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Development of a Low Cost System for Pass-by Noise Beamforming Measurements

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ABSTRACT

Beamforming is an acoustic imaging technique that can estimate the radiation pattern of simple or complex sound sources and produce a map of the results. The pass-by noise test is a standardized test that aims to evaluate the overall noise of vehicle's sideline. By coupling the idea of pass-by test with the extension of beamforming technique to moving sources provides the access to the recognition of sound sources produced by vehicle movements, for example, rolling tires, engines and exhaust systems. The present paper aims to describe a low cost system to apply the beamforming technique to pass-by noise test. The system is based on the use of low cost electret microphones mounted in a metallic array which are connected by a coaxial cable to the acquisition system. Later in this document in the application section the results of beamforming maps of pass-by noise test can be viewed in more detail.

INTRODUCTION

The understanding of how an individual behaves toward a sound involves a considerable number of complex factors that are between physics and philosophy. That is, in a practical example may be cited the case of noise due to the closing of the door of a vehicle. The characteristics of this noise will influence the consumer's decision about the quality of the product.

Such facts and governmental laws lead the companies to invest in the acoustics of their products to improve it. And in this quest a more accurate characterization of the sound source is needed. Considering this context the visualization techniques of the acoustic field have a remarkable role.

This work was realized with the purpose to develop a complete "low cost" beamforming system. And then apply it to a known acoustical procedure, in this case, the pass-by noise test.

FUNDAMENTALS

Beamforming

Classical Time Domain

Using the time domain beamforming there is a possibility to listen and auralize the data recorded, since the data in time is not lost.

Time domain processing has no bandwidth limitation, so it can be computationally efficient for wide band processing, especially for a small beamforming grid. Its major disadvantage is the fact that all of the microphone data must be processed for every grid point, so it is time consuming for large grids and long datasets.

Considering a point monopole at the position $\vec{x}'$ as shown in the Figure 1, the pressure field is determined by the solution of the spherically wave equation [1] given by:

$$f(r,t) = \frac{s(t - |\vec{x} - \vec{x}'|/c)}{|\vec{x} - \vec{x}'|}, \quad (1)$$

where $r = |\vec{x} - \vec{x}'|$ is the distance from the source to an arbitrary field point and c is the wave propagation speed.

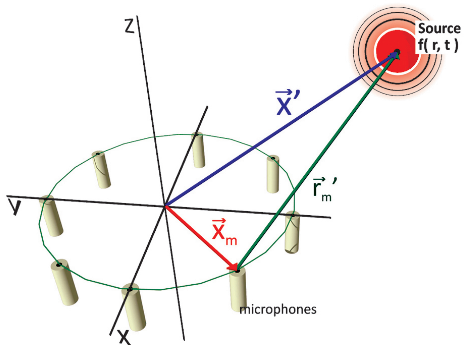


Figure 1: Simple five microphone array, point monopole source.

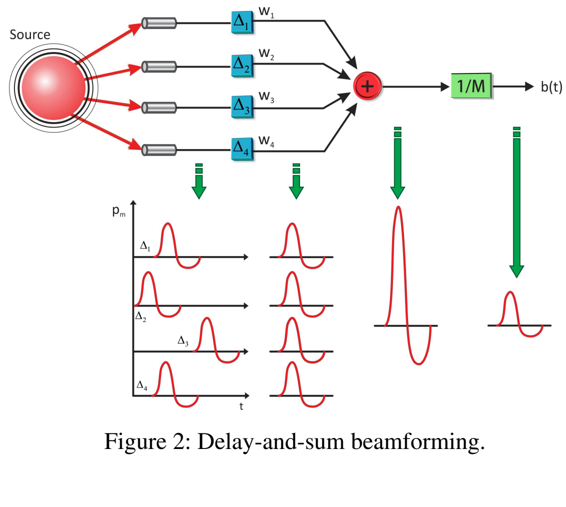


Figure 2: Delay-and-sum beamforming.

Thus the signal received by the m^{th} microphone is $p_m(t) = f(r'_m, t)$, where r'_m is the distance between the source and the m^{th} microphone.

Considered an array of M microphones whose locations are given by $\{x_m\}$, $m = 1, 2, ..., M$, with its center as the origin of coordinates. The classical delay-and-sum beamforming [2, 3] is given by:

$$b(t) = \frac{1}{M} \sum_{m=1}^{M} w_m \, p_m(t - \Delta_m) \,, \qquad (2)$$

where w_m is a weighting, Δ_m is the time delay applied to the mth microphone and M is the total of microphones. Setting the time delays equal to the acoustic propagation times produces signal reinforcement as shown in Figure 2.

Frequency Domain

The frequency domain method [2] can offer some benefits if the data in time is not required. The linear algebraic relationship between the sources and the array data permits the application of specialized algorithms to suppress sidelobes and remove spurious noise. The formulation is inherently narrowband, but results from processing multiple frequencies can be combined [4].

Frequency domain beamforming is based on the property that a delay in time domain corresponds to a phase shift in the frequency domain. Therefore the Equation (2) can be written as:

$$B(\omega) = \frac{1}{M} \sum_{m=1}^{M} w_m P_m(\omega) e^{-j\omega\Delta_m} \,, \qquad (3)$$

where $P_m(\omega)$ and $B(\omega)$ are the Fourier transformed signals.

The ensemble average of the magnitude-squared of Equation (3) can be written:

$$|B|^2 = \frac{g_k^\dagger C g_k}{M^2} \,, \qquad (4)$$

where g_k represents the steering vector which contains the weights and the time delays for each microphone, C represents the *cross spectral matrix* (CSM), and the dagger denotes the Hermitian conjugate.

The pressure response of the array is the square root of the array power response. Figure 3 illustrates the method.

Other advanced techniques can be found in [5, 6, 7].

Doppler Effect

One of the known effects in acoustics is the Doppler Effect. The effect is perceived when cars or ambulances transit in a high speed emitting a pass-by noise. Moving the sound source or even the receiver causes changes in the reception rate of the sound pressure levels and changes of frequency perceived by the receiver. If it has relative approach, the received frequency rises progressively, because it is received a great number of wavefronts per second.

The difference between a perceived frequency and a real frequency of the sound was studied by the Austrian physicist Christian Jonhann Doppler (1803-1853), that discovered the phenomenon in 1842, known as Doppler Effect. Later in 1848, the French Armand H. L. Fizeau (1819-1896) had complemented his theory with some corrections.

The Doppler Effect [8, 9] is detected when the relative speed is greater than 30 km/h, in other words, a speed variation in the extension of 3 to 4% of the sound speed in air $\approx$ 1200km/h or 340 m/s.

Doppler Effect in Beamforming

In beamforming measurements for moving sources, the Doppler Effect causes mainly three effects: the blurring of the images, frequency shift and amplitude shift.

The blur effect is caused by the constant change of angles in the integration time. The fluctuation of frequency and amplitude, are caused by the approaching and displacing of the sound source to array's position. Due to this issue a de-dopplerization procedure was adopted (further informations could be consulted in Fonseca, 2009 [10]). The procedure consists in creating a virtual grid on vehicle's sideline. This means the virtual grid moves with the vehicle and each grid point is a potential moving acoustic source, Figure 4.

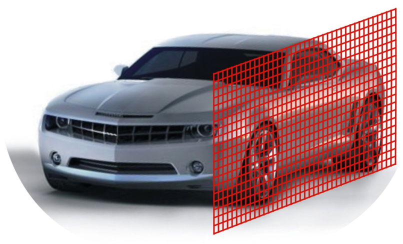


Figure 4: Virtual grid on vehicle's sideline.

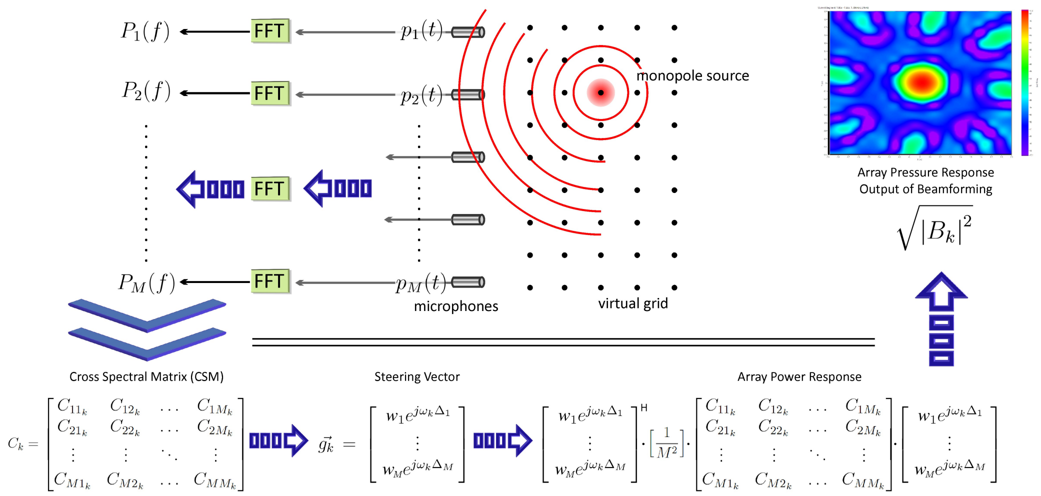


Figure 3: Frequency domain beamforming method.

In order to gain a better understanding of the de-dopplerization method, a comparison between two results for the same measurement at the same frequency is presented. The Figure 5 shows the result without correction of the Doppler effect, which is blurred and the amplitude does not express the real value of the pressure. Alternatively the Figure 6 shows the same result with de-dopplerization method applied.

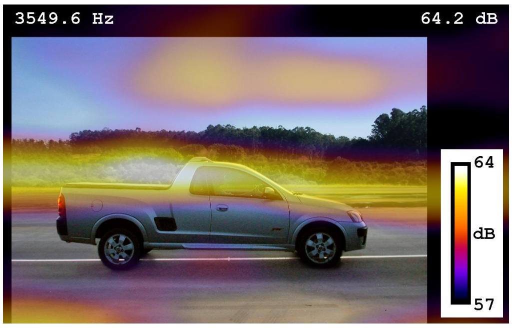


Figure 5: Beamforming data without de-dopplerization.

The Figures 5 and 6 are from the same measurement. This measurement was performed with buzzer, this is the reason the Figure 6 has only one direct sound source and a part of the reflection on the ground.

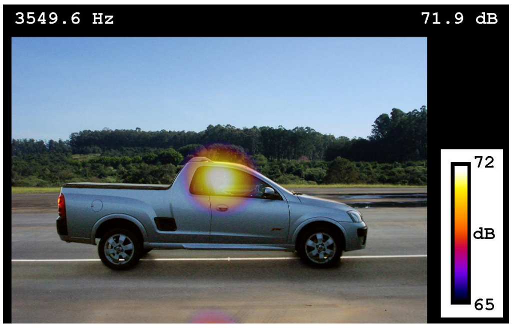


Figure 6: Beamforming data with de-dopplerization.

INSTRUMENTATION

In the beamforming implementation many microphones are used in the way to build one important tool of the technique, the microphone array. This array will sample the sound waves in a similar way as a digital camera sample the light waves. And all the post-processing is based in the microphone acquisitions.

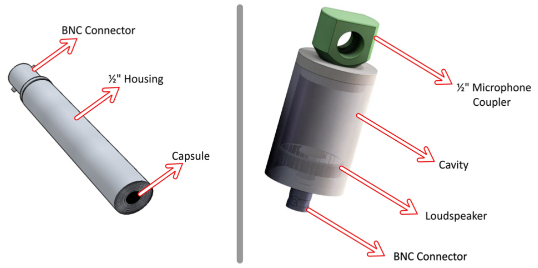
Microphones

As a microphone array has a great number of microphones the utilization of Type 1 measurement microphones [11] is difficult due to its expensive cost. Therefore a research was performed evaluate the cheaper possibilities. One of the best low cost options found is the Panasonic WM-61A electret microphone capsule [12]. This capsule meets the specification of omnidirectionality and low cost. And furthermore its applications are known in acoustics and microphone array field [13, 14]. The microphone was built in a 1/2” aluminum housing, with a simple gain circuit inside [15] and a BNC connector on the other extremity, Figure 7a.

Microphones Responses

To evaluate all the relative microphone responses a simple cavity with a loudspeaker and a coupler was developed, Figure 7b.

All the microphones were measured against one of the microphones built, microphone #45, Figures 8 and 9.



(a) (b)

Figure 7: (a) Microphone model. (b) Tool for evaluation of relative microphone responses.

A total of 50 microphones were built. The Figure 8 shows the magnitude relative response and the Figure 9 the phase relative response. Analyzing these two Figures the 32 more similar microphones were selected for use. The unshaded area shows the selection criterion.

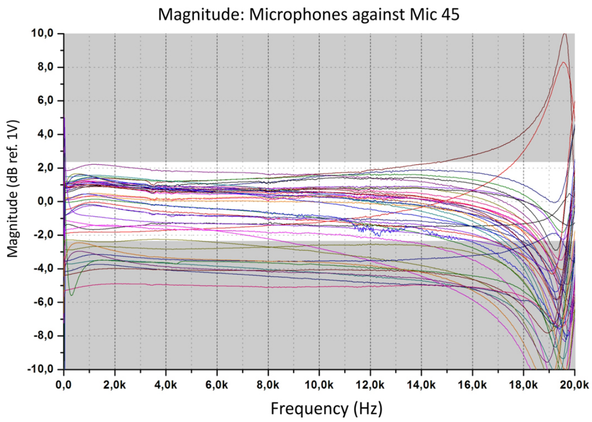


Figure 8: Magnitude: Microphones against Mic. #45.

It is possible to notice that up to frequencies of 14k Hz the deviations inside the unshaded part are small, i. e., ± 2.5dB V in magnitude and $\pm 4^{o}$ in phase. Some microphones were discarded by phase and some others by magnitude criterion.

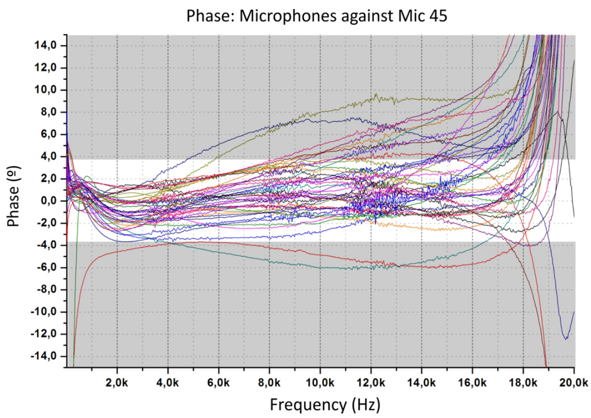


Figure 9: Phase: Microphones against Mic. #45.

As the microphones do not have the same sensitivities, their susceptibility to temperature were also evaluated. All the microphones were tested and it was possible to observe that all of them have the same behavior. A temperature sweep was performed in a temperature chamber starting with 5^{o}C and ending in 40^{o}C. At each 5^{o}C the sensibilities was tested with a calibrator of 94 dB, 1k Hz. The Figure 10 shows one of these results, in this case, for the microphone #45.

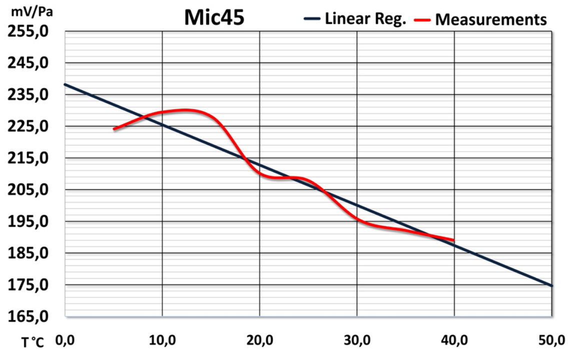


Figure 10: Microphone #45: Sensibility *x* Temperature. Measurements and Linear regression.

Array

The array was built with an aluminum structure and its dimensions are 1.5 *x* 1.2 m. A spiral based geometry with diameter of 1.0 m was utilized in all the results presented in this work. The 32 microphones are mounted in this structure which can be considered acoustically transparent up to frequencies near 15k Hz. The Figure 11 shows the complete array.

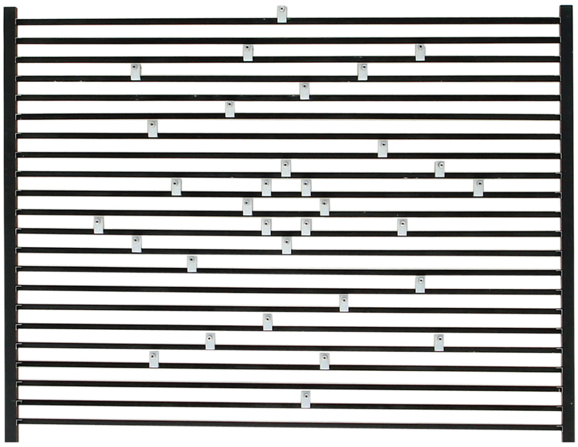


Figure 11: The 32 microphone spiral array.

All the 32 microphones are pre-calibrated enabling absolute sound pressure results. Complete information about the system and the geometry performance can be found in [10].

Data Acquisition System

The data acquisition system is based on *National Instruments* [16, 17] portable hardware which can acquire 32 channels simultaneously at a rate of 50k Hz, Figure 12. All the beamforming measurements were performed with this maximum rate.

(a) (b)

Figure 12: Data acquisition system. (a) NI-9233, 4 channels. (b) NI-cDAQ-9172 with 8 modules.

The frequency response between the first channel and all the 31 others were measured to certify that the data acquisition system does not insert important errors.

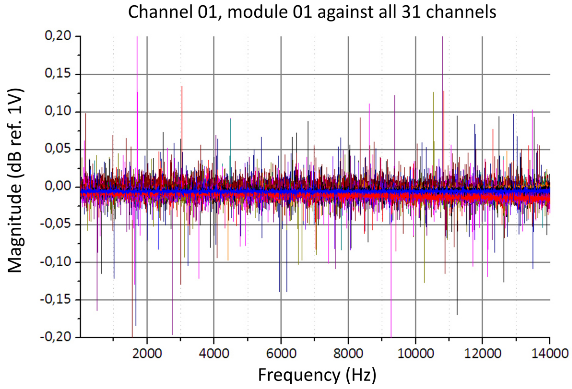


Figure 13: Mag.: Channel 1, module 1 against all 31 channels.

The Figure 13 shows the magnitude and the Figure 14 shows the phase between the channel 1 of the module 1 against all the 31 others channels. All the deviations evaluated are below the manufacturer specifications.

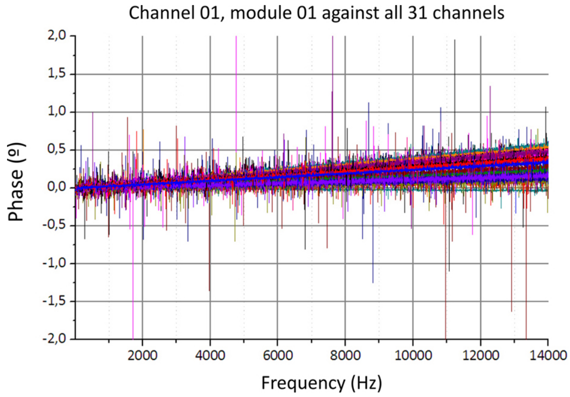


Figure 14: Phase: Channel 1, module 1 against all 31 channels.

If the phase errors are summed, i. e., the data acquisition system (Figure 14) with the microphones (Figure 9), they still remain under the 10º recommended by Mosher [14] for all the frequencies of interest.

The reference photos were taken by a digital camera placed at array's center. The photo calibration procedure can be reviewed in [18, 10].

PASS-BY NOISE TEST

The Pass-by Noise Test is a standardized test described in the standard ISO 362-1:2007 [19]. This test evaluates the sound pressure level of two microphones positioned 7.5 m far from the central line of the car's path, Figure 15. Every vehicle has to pass this test and satisfy the limits to join the vehicles market.

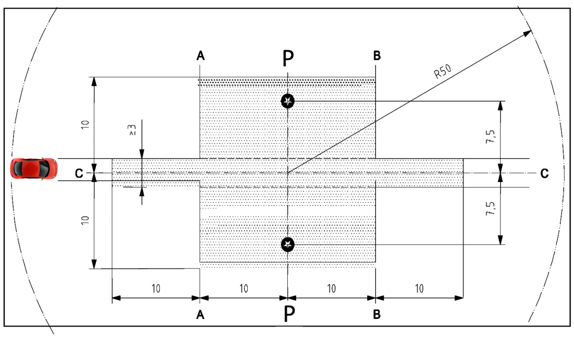


Figure 15: Pass-by Noise Test, standard layout [19].

The procedure of acceleration in the ISO 362-1:2007 describes:

> “The vehicle shall approach AA’ at a constant speed corresponding to 50 km/h. ... When the front of the vehicle reaches AA’, the acceleration control unit shall be fully engaged and held fully engaged until the rear of the vehicle reaches BB’. The acceleration control unit shall then be released. ... The path of the centerline of the vehicle shall follow line CC’ as closely as possible throughout the entire test, from the approach to line AA’ until the rear of the vehicle passes line BB’ (see Figure 15).”

The starred PP positions in the Figure 15 are the microphones positions.

Experimental Setup

The procedures of acceleration and displacement were preserved the same as described in the ISO362-1:2007. The array positioning was the same as the single microphone position in the standardized test, i.e., 7.5 m from the central line with its center 1.2 m from the ground. The data used in the post processing were cut and de-dopplerized from -5.0 m to +5.0 m, considering the array’s central line, Figures 16 and 17.

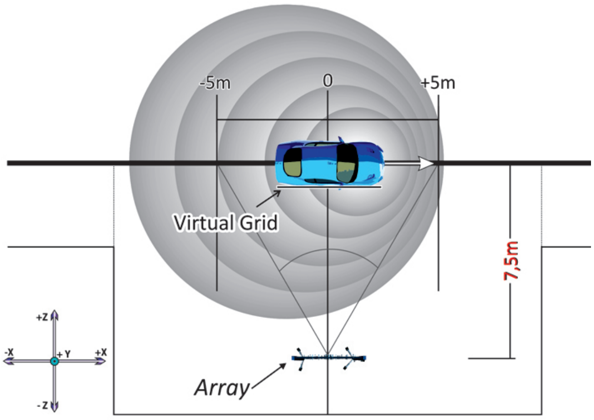


Figure 16: Pass-by Noise Test, beamforming layout.

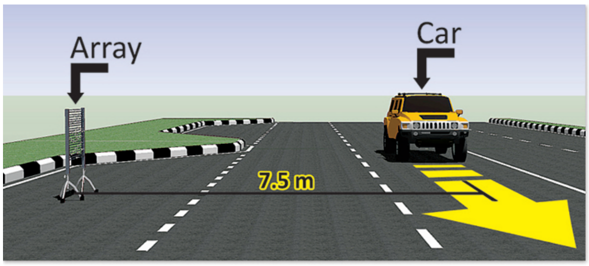


Figure 17: Pass-by Noise Test, beamforming layout. 3D View.

MEASUREMENTS AND RESULTS

Several pass-by beamforming were recorded and some of these results will be shown in this section. The distance of 50 m with no buildings was preserved in all measurements, Figure 15. However the special asphalt specification predicted in the standard was not attended in all of them.

A brief comment will be given in the presented results, however it is important to notice that a more precise analysis has to be done by a specialist of the vehicle. Moreover if a vehicle is being tested to find some problems or even to identify the prominent sources, some repetitions can improve the confidence of the analysis.

In all pass-by tests the beamforming was measured from left to right (LR) and from right to left (RL) due to the fact the vehicles have directional sources and different sources in their left and right sides.

Hammer H3

Two different models of Hummer H3 were measured and some of their results are presented from Figure 18 to Figure 21. The differences between these models include engine and exhaust system.

Figure 18: H3 Silver: LR, 1.3k Hz, 1/12 octave band.

It is possible to observe in the case of LR that the overall noise is quite the same, however the prominent sources are different. The engine noise is dominant in relation to the exhaust system, this is more evident in the Figure 19 than the Figure 18.

Figure 19: H3 Yellow: LR, 1.3k Hz, 1/12 octave band.

Now in the case of RL the end of the exhaust system has a greater contribution than the LR, Figures 20 and 21.

Figure 20: H3 Silver: RL, 1.3k Hz, 1/12 octave band.

Figure 21: H3 Yellow: RL, 1.3k Hz, 1/12 octave band.

In the Figure 20 it can be observed that the end of the exhaust system and the engine have similar levels (near to 60 dB).

However in the Figure 21 the engine noise is still dominant over the exhaust system. This is possibly by the strong “knocking” of the vehicle’s engine.

All engines’ noise are on the ground due to the directivity of the sound. As the H3 is relatively high from the ground it is easier to see its sound pattern (direct or reflected) than a small vehicle closer to the ground.

Commonly the engine noise is mixed with other sources, such as rolling tires, as evident in the Figure 18.

Volkswagen Gol

A Volkswagen Gol was measured and it is important to know that its engine, suspension and exhaust system are not the originals. This vehicle has a very characteristic noise, two results are presented, Figures 22 and 23.

Figure 22: Gol: LR, 1k Hz, 1/12 octave band.

This vehicle has only two strong sources, the engine and the end of the exhaust system, one in each side of the vehicle. All the others possible sources are not able to identify due to its modified characteristics. Its “purr” noise is very strong and even in other frequencies only these two sources are noticed.

Figure 23: Gol: RL, 1k Hz, 1/12 octave band.

FUTURE WORK

The system will be expanded to 48 and 64 microphones with the change to an acquisition system capable to sample all the channels. Further researches about the sound attenuation and sound velocity related to beamforming results are ongoing. And for the pass-by application triggering and tracking are being improved.

CONCLUSIONS

The construction of a big part of the whole system can give some essential information about the possible problems and about next steps to improve it.

The use of low cost microphones proved that they are capable to carry out this technique if some cautions are taken. It is important to evaluate all the systems’ responses to prevent mistakes in the measurements and in the post-processing.

Beamforming is a suitable technique to evaluate sound sources in the pass-by noise test.

ACKNOWLEDGMENTS

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