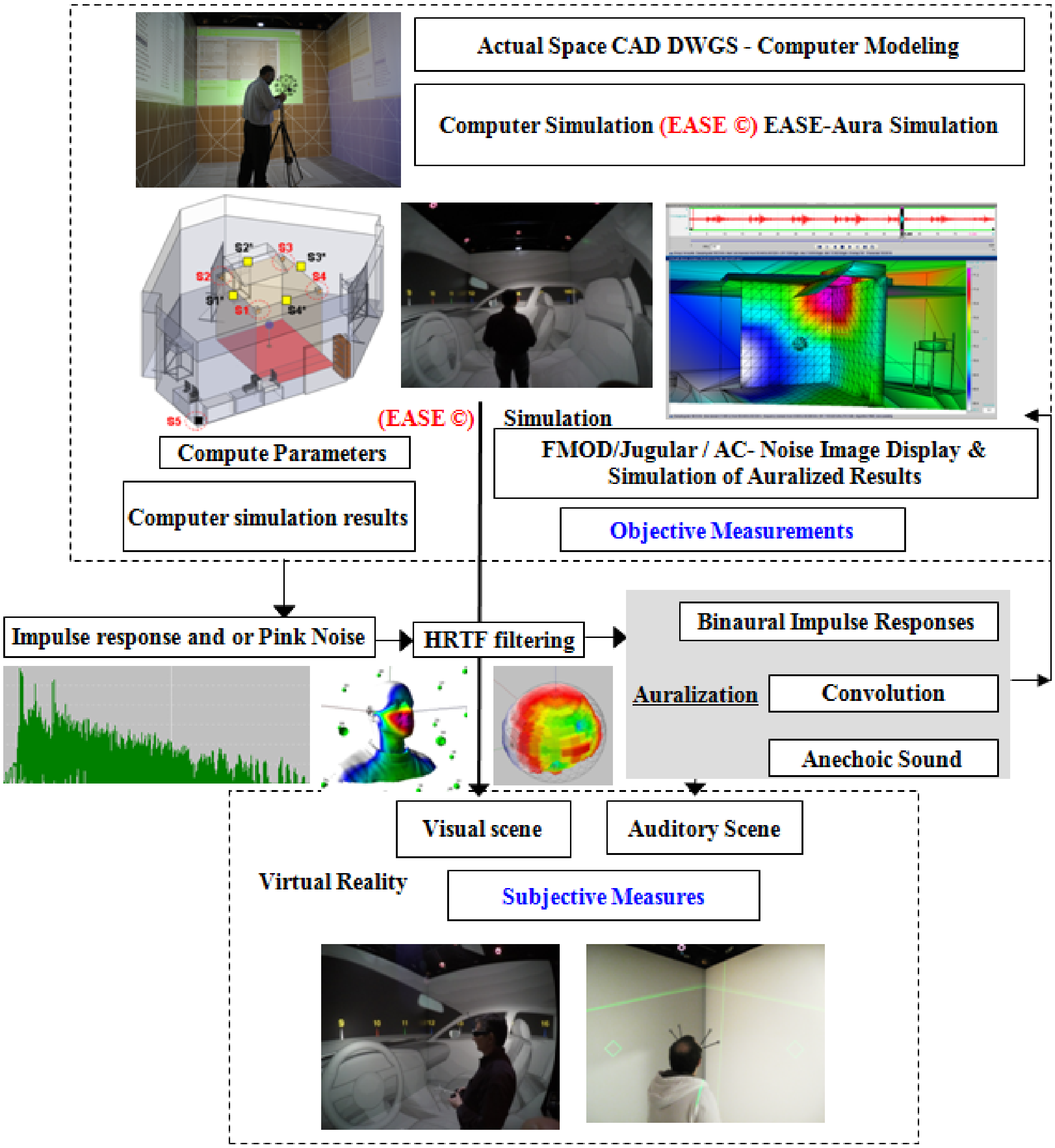


Figure - 9: Measurements and simulation paths within real and virtual environments.

5 ANALYSIS & DISCUSSION

Determination of position and direction of arrays inside of a car interior: The HRTF rotation option available within the AC/NI software was used to determine the coordinates and the orientation of the array in relation to the 3D-model of the car interior. The precision of this fitting determines the quality of the results. To give an example, a rotation of the array of only 2 degrees will result in a location error of 0.07m at a 2m distance. A determination of position and direction of the array was made possible through the use of an advanced technique derived from photogrammetry. All arrays include a built-in video camera. If the array is placed inside the car, the video camera sends a video stream of the car interior. The software module for preparing three dimensional mapping receives the video stream, and overlays the pictures with a 3D–model of the car interior. By moving and rotating the 3D model, the pictures from the video or the still capture images made by the camera are matched with the 3D model. If an exact match is found, the position of the microphone array in relation to the 3D-model will be determined. The final fitted scene is saved as a VRL formatted file to be use in the CAVE for subjective assessment within the virtual environment as shown in **Figure 6**. The results indicate that the inter-reflection and or the missing 4th wall (back wall between speakers # 4 and #1) within the virtual environment have impacted the subjects' localization of the sound source due to the loss of reflections. The exact location of each source and the percent of the subjects identifying the correct locations of the sources, within positive or negative distances, show the impact of such a condition. See 3rd and 5th rows within **Table 2.** These result in positive or negative distances (cm) from the center and or the exact location of the speakers, validate the hypothesis that the wall surfaces (e.g., interior car reflective surface such as windows and dashboard) do contribute to the inter-aural time difference (ITD). However, parametric tests for examining the interactions between large numbers variables require a larger number of subjects. This experiment includes only 61 subjects. It is important to acknowledge that error rate. The large variation in times between the test subjects for locating the real and virtual sources needs to be fully examined. It is worth noting the percentage of the subjects and the times it took them to identify the real and virtual speakers, and the correlation between the location and the missing back wall (5th projection screen). One explanation might be the way the test subjects started and the sequence of the noise sources as each subject tried to locate each target as close as possible, without caring how much time they spent. The application of the HRTF would provide the future opportunity to examine this condition more carefully for room acoustic design and architectural application. The results of this work show it is possible to navigate within a virtual environment while using even limited auditory cues. The subjects completed the navigation tasks, and had no difficulty in localizing the sound given all available vehicle interior material acoustic characteristics for accurate stimulation within the virtual environment. The findings of this study were similar to other past studies using simple models of spatial hearing given enough cues for auditory navigation and equal perception of the sound for close to real space. This approach, as shown in **Figure 9,** allows one to experience the reverberation, ITD within the simulated space as well as validating the HRTF using various sound auralizations software.

6 CONCLUSION AND FUTURE WORK

Prerequisite for this proposed application of mapping of sound fields in car interiors is a 3D-digital model which can be created quickly within this Computer Aided Virtual Environment. The results show that the subjects were able to navigate and locate a real and virtual sound source outside of the vehicle in a dynamic virtual acoustic environment. The findings from these simulations, both the auditory navigation experiments via noise mapping and a visualization technique within the virtual environment, demonstrate the beamforming method combined with human subject data obtained in the virtual environment advantages. They also provide opportunities to study sound localization and fine tune the current HRTF for various interior vehicle acoustic design studies. Future applications for autonomous cars include identifying source location, direction and reducing the noise source automatically as part of increasing the comfort of drivers and passengers of all ages. Given the good correlations observed between subjective assessment of Salford Criterion and other indicators such as Zwicker Phones, the results indicate that the subjective response to internal car noise, adequately measures subjective assessment using A weighting network using ANSIS3.4:2007 and DIN-45631:1991 standards.

In the future we will make more listening tests for the analysis of interactions between two or more statistical variables for auditory navigation in virtual environments as well as testing within an actual vehicle interior utilizing HRTF filtering for frequency and intensity toward judging source distances.

REFERENCES

[1] M.E. Bryan, “A tentative criterion for acceptable noise in passenger Vehicles”, Journal of sound and vibration, 478(4), Oct, 1976, p525
[2] S.M. J. Ali and S.P. Sarna, “Noise level inside the passenger cars, journal of applied acoustic, (11), 1978, Applied science publisher Ltd, UK.
[3] R. D. Ford, G. M. Hughes and D. J. Saunders, “The measurements of noise inside Cars”, journal of applied acoustic, (3), 1970, Elsevier publisher Ltd, UK
[4] Joachim Scheuren, “ISO532- living and working with alternative loudness standards”, Inter-noise, 2014, Melbourne Australia, proceeding, 16-19 November, 2014,
[5] http://www.soundsnap.com/tags/pedestrian_crossing)
[6] Loomis, J., Golledge, R., and Klatzky, R. Navigation system for the blind: Auditory display modes and guidance. Presence: Teleoperators and Virtual Environs 7, 2 (Apr 1998), 193–203.
[7] Rutherford, P. Virtual acoustic technology: Its role in the development of an auditory navigation beacon for building evacuation. In Proc. 4th UK Virtual Reality SIG Conference (London, UK, 1997), R. Bowden, Ed., Brunel University.
[8] Begault, D. 3-D Sound for Virtual Reality. Academic Press, Cambridge, MA, 1994.
[9] Hall Ted W., Navvab, M., 2011, Ch. 11, PART III: "Virtual Reality as a Surrogate Sensory Environment. Book by Tauseef Gulrez“ Advances in Robotics and Virtual Reality", Springer's book/978-3-642-23362-3, http://www.springer.com/engineering/computational + intelligence+and+complexity/book/978-3-642-23362-3
[10] G. Heilmann, A. Meyer, D. Döbler: Time-domain beamforming using 3Dmicrophone arrays, Proceedings of the BeBeC 2008, Berlin, Germany, 2008
[11] Don H. Johnson, Dan E. Dudgeon, “Array Signal Processing,”1993 by PTR Prentice- Hall, Inc.
[12] Dirk Döbler and Gunnar Heilmann, “Perspectives of the Acoustic Camera,” The 2005 Congress and Exposition on Noise Control, August 2005.
[13] Enhanced Acoustic Simulator for Engineers - E.A.S.E. Software- Acoustic modeling , http://afmg.eu/index.php/company.html.
[14] Hartmann, W. M. (1999). "How We Localize Sound," in Physics Today, pp. 24-29.
[15] J. Lanslots, F. Deblauwe, K Janssens, “Selecting sound source localization techniques for industrial application”, sound & vibration, June 2010
[16] Beier KP (2000) Web-Based Virtual Reality in Design and Manufacturing Apps. In: Hansa International Maritime Jour 137/5:42–47
[17] http://www.fmod.org/index.php/fmod
[18] ISO/IEC (1997), the Virtual Reality Modeling Language (ISO/IEC 14772¬1:1997). ISO, Geneva http://www.web3d.org/x3d/specifications/vrml/ISO-IEC-14772-VRML97/
[19] O. Jaeckel: Strengths and Weaknesses of calculating Beamforming in the time domain, Proceedings of the BeBeC 2006, Berlin, Germany, 2006
[20] Any. Meyer, D. Döbler, "Noise source localization within a car interior using 3D microphone arrays, Proceedings of the BeBeC 2006, Berlin, Germany, 2006
[21] Andy Meyer, Dirk Döbler, Jan Hambrecht, Manuel Matern , "Acoustic Mapping on three-dimensional models", International Conference on Computer System & Technologies, 2011.
[22] Valimaki, V. Discrete-Time Modeling of Acoustic Tubes Using Fractional Delay Filters. Doctoral thesis, Helsinki University of Technology, Lab. of Acoustics and Audio Signal Processing, Report 37, 1995. Available at http://www.acoustics.hut../-vpv/publications/vesa phd.html.
[23] Tapio Lokki, Matti Gr¨ohn, Lauri Savioja, and Tapio Takala, A Case Study of Auditory Navigation in Virtual Acoustic Environments, IEEE, 2000, ISSN: 1070-986X http://ieeexplore.ieee.org/servlet/opac?punumber=93