

Degree in Engineering in Telecommunications Technologies

Bachelor's final project

OBSIDAS: On-Board Sound Intensity Data Acquisition System
LabVIEW Implementation of the AASHTO T360 Standard for Signal Acquisition and Processing for Tire-Pavement Noise Measurement

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Author: Javier González Santamaría
Director: Alan Pisano, Ph.D. — Boston University

Madrid 2024

Declaro, bajo mi responsabilidad, que el Proyecto presentado con el título

OBSIDAS: On-Board Sound Intensity Data Acquisition System

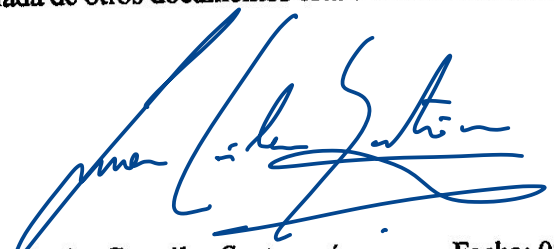
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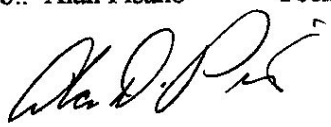


Fdo.: Javier González Santamaría Fecha: 08/05/2024

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EL DIRECTOR DEL PROYECTO

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Madrid 2024

To my parents and sisters:
you are the cornerstone of my life

OBSIDAS: SISTEMA DE ADQUISICIÓN DE INTENSIDAD DE SONIDO A BORDO, DESARROLLO EN LABVIEW DEL ESTÁNDAR AASHTO T360 PARA LA ADQUISICIÓN Y PROCESAMIENTO DE SEÑAL PARA LA MEDICIÓN DEL RUIDO ENTRE NEUMÁTICO Y ASFALTO

Autor: González Santamaría, Javier.
Director: Pisano, Alan.
Entidad Colaboradora: John A. Volpe National Transportation Systems Center, Departamento de Transporte de los Estados Unidos de América.

RESUMEN DEL PROYECTO

Este proyecto se centra en la medición y cuantificación del ruido producido por los vehículos entre el neumático y el asfalto. OBSIDAS: Sistema de Adquisición de Intensidad de Sonido a Bordo, es un programa de software desarrollado en LabVIEW, el cual está diseñado para adquirir la señal de ruido, calcular las mediciones de sonido necesarias y mostrar gráficamente los resultados para implementar el estándar AASHTO (American Association of State Highway and Transportation Officials) T360. OBSIDAS no solo se adhiere a todos los procedimientos del T360, sino que también presenta los resultados en gráficas e indicadores en una interfaz de usuario intuitiva. Además, realiza mediciones de nivel y calidad de sonido adicionales, así como análisis de frecuencia, ofreciendo al usuario más información sobre la prueba OBSI. Los distintos pasos por los que el programa desarrolla el estudio son los siguientes: primero, el programa adquiere la señal a través de los drivers de la plataforma LabVIEW pertinentes; segundo, el programa realiza todos los cálculos que se detallan en el estándar ya nombrado y los procesos adicionales que el programa ofrece al usuario; tercero, el programa expondrá gráficamente todos los resultados de los cálculos a través de la interfaz de usuario; cuarto, el usuario tendrá la posibilidad de seleccionar si los datos originales de la señal adquirida y los datos procesados calculados se guarden en un archivo tipo CSV. Cabe resaltar que, los requisitos que detalla el estándar AASHTO, los resultados que se añadirán al CSV de los datos obtenidos, son los siguientes: los niveles normalizados de intensidad sonora con ponderación A en bandas de un tercio de octava para cada micrófono, la media aritmética en cada par de micrófonos de los niveles de intensidad sonora en cada banda, la media de energía de los niveles de intensidad sonora entre pares, el índice de presión de intensidad en cada micrófono, nivel OBSI de cada micrófono, nivel OBSI en cada par de micrófonos y nivel OBSI total del test.

Palabras clave: LabVIEW, OBSI, AASHTO, ruido neumático-asfalto, nivel de

presión sonora, ponderación A, banda de un tercio de octava.

Introducción

Actualmente, residir en el centro de la ciudad, o incluso en la parte suburbana está asociado con el ruido que produce el tráfico de vehículos. Como resalta el autor de [1], las principales fuentes de ruido urbano son el tráfico y la industria.

El ruido realizado por los vehículos, ya sean automóviles, camiones, autobuses o motocicletas, en las carreteras multiplica las posibilidades de padecer hipertensión, según un estudio realizado por la revista *JACC: Advances* [2], liderado por Jing Huan, profesor de la Universidad de Pekín.

Este proyecto: OBSIDAS: Sistema de Adquisición de Intensidad de Sonido a Bordo, se centra en la medición y cuantificación del ruido producido por los vehículos entre el neumático y el asfalto. Este ruido es el contribuidor más significativo al total del ruido producido por el tráfico [3].

Por lo tanto, la medición del ruido de las carreteras es primordial para identificar las áreas que necesitan la mayor mejora de condiciones. Es por ello, que cuando se requiera de una mejora o un cambio en una zona, OBSIDAS es el instrumento ideal para poder cuantificar esta mejora en el ruido, y decretar si ha sido satisfactoria.

OBSIDAS es un programa de software desarrollado en LabVIEW el cual está diseñado para adquirir la señal de ruido, calcular las mediciones de sonido necesarias y mostrar gráficamente los resultados para implementar el estándar AASHTO (American Association of State Highway and Transportation Officials) [4] T360 [5].

OBSIDAS no solo se adhiere a todos los procedimientos del T360, sino que también presenta los resultados en gráficas e indicadores en una interfaz de usuario intuitiva. Además, realiza mediciones de nivel y calidad de sonido adicionales, así como análisis de frecuencia, ofreciendo al usuario más información sobre la prueba OBSI.

Los distintos pasos por lo que el programa desarrolla el estudio son los siguientes: primero, el programa adquiere la señal a través de los drivers de la plataforma LabVIEW pertinentes; segundo, el programa realiza todos los cálculos que se detallan en el estándar ya nombrado y los procesos adicionales que el programa ofrece al usuario; tercero, el programa expondrá gráficamente todos los resultados de los cálculos a través de la interfaz de usuario; cuarto, el usuario tendrá la posibilidad de seleccionar si los datos originales de la señal adquirida y los datos procesados calculados se guarden en un archivo tipo CSV.

Los programas de software para la medición del ruido asfalto-neumático producido por los vehículos que actualmente se encuentran en el mercado son los siguientes: “OBSI Module: Real-Time On Board Sound Intensity measurement and post-processing” [6] de la empresa SoftdB y el “OBSI System” de la em-

presa AVEC (Acoustical and Vibration Engineering Consultants) [7]. Además, tecnologías y sistemas que no se basan en el sistema OBSI pero que se relacionan y que pueden llegar a usarse para realizar dicho sistema también se detallarán, como por ejemplo los sonómetros o medidores de nivel de sonido.

Las tecnologías nombradas son escasas y no están actualizadas. Por ejemplo, la tecnología de SoftdB se basa en el estándar AASHTO T-76, el cual es un estándar provisional y el antecedente del AASHTO T360. Es decir, SoftdB se adhiere a un estándar que no se usa actualmente. Por otro lado, el programa de la empresa AVEC está diseñado para funcionar exclusivamente en el sistema operativo Windows 7, el cual Microsoft ha dejado de hacer actualizaciones desde 2012.

La principal motivación del proyecto ha sido desarrollar un nuevo sistema que se adhiera al actual estándar T360 y que presente una mayor compatibilidad.

El software OBSIDAS es accesible y descargable en cualquier máquina actual. Su propósito es cuantificar y comprender mejor el problema del ruido de automóviles en las carreteras. También tiene como objetivo obtener los datos necesarios y por ende, ayudar a descubrir nuevas tecnologías para reducir el nivel de ruido en las carreteras.

Metodología

Durante todo el proceso de trabajo se usarán las prácticas del método *agile* [8], el cual se basa en el esfuerzo colaborativo entre el usuario final y los desarrolladores, la planificación adaptativa a dicho feedback por parte de los usuarios finales, el desarrollo evolutivo, la entrega temprana y la mejora continua. Por lo tanto, al ser John A. Volpe National Transportation Systems el cliente principal de este software, se tendrán en cuenta su feedback y propuestas de mejora. Se incorporarán al proyecto durante su producción en vez de esperar a la última versión. Esta metodología ofrece facilidades a ambas partes, ya que el desarrollador conoce qué quiere y cómo lo quiere el usuario, y este puede dar su opinión y valoración de las primeras versiones del software.

El procedimiento principal empleado durante todo el proyecto para abordar un problema o implementar una característica ha sido el siguiente.

1. Establecer un plan en el plazo necesario para lograr el objetivo.

2. Revisar e investigar las bases teóricas de la nueva característica; estableciendo la base teórica, cada nuevo desarrollo puede ser abordado con un conocimiento completo de este. Cabe destacar que la página web `https://sengpielaudio.com/` del ingeniero de sonido alemán Eberhard Sengpiel, ganador de dos Premio Grammy, ha sido de gran valor y orientación para establecer los conocimientos teóricos fundamentales.

3. Implementación de la característica en LabVIEW, cabe resaltar que el *Sound and Vibration Toolkit User Manual* de LabVIEW ha sido de gran valor y guía para establecer el conocimiento fundamental sobre los procesos de procesamiento de señales que se requiere para plasmarlos en LabVIEW correctamente.

4. Probar la nueva característica.

5. Contactar con el cliente para obtener comentarios y feedback sobre la nueva funcionalidad.

Interfaz de Usuario

La interfaz se divide en dos tabs:

1. Acquisition Tab; la cual contiene los menús necesarios para, por ejemplo, seleccionar el modo de la aplicación, la duración del test o acceder a los botones y la barra de progreso. En la figura 1 se tiene esta acquisition tab.

2. Results Tab; en la cual están reflejados todos los resultados numéricos y gráficas de los mismos. En las figuras 2 y 3 se tiene esta results tab.

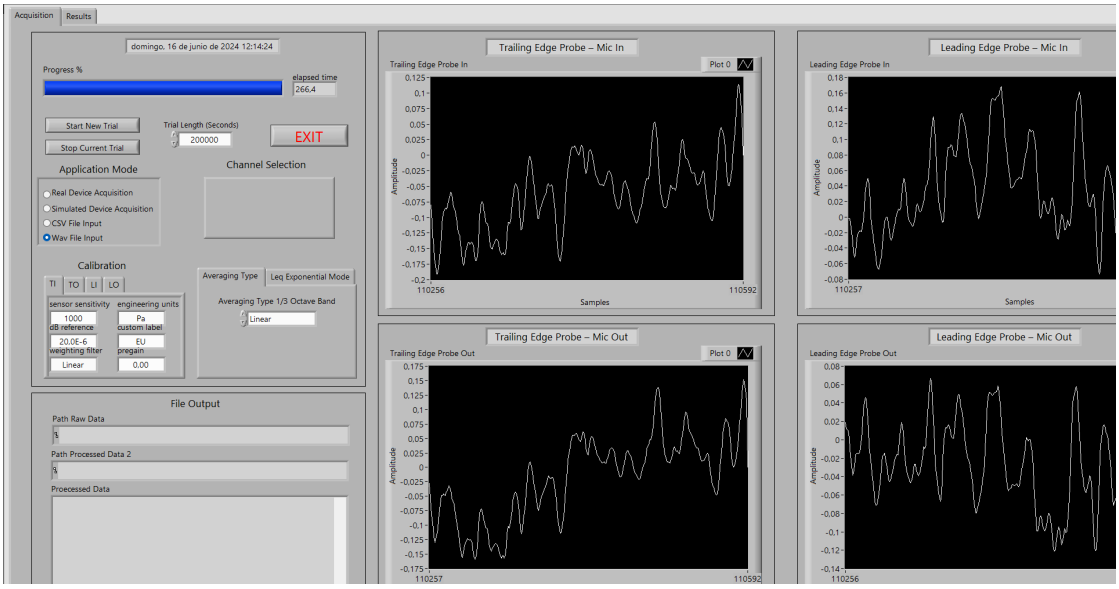


Figure 1: Acquisition Tab de la Interfaz de Usuario de OBSIDAS. Fuente: elaboración propia.

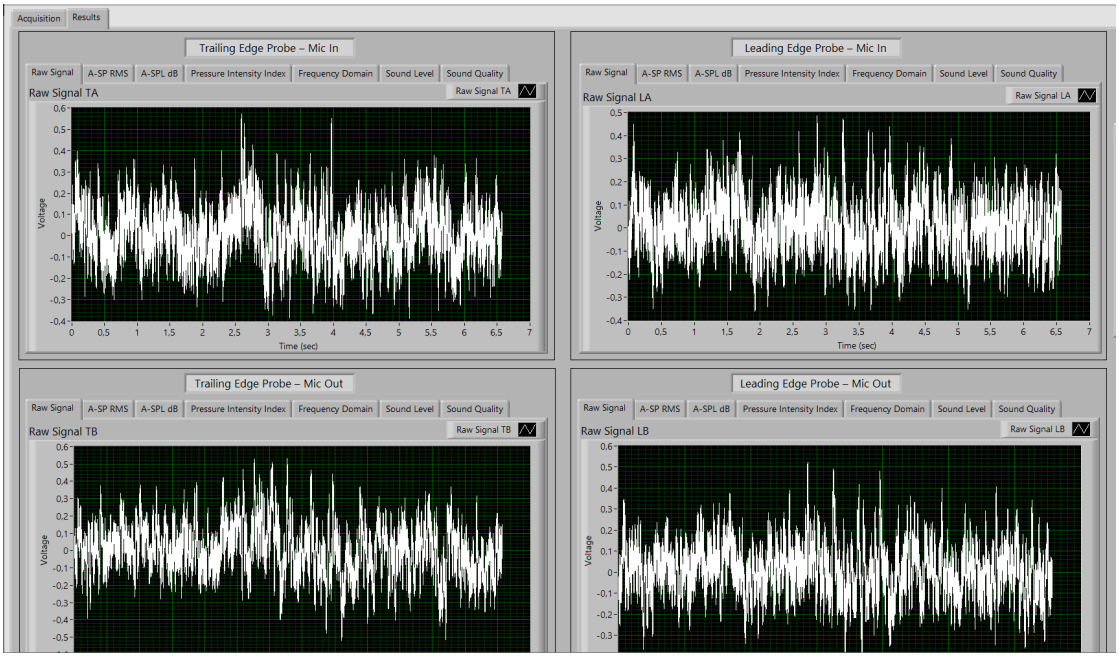


Figure 2: Results Tab de la Interfaz de Usuario de OBSIDAS Parte I. Fuente: elaboración propia.

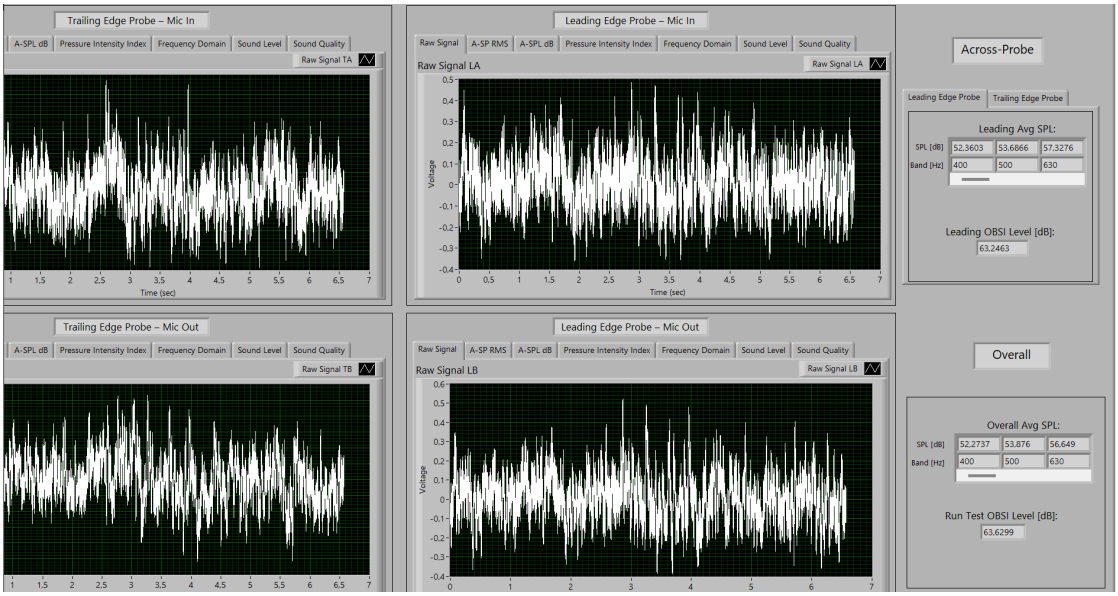


Figure 3: Results Tab de la Interfaz de Usuario de OBSIDAS Parte II. Fuente: elaboración propia.

Resultados

En esta sección se pretende resumir el análisis de un ensayo OBSI real. Este proyecto se ha desarrollado para poder realizar este tipo de ensayos OBSI. En este análisis, cada característica del programa OBSIDAS será utilizada. El CSV con los resultados de este se encuentra en la figura 4.

	A	B	C	D	E	F	G	H	I	J	K	L	M
1	file name:	VOLPE OBSI CALIBRATED_processed.csv											
2	timestamp:	sábado, 15 de junio de 2024 13:46:20											
3	LI Normalized A-weighted One-Third Octave Band SPL:	91,301976	92,604261	96,245959	97,362593	92,909724	89,316975	85,854033	83,79025	80,040521	76,004942	70,908507	67,387878
4													
5	LO Normalized A-weighted One-Third Octave Band SPL:	90,251464	91,601761	95,242009	96,477338	91,830852	88,25002	84,707115	82,664948	78,974372	75,298579	70,147296	66,567656
6													
7	TI Normalized A-weighted One-Third Octave Band SPL:	87,928463	89,777194	91,531521	94,949731	92,629287	89,275473	84,632828	80,703865	78,365418	75,170118	70,500393	66,336229
8													
9	TO Normalized A-weighted One-Third Octave Band SPL:	86,79204	88,687474	90,506821	93,967743	91,450661	88,109374	83,409274	79,472757	77,180293	74,19863	69,402092	64,848639
10													
11	Across Leading Probe Arithmetic Averaged SPL:	90,77672	92,103011	95,743984	96,919966	92,370288	88,783497	85,280574	83,227599	79,507446	75,651761	70,527901	66,977767
12													
13	Across Leading Probe Arithmetic Averaged SPL:	87,360251	89,232334	91,019171	94,458737	92,039974	88,692424	84,021051	80,088311	77,772855	74,684374	69,951242	65,592434
14													
15	Energy Averaged Overall SPL:	89,396119	90,900665	93,994718	95,86142	92,20827	88,738199	84,696314	81,93564	78,726182	75,194947	70,249136	66,340105
16													
17	Pressure Intensity Index LI:	-0,525256	-0,50125	-0,501975	-0,442627	-0,539436	-0,533478	-0,573459	-0,562651	-0,533075	-0,353182	-0,380605	-0,410111
18													
19	Pressure Intensity Index LO:	0,525256	0,50125	0,501975	0,442627	0,539436	0,533478	0,573459	0,562651	0,533075	0,353182	0,380605	0,410111
20													
21	Pressure Intensity Index TI:	-0,568211	-0,54486	-0,51235	-0,490994	-0,589313	-0,58305	-0,611777	-0,615554	-0,592562	-0,485744	-0,549151	-0,743795
22													
23	Pressure Intensity Index TO:	0,568211	0,54486	0,51235	0,490994	0,589313	0,58305	0,611777	0,615554	0,592562	0,485744	0,549151	0,743795
24													
25	OBSI Level LI, LO, TI, TO:	102,15478	101,170544	99,695529	98,618782								
26													
27	Across Leading-Trailing Probe Arithmetic Averaged OBSI Level:	101,662662	99,157155										
28													
29	RUN TEST OBSI LEVEL [dB]:	100,5881											

Figure 4: CSV con los Resultados de un Test Real Usando OBSIDAS. Fuente: elaboración propia

La medida principal para definir y establecer si una carretera es ruidosa es el nivel OBSI. Solo con esta medida se podrían sacar conclusiones respecto al ensayo general.

Se resalta que cada uno de los micrófonos tiene un nivel OBSI distinto. El micrófono con el nivel mayor es el micrófono in del par delantero. Seguido del micrófono out del mismo par. El micrófono con el siguiente nivel menor es el micrófono in del par trasero. Y por último, el micrófono con el nivel OBSI menor es el micrófono out del par trasero. En cada par de micrófonos (delanteros y traseros) se encuentra un cambio de nivel de 1 dB.

Como se verá en la sección con los resultados completos, el par de micrófonos delanteros presenta más ruido que los traseros. Se puede observar una diferencia de 1.5 dB entre los dos pares. Esto demuestra la precisión del software OBSIDAS, aunque la diferencia sea mínima, esta se refleja en los resultados..

Por último, el nivel total OBSI del test es de 100,5881 dB. El cual es relativamente alto, como se verá en el desglose completo de los resultados. Por lo tanto, se recomienda implementar mejoras en la calidad del asfalto para hacerlo más silencioso.

Conclusiones

El objetivo principal de este proyecto es el de implementar un programa de software desarrollado en LabVIEW, diseñado para adquirir la señal de ruido, calcular y realizar el procesamiento de señales necesarios y mostrar gráficamente los resultados. Todo esto basándonos y siguiendo el estándar AASHTO T360.

OBSIDAS no solo se adhiere a todos los procedimientos del T360, sino que también presenta los resultados en gráficas e indicadores en una interfaz de usuario intuitiva. Además, realiza mediciones de nivel y calidad de sonido adicionales, así como análisis de frecuencia, ofreciendo al usuario más información sobre la prueba OBSI. En conclusión, cada uno de los objetivos ha sido cumplido.

Con la creación de este software se ha contribuido al sector. Se ha logrado desarrollar una herramienta la cual obtiene los datos necesarios para descubrir nuevas tecnologías destinadas a reducir el nivel de ruido en las carreteras.

Como conclusión, este proyecto contribuye a la investigación para la reducción del ruido que causan los vehículos en la carretera, ayudando a mitigar las consecuencias negativas que se ha visto que produce tanto en los humanos como en la fauna local.

Referencias

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OBSIDAS: ON-BOARD SOUND INTENSITY DATA ACQUISITION SYSTEM; LABVIEW IMPLEMENTATION OF THE AASHTO T360 STANDARD FOR SIGNAL ACQUISITION AND PROCESSING FOR TIRE-PAVEMENT NOISE MEASUREMENT

Author: González Santamaría, Javier.
Supervisor: Pisano, Alan.
Collaborating Entity: John A. Volpe National Transportation Systems Center, United States Department of Transportation

ABSTRACT

This project is focused on the measurement and quantification of the tire-pavement noise made by a vehicle. OBSIDAS: On-Board Sound Intensity Data Acquisition System, is a software program developed in LabVIEW that is designed to acquire the noise signal, compute the necessary sound measurements, and graphically display the results in order to perform the AASHTO (American Association of State Highway and Transportation Officials) standard T360. OBSIDAS not only adheres to all T360 procedures, but also presents the results in graphs and schemes in a highly intuitive user interface. Furthermore, it performs additional sound quality and level measurements, as well as frequency analysis, thereby offering the user more information about the OBSI test. The phases of this project can be delineated as follows: first, the acquisition of the signal through the pertinent LabVIEW drivers; second, the computation of all T360 processes and additional processes; third, all the computed data and results will be graphically exposed to the user via UI; fourth, the user may select the option to download the raw data acquired by the microphone and the processed computation into a CSV-type file. It is noteworthy that the requisite AASHTO standard measurements, the results which are added into the CSV file are as follows: Single Microphone Normalized A-weighted One-Third Octave Band SPL, Across-Probe Arithmetic Averaged SPL, Energy Averaged Overall SPL, Single Microphone Pressure Intensity Index, Single Microphone OBSI Levels, Across-Probe OBSI Levels, and Overall OBSI Level.

Key words: LabVIEW, OBSI, AASHTO, tire-pavement noise, sound pressure level, A-weighting, one-third octave band

Introduction

Currently, residing in the city center or even in suburban areas is often associated with the noise generated by road traffic. As observed by the author of [9], the primary sources of urban noise are traffic and industry.

Not only is noise itself an unpleasant experience, but it can also contribute to the development of physical illness and other health problems. The research conducted by Professor Jung Huan of Peking University and the University of Rochester, published in the journal *JACC: Advances* [10], elucidates the high correlation between traffic noise and the incident of primary hypertension (high blood pressure).

This project, OBSIDAS: On-Board Sound Intensity Data Acquisition System, is focused on the measurement and quantification of the tire-pavement noise made by a vehicle. This noise is the most significant contributor to the overall traffic noise environment [11].

Consequently, the measurement of road noise is of paramount importance in order to identify the areas that require the most improvement. Furthermore, when a change or improvement to the pavement is to be implemented, OBSIDAS is an ideal instrument for determining whether the upgrade has resulted in a reduction in noise.

OBSIDAS, is a software program developed in LabVIEW that is designed to acquire the noise signal, compute the necessary sound level measurements, and graphically display the results in order to perform the AASHTO (American Association of State Highway and Transportation Officials) [12] standard T360 [13].

OBSIDAS not only adheres to all T360 procedures, but also presents the results in graphs and schemes in a highly intuitive user interface. Furthermore, it performs additional sound quality and level measurements, as well as frequency analysis, thereby offering the user more information about the OBSI test.

The phases of this project can be delineated as follows: first, the acquisition of the signal through the pertinent LabVIEW drivers; second, the computation of all T360 processes, and additional processes, which will be explained in the following sections; third, all the computed data, and results will be graphically exposed to the user via UI; fourth, the user may select the option to download the raw data acquired by the microphone and the processed computation into a CSV-type file.

The current software and systems products available in the market for measuring tire-pavement noise using the OBSI methods are as follows: "OBSI Module: Real-Time On Board Sound Intensity measurement and post-processing" [14] from the SoftdB company and the "OBSI System" from AVEC (Acoustical and Vibration Engineering Consultants) [15]. Additionally, technologies and systems not based on the OBSI method but related to sound intensity measurements are also offered in the market, such as sound level meters.

The current technologies that address this significant issue are scarce and outdated. For instance, SoftdB's technology adheres to the standard AASHTO TP-76, which is a provisional standard antecedent to the AASHTO T360. In other words, SoftdB is based on an outdated standard. Furthermore, AVEC OBSI program is designed to operate exclusively within the operating system Mirosoft Windows 7 version, which Microsoft has ceased to provide updates for since 2012.

The principal motivation of the project has been to develop the requisite software to implement an updated OBSI method that adheres to the AASHTO T360 standard. This has involved the creation of a program.

The software is accessible and downloadable on any current machine. Its purpose is to quantify and better understand the problem of automotive noise on the roads. It must also obtain the necessary data and discover new technologies to reduce the noise level on the roads.

Methodology

The agile method [16], which is based on the collaborative effort between the end user and the developers, adaptive planning to such feedback from the end users, evolutionary development, early delivery, and continuous improvement, will be employed throughout the work process. Therefore, since John A. Volpe National Transportation Systems is the main customer of this software, their feedback and proposals for improvement will be taken into account. These inputs will be integrated into the project during its production phase, rather than being incorporated into the latest version. This methodology offers benefits to both parties, as the developer is able to idenitfy the user's needs and preferences, while the user is able to provide feedback and assessment of the initial software versions.

The primary procedure employed throughout the project to address an issue or implement a feature has been as follows.

1. Establish a plan for the necessary time frame to achieve the objective.

2. Review the theoretical background of the new feature or issue; by establishing the theoretical basis, every new objective can be addressed with complete knowledge of this objective. It is noteworthy that the webpage `https://sengpielaudio.com/` by the Grammy Award-winning German sound engineer Eberhard Sengpiel has been of great value and guidance in order to establish the fundamental theoretical knowledge.

3. Implementing the feature in LabVIEW; it is of note that the *Sound and Vibration Toolkit User Manual* by LabVIEW has been of great value and guidance in establishing the fundamental knowledge regarding signal and sound processing measurements that is required to use LabVIEW effectively.

4. Test the new feature.

5. Contact the customer to obtain feedback on the newly added feature.

User Interface

The interface developed is divided into two main tabs:

1. Acquisition Tab; contains menus to select the application mode, control trial length, and access buttons and the progress bar, among other features. Figure 5 shows the representation of this acquisition tab.

2. Results Tab; reflects all the graphical and numerical computations through graphs and indicators. Figure 6 and 7 show the representation of this results tab.

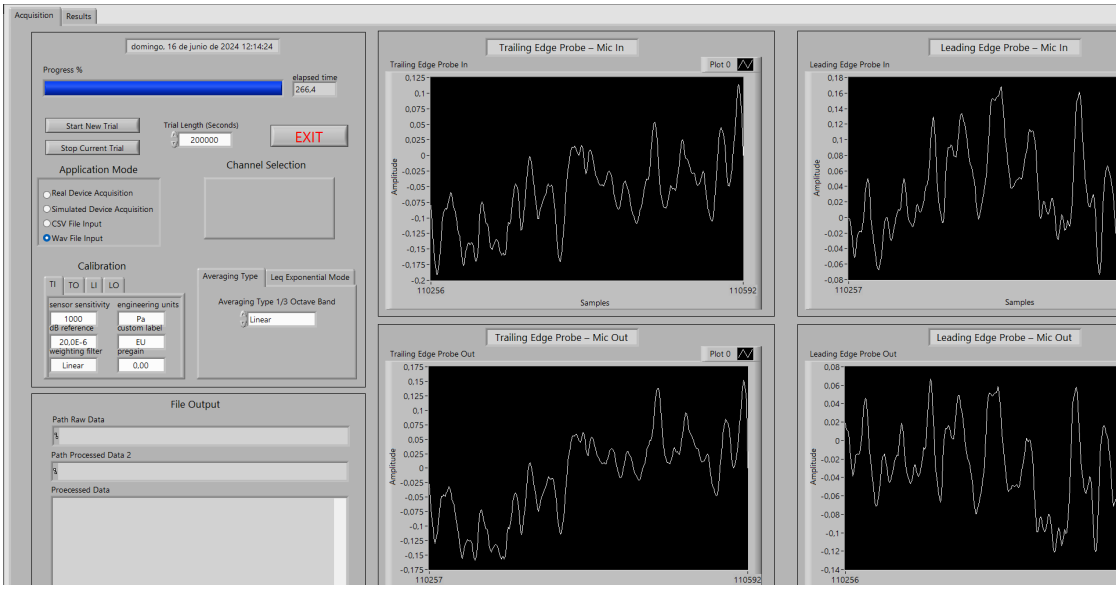


Figure 5: Acquisition Tab of the OBSIDAS User Interface. Source: own elaboration

Results

This section will analyze a real-world OBSI trial that adheres to the AASHTO standard. This project has been developed in order to be able to perform this

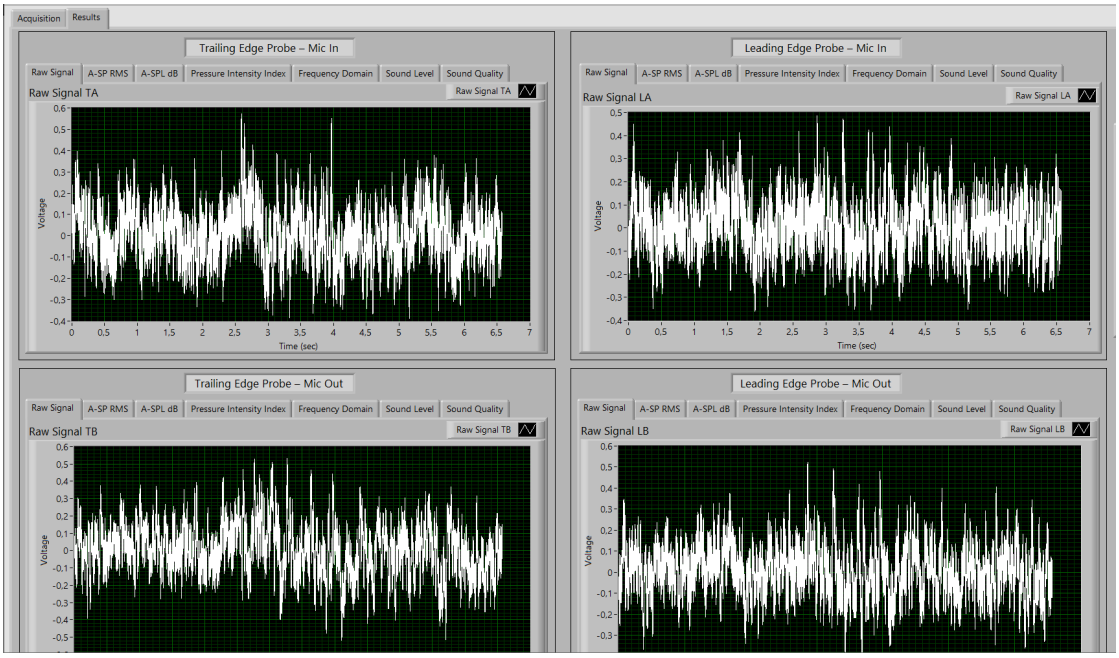


Figure 6: Results Tab of the OBSIDAS User Interface Part I. Source: own elaboration

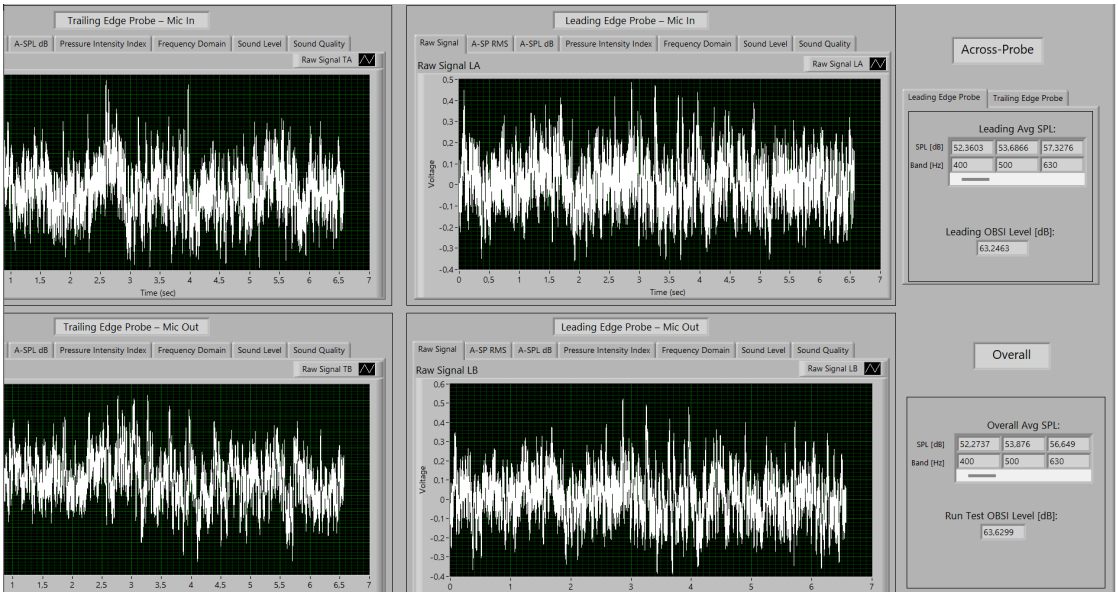


Figure 7: Results Tab of the OBSIDAS User Interface Part II. Source: own elaboration

type of OBSI trials and to conduct the required measurements. In this analysis, every feature of the OBSIDAS program will be utilized. The CSV file with the results of the trial is shown in figure 8.

	A	B	C	D	E	F	G	H	I	J	K	L	M
1	file name:	VOLPE OBSI CALIBRATED_processed.csv											
2	timestamp:	sábado, 15 de junio de 2024 13:46:20											
3	LI Normalized A-weighted One-Third Octave Band SPL:	91,301976	92,604261	96,245959	97,362593	92,909724	89,316975	85,854033	83,79025	80,040521	76,004942	70,908507	67,387878
4													
5	LO Normalized A-weighted One-Third Octave Band SPL:	90,251464	91,601761	95,242009	96,477338	91,830852	88,25002	84,707115	82,664948	78,974372	75,298579	70,147296	66,567656
6													
7	TI Normalized A-weighted One-Third Octave Band SPL:	87,928463	89,777194	91,531521	94,949731	92,629287	89,275473	84,632828	80,703865	78,365418	75,170118	70,500393	66,336229
8													
9	TO Normalized A-weighted One-Third Octave Band SPL:	86,79204	88,687474	90,506821	93,967743	91,450661	88,109374	83,409274	79,472757	77,180293	74,19863	69,402092	64,848639
10													
11	Across Leading Probe Arithmetic Averaged SPL:	90,77672	92,103011	95,743984	96,919966	92,370288	88,783497	85,280574	83,227599	79,507446	75,651761	70,527901	66,977767
12													
13	Across Leading Probe Arithmetic Averaged SPL:	87,360251	89,232334	91,019171	94,458737	92,039974	88,692424	84,021051	80,088311	77,772855	74,684374	69,951242	65,592434
14													
15	Energy Averaged Overall SPL:	89,396119	90,900665	93,994718	95,86142	92,20827	88,738199	84,696314	81,93564	78,726182	75,194947	70,249136	66,340105
16													
17	Pressure Intensity Index LI:	-0,525256	-0,50125	-0,501975	-0,442627	-0,539436	-0,533478	-0,573459	-0,562651	-0,533075	-0,353182	-0,380605	-0,410111
18													
19	Pressure Intensity Index LO:	0,525256	0,50125	0,501975	0,442627	0,539436	0,533478	0,573459	0,562651	0,533075	0,353182	0,380605	0,410111
20													
21	Pressure Intensity Index TI:	-0,568211	-0,54486	-0,51235	-0,490994	-0,589313	-0,58305	-0,611777	-0,615554	-0,592562	-0,485744	-0,549151	-0,743795
22													
23	Pressure Intensity Index TO:	0,568211	0,54486	0,51235	0,490994	0,589313	0,58305	0,611777	0,615554	0,592562	0,485744	0,549151	0,743795
24													
25	OBSI Level LI, LO, TI, TO:	102,15478	101,170544	99,695529	98,618782								
26													
27	Across Leading-Trailing Probe Arithmetic Averaged OBSI Level:	101,662662	99,157155										
28													
29	RUN TEST OBSI LEVEL [dB]:	100,5881											

Figure 8: OBSIDAS CSV Results of the OBSI Trial. Source: own elaboration

The primary and most important measurement to define and state whether a road or pavement is noisy or not is the OBSI level. If only one measurement from the trial is needed to draw conclusions, it would be the OBSI level.

It is noteworthy that the OBSI level of each microphone is distinct. The microphone with the highest decibel level is the leading edge probe microphone in, followed by the leading edge microphone out, along with the trailing edge probe microphone in. The trailing edge microphone out represents the least noisy levels. A change of approximately 1 dB can be observed in both microphones of a single probe.

As identified in the previous section, the across-probe OBSI level indicates that the leading edge probe is noisier than the trailing. A difference of more than 1.5 decibels can be observed. This demonstrates that, although the difference is not significant, OBSIDAS can detect minimum changes between microphones.

Finally, the overall run test OBSI level is 100,5881 dB. Which, compared to the table in figure 2.3 indicates that the road is highly noisy. Therefore, it is recommended that improvements in the quality of the pavement should be implemented in order to make it quieter. Note that this decibel level could be seen as high, however, the measurement is realized in a really close distance, so the pedestrians would not experience this level but rather a lower one, depending on the distance they are from the road.

Conclusions

The main goal of this project was to implement a software program developed in LabVIEW, designed to acquire the noise signal, compute the necessary sound level measurements, and graphically display the results in order to perform the AASHTO T360 standard.

The OBSIDAS program not only adheres to all AASHTO T360 procedures, but also presents the results in graphs and schemes in a highly intuitive user interface. Furthermore, it performs additional sound quality and level measurements, as well as frequency analysis, thereby offering the user more information about the OBSI test. In conclusion, each part of the objectives has been achieved.

This project has contributed to the sector by creating this software. Which is accessible and downloadable on any current machine. It has achieved its main purpose: to quantify and better understand the problem of automotive noise on the roads. Additionally, it obtains the necessary data in order to discover new technologies to reduce the noise level on the roads.

In conclusion, this project contributes to the investigation of the reduction of tire pavement noise, which is of paramount importance given its adverse health and welfare consequences for humans and wildlife.

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Chapter 1

Introduction

> Noise: sound or sounds, especially when it is unwanted, unpleasant, or loud.
>
> *Cambridge Dictionary [17]*

The proliferation of new modes of transportation, such as automobiles, trains, or passenger planes, has transformed urban areas into a highly noise-polluted environments. Currently, residing in the city center or even in suburban areas is often associated with the noise generated by road traffic and industry. As observed by the author of [18], the primary sources of urban noise are traffic and industry.

Not only is noise itself an unpleasant experience, but it can also contribute to the development of physical illness and other health problems. The research conducted by Professor Jung Huan of Peking University and the University of Rochester, published in the journal *JACC: Advances* [19], elucidates the high correlation between traffic noise and the incident of primary hypertension (high blood pressure).

Hypertension is a significant contributor to premature mortality worldwide [20]. It increases the risk of developing a multitude of illnesses, including myocardial infarction, renal failure,thrombosis, atherosclerosis, and stroke [21]. In Spain, the governmental department of transportation, the DGT (Dirección General de Tráfico), asserts that 76% of the population is affected heavily by road noise. For 30% of the Spanish population, this phenomenon is perceived as a significant source of annoyance. Nearly 9 million individuals are in this situation, as cited in the DGT report [22].

One further negative consequence of the noise generated by vehicles on roads is the impact this has on local fauna. This noise causes displacement, reduces available habitat and has an adverse effect on reproduction. All of this is due to

a loss of hearing, which affects communication between animals, and therefore, their behavior [23]. Furthermore, authors of [24], -Forman and Alexander (1998)- assure that the consequences of noise, environmental and visual pollution caused by vehicular traffic result in greater losses to wildlife than road traffic collisions themselves.

One of the fundamental steps in the engineering method is the collection of measurements, which serves as the foundation for the subsequent analysis and interpretation of results. The British physicist and engineer Lord Kelvin assured that "when you can measure what you are speaking about, and express it in numbers, you know something about it; but when you cannot measure it, when you cannot express it in numbers, your knowledge is of a meagre and unsatisfactory kind: it may be the beginning of knowledge, but you have scarcely, in your thoughts, advanced to the stage of science" (Thomson 1889: 73). Quote from [25].

This project, OBSIDAS: On-Board Sound Intensity Data Acquisition System (from now on it will be called just by its acronym: OBSIDAS), is focused on the measurement and quantification of the tire-pavement noise made by a vehicle. This noise is the most significant contributor to the overall traffic noise environment [26].

Consequently, the measurement of road noise is of paramount importance in order to identify the areas that require the most improvement. Furthermore, when a change or improvement to the pavement is to be implemented, OBSIDAS is an ideal instrument for determining whether the upgrade has resulted in a reduction in noise.

OBSIDAS, is a software program developed in LabVIEW that is designed to acquire the noise signal, compute the necessary sound level measurements, and graphically display the result in order to perform the AASHTO (American Association of State Highway and Transportation Officials) [27] standard T360 [28].

In essence, the OBSI method entails the implementation of two probes (leading and trailing), each equipped with two microphones (in and out), resulting in a total of four microphones attached to a vehicle's tire. These microphones are used to record the sound intensity over a specific time and velocity. The recorded sound intensity is then processed to yield measurements that can be used to quantify pavement noise. The tire must adhere to the ASTM F2493 standard [29] and the computation and setup are described in the AASHTO T360 standard [28]. Figure 1.1 depicts the overall setup to perform the OBSI method.

It is worth noting that the entire hardware setup was provided by the *John A. Volpe National Transportation Systems Center*, which is part of the *United States Department of Transportation* and is located in Cambridge, Massachusetts.

The AASHTO T360 is the standard of test for measuring of tire-pavement noise using the OBSI (On-Board Sound Intensity) method. This is the most widely spread method in the sector and its procedure and the reported data needed to

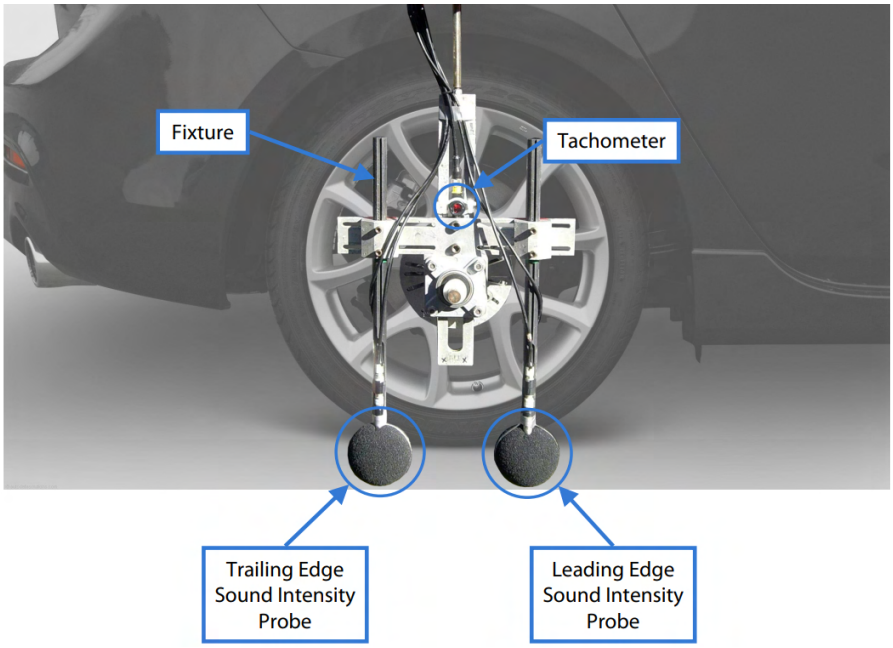


Figure 1.1: Setup to perform the OBSI method, Source: [30]

adjust the measurement to the standard are precisely detailed in [28]. In the section 2.5.

In order to provide an initial overview of the computational requirements, T360 [28] states that in order to quantify the level of sound, and OBSI-qualifying software must:

1. Acquire the sound intensity
2. Compute the sound intensity into sound pressure level (SPL) utilizing a reference value for the absolute threshold of hearing (ATH): $20\mu Pa$
3. Evaluate this SPL across one-third octave bands
4. Final results are reported as overall A-weighted SPL, and as A-weighted one-third octave band levels

OBSIDAS not only adheres to all T360 procedures, but also presents the results in graphs and schemes in a highly intuitive user interface. Furthermore, it performs additional sound quality and level measurements, as well as frequency analysis, thereby offering the user more information about the OBSI test.

It is important to note that this test method standard describes the procedure, requirements, and validation of both hardware devices and software computations.

This project will focus on the software scope, as OBSIDAS is emphasized in the signal acquisition, processing, graphical representation, and file creation of the results, rather than the physical attributes, such as the tire inspection, pavement surface, or weather limitations prior to the test.

The phases of this project can be delineated as follows: first, the acquisition of the signal through the pertinent LabVIEW drivers; second, the computation of all T360 processes, and additional processes, which will be explained in the following sections; third, all the computed data, and results will be graphically exposed to the user via UI; fourth, the user may select the option to download the raw data acquired by the microphone and the processed computation into a CSV-type file.

The following sections provide a comprehensive overview of the scope and development of the project. The paper will proceed to examine the requisite hardware for acquiring sound intensity, the LabVIEW drivers that bridge hardware and software, and the software developed (OBSIDAS). Illustrating all the signal processing and computation required to adhere to the standard T360.

In light of the adverse health and welfare consequences that noise generated by vehicles has on humans and wildlife, this phenomenon is regarded as a contemporary and global issue that requires comprehensive investigation.

The current technologies that address this significant issue are scarce and outdated (As will be explained in detail in 4). For instance, SoftdB's technology is designed to operate within a Windows 7 environment, which Microsoft has ceased to provide updates for since 2012.

The principal motivation of the project has been to develop the requisite software to implement an updated OBSI method that adheres to the AASHTO T360 standard. This has involved the creation of a program.

The software must be accessible and downloadable on any current machine. Its purpose is to quantify and better understand the problem of automotive noise on the roads. It must also obtain the necessary data and discover new technologies to reduce the noise level on the roads.

Chapter 2

Theoretical Concepts

This section presents the primary theoretical concepts required to provide a comprehensive overview of the project. The topics that will be explained are: one-third octave band, A-weighting, the distinction between sound intensity and sound pressure, and microphone sensitivity and calibration.

2.1 One-third Octave Band

Throughout the project, the frequency spectrum is divided into one-third octave bands. This is done to achieve some improvements in the way noise is handled and processed. A 1/1 octave band is a range of frequencies where the highest frequency is twice the lowest frequency. To illustrate this, an octave band starting at 1,000 Hz would cover the range from 1,000 Hz to 2,000 Hz. According to the ANSI S1.11 standard [31], the different 1/1 octave bands are those described in Figure 2.1.

Conversely, a 1/3 octave band is a narrower band that divides each octave into three equal parts. According to the ANSI S1.11 standard [31], the different 1/3 octave bands are those described in Figure 2.1.

2.1.1 Benefits and Applications of One-Third Octave Band Analysis

The reasons for the utilization of one-third octave band analysis in various fields, most notably in acoustics and related fields such as sound engineering, can be summarized as follows:

- A more detailed analysis of the frequency spectrum is achieved when compared to broader octave bands. Consequently, a finer resolution is performed,

Octave Bands			1/3 Octave Bands		
Lower Band Limit (Hz)	Center Frequency(Hz)	Upper Band Limit (Hz)	Lower Band Limit (Hz)	Center Frequency(Hz)	Upper Band Limit(Hz)
			14.1	16	17.8
11	16	22	17.8	20	22.4
			22.4	25	28.2
			28.2	31.5	35.5
22	31.5	44	35.5	40	44.7
			44.7	50	56.2
			56.2	63	70.8
44	63	88	70.8	80	89.1
			89.1	100	112
			112	125	141
88	125	177	141	160	178
			178	200	224
			224	250	282
177	250	355	282	315	355
			355	400	447
			447	500	562
355	500	710	562	630	708
			708	800	891
			891	1000	1122
710	1000	1420	1122	1250	1413
			1413	1600	1778
			1778	2000	2239
1420	2000	2840	2239	2500	2818
			2818	3150	3548
			3548	4000	4467
2840	4000	5680	4467	5000	5623
			5623	6300	7079
			7079	8000	8913
5680	8000	11360	8913	10000	11220
			11220	12500	14130
11360	16000	22720	14130	16000	17780
			17780	20000	22390

Figure 2.1: 1/1 and 1/3 Octave Bands Frequencies Table. Source: [32]

which is useful when the sound analysis is complex.

- This analytical approach is widely employed within the sector. Standards set by the International Electrotechnical Commission (IEC) and the American National Standards Institute (ANSI) utilise this method, thereby ensuring consistency and comparability in measurements across different studies and programmes.

- This analysis is more consistent with the non-linearity of the human ear. The method of dividing the frequency spectrum into one-third octave bands closely approximates the non-linearity of the human ear, which functions in a logarithmic manner. These octave bands have a fixed ratio between the

lower and higher bands throughout the spectrum, which aligns with the way the human ear perceives noises. This method represents a highly consistent and efficient approach to understanding the human ear's perception of sound. It is a widely utilized method in human ear-related applications because it reflects the natural way in which humans perceive sound [33].

2.1.2 Calculation of One-Third Octave Bands

In order to calculate the one-third octave bands, a reference has to be established. Along the audible spectrum (from 20 to 20000 Hz) 31 one-third octave bands could be created. The reference is the band with the center frequency in 1000 Hz, in accordance with the agreement, the number 19 along the octave bands. Therefore and having in mind this reference, all lower center frequencies for one-third octave bands can be obtained using the following equation 2.1 (source: [32]).

$$f_{n-1} = \frac{f_n}{2^{1/3}} \tag{2.1}$$

Where:

- f_{n-1} is the lower center frequency for one-third octave band;
- f_n is the reference center frequency for one-third octave band;

On the other hand, all higher center frequencies for one-third octave bands can be obtained using the following equation 2.2 (source: [32]).

$$f_{n+1} = f_n \cdot 2^{1/3} \tag{2.2}$$

Where:

- f_{n+1} is the higher center frequency for one-third octave band;
- f_n is the reference center frequency for one-third octave band;

Furthermore, for each center frequency, all lower limit frequencies for one-third octave bands can be obtained using the following equation 2.3 (source: [32]).

$$f_n^{low} = \frac{f_n}{2^{1/6}} \tag{2.3}$$

Where:

- f_n^{low} is the lower limit frequency for one-third octave band;
- f_n is the reference center frequency for one-third octave band;

On the other hand, for each center frequency, all higher limit frequencies for one-third octave bands can be obtained using the following equation 2.4 (source: [32]).

$$f_n^{high} = f_n \cdot 2^{1/6} \tag{2.4}$$

Where:

- f_n^{high} is the higher limit frequency for one-third octave band;
- f_n is the reference center frequency for one-third octave band;

2.2 A-weighting

This section presents the primary theoretical concepts necessary to provide a comprehensive overview of A-weighting filters. This type of filter adapts the sound captured by the microphones to the human ear's perception of sound. It is well known that the human ear does not perceive different frequencies in the same way; for example, the ear is more sensitive to frequencies between 1 and 4000 Hz than to lower or higher frequencies. This means that a higher sound pressure level outside of this range could be perceived in a similar way to a lower sound pressure level in this frequency range. [34]

Therefore, the A-weighting filter is designed to attenuate frequencies below about 1000 Hz and above 4000 Hz, with a peak sensitivity around 1000 to 4000 Hz, while also attenuating frequencies above 8000 Hz [35]. Figure 2.2 represents the gain curve of the A-weight filter. The AASHTO T360 standard describes that the A-weighing shall conform to ANSI S1.42 [36], and the units are dbA or dB(A).

It is widely used because this type of filter approximates an equal-loudness contour, meaning that a constant loudness is perceived across different frequencies.

2.3 Sound Intensity, Sound Pressure, and their Levels

In this section, an overview of the theoretical concepts regarding sound intensity, sound pressure, and their levels, will be examined.

2.3.1 The Role of Sound Pressure in Auditory Perception

It is important to note at the outset that the human ear is not capable of detecting sound intensity or sound power. Instead, it is able to capture sound pressure. As

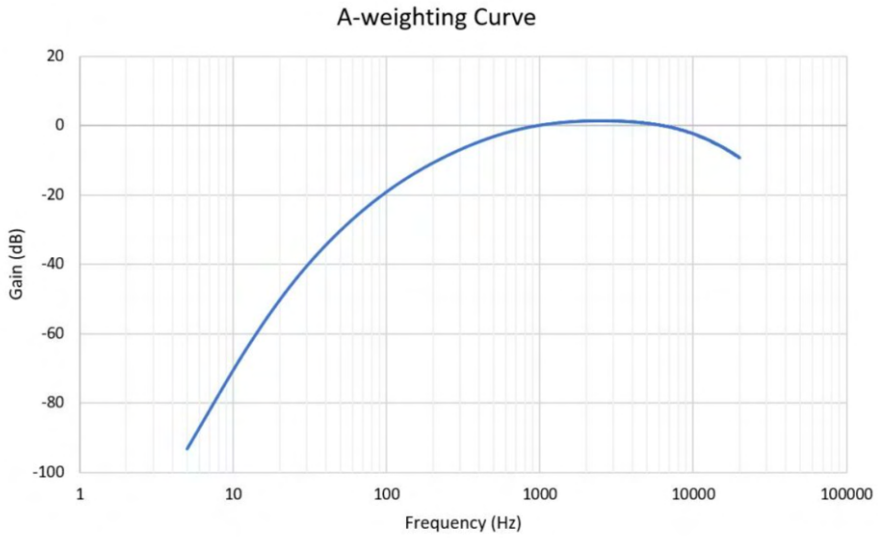


Figure 2.2: A-weighting Table. Source: [35]

expressed in [37]:

> *The tympanic membranes are moved by sound pressure as a sound field quantity. That is the important effect of the excess sound pressure. It is not an energy receiver. When listening to music, forget the sound intensity as energy quantity. The perceived sound consists of periodic acoustic pressure vibrations (sound pressure) which are superimposed on the surrounding static air pressure of 101325 Pascal (mean atmospheric pressure). The sound is the change of sound pressure p, which is measured in pascal.* $1Pa \equiv N/m^2 \equiv 1J/m^3 \equiv 1kg/(m \cdot s^2)$ *Usually p is the RMS value.*

2.3.2 Understanding the Causal and Effectual Relationship Between Sound Pressure and Sound Power

The relationship between sound power and sound pressure is a cause-and-effect relationship. Sound pressure is the effect, which was caused by a sound power (hair dryer, car engine, or alarm clock). This relationship is analogous to the foundation of the temperature: the temperature is the effect, which was caused by an electrical power.

One of the key aspects is distance. In both cases, knowing the effect (temperature or sound pressure) is not sufficient to know how cold-hot a room is or how loud-quiet a sound is. In order to achieve a meaningful quantification of temper-

ature or noise detection, distance is a crucial factor. The further the electrical or sound power is from the source, the lower the temperature or sound will be. However, in this project, the distance is fixed by the standard, which is outlined in section 2.5, so this information will be incorporated into the study. Consequently, this measurement will not be explicitly included; however, it is important to note that distance is always necessary for the purpose of establishing a reference.

2.3.3 Sound Intensity, Sound Intensity Level, Sound Pressure, and Sound Pressure Level Definitions

Sound Intensity

The sound intensity (*SI* or I), as expressed in [38]:

> *Is defined as the power carried by sound waves per unit area in a direction perpendicular to that area.*

The unit of sound intensity is watt per square meter, W/m^2. Sound intensity is a way to measure the sound energy quantity [37]. The auditory threshold is $I_0 = 10^{-12} W/m^2$. It is used as the reference in order to compute the sound intensity level (*SIL*).

Sound Intensity Level

The sound intensity level (*SIL* is the logarithmic expression of the sound intensity when compared to a reference, usually this reference is the auditory threshold $I_0 = 10^{-12} W/m^2$. The unit is decibels (dB), and if the level is A-weighted beforehand the unit is represented as dBA or db(A). This threshold of hearing equals to SIL of 0dB at 1kHz frequency.

In the equation 2.5 (source: [39]) is shown how to obtain the sound intensity level.

$$\mathrm{SIL}\,[\mathrm{dB}] = 10 \log_{10} \left(\frac{I\,[\mathrm{W/m^2}]}{I_0\,[\mathrm{W/m^2}]} \right) \tag{2.5}$$

Where:

- I is the sound intensity
- I_0 is the reference sound intensity, $10^{-12} W/m^2$

Sound Pressure

The sound pressure (SP or p), as expressed in [40]:

> *Is the local pressure deviation from the ambient (average or equilibrium) atmospheric pressure, caused by a sound wave.*

The unit of sound intensity is the pascal, Pa. Sound pressure is a way to measure the sound field quantity [37]. The auditory threshold is $p_0 = 20\mu Pa = 2 \cdot 10^{-5} Pa$. It is used as the reference in order to compute the sound pressure level (SPL). The root mean square (RMS) method is the most commonly employed approach for measuring sound pressure. This method is based on the premise that sound waves naturally fluctuate between positive and negative peaks. By averaging these fluctuations, the RMS method provides a steady number that represents the average effective pressure of these oscillations. Additionally, RMS measurements are more closely aligned with the human perception of sound loudness, making them more relevant to practical applications.

Sound Pressure Level

The sound pressure level (SPL is the logarithmic expression of the sound pressure when compared to a reference, usually this reference is the auditory threshold $p_0 = 20\mu Pa = 2 \cdot 10^{-5} Pa$. The unit of measurement is decibels (dB), and if the level is A-weighted beforehand, the unit is represented as dBA or dB(A). This threshold of hearing is equivalent to the sound intensity level (SIL) of 0 dB at a frequency of 1 kHz.

In the equation 2.6 (source: [39]) is shown how to obtain the sound intensity level.

$$\mathrm{SPL}\,[\mathrm{dB}] = 10\log_{10}\left(\frac{p_{\mathrm{rms}}^2\,[\mathrm{Pa}^2]}{p_0^2\,[\mathrm{Pa}^2]}\right) = 20\log_{10}\left(\frac{p_{\mathrm{rms}}\,[\mathrm{Pa}]}{p_0\,[\mathrm{Pa}]}\right) \tag{2.6}$$

Where:

- p_{rms} is the root mean square pressure of the sound
- p_0 is the reference sound pressure, $20\mu Pa$

Relation Between SPL and SIL

Both SPL and SIL measure different sound aspects, yet they are mathematically related through the acoustic impedance. The sound intensity I and the sound pressure p are related to the acoustic impedance Z_o as expressed in the equation 2.7 (source: [39]).

$$I = \frac{p^2}{Z_o} \rightarrow p = \sqrt{I \cdot Z_o} \tag{2.7}$$

Where:

- p is the root mean square pressure of the sound
- I is the sound intensity
- Z_o is the acoustic impedance with units $N \cdot s/m^3$

It is noteworthy that when an acoustic impedance of $400N \cdot s/m^3$, which is an approximation of the acoustic impedance of the air, is employed, both the reference of the sound intensity and the reference of the sound pressure match, as represented in equation 2.8 (own elaboration).

$$I = \frac{p^2}{Z_o} \rightarrow 10^{-12}[W/m^2] = \frac{(20 \cdot 10^{-6}[Pa])^2}{400[N \cdot s/m^3]} \tag{2.8}$$

Where:

- p is the sound pressure
- I is the sound intensity
- Z_o is the acoustic impedance with units $N \cdot s/m^3$

Therefore, when using $Z_0 = 400N \cdot s/m^3$ the calculations for SPL and SIL give the same numerical value in decibels because the conversion factor between pressure and intensity aligns perfectly with the chosen reference values. Consequently, the sound pressure level in the air is equivalent to the sound intensity level. Consequently, throughout the project, there will be no distinction between SPL or SIL, since both, due to the assumption of air, are assumed to be the same value. As illustrated in equation 2.9 (own elaboration).

$$\text{SPL} = 20\log_{10}\left(\frac{p_{\text{rms}}}{p_0}\right) = 10\log_{10}\left(\frac{p_{\text{rms}}^2}{p_0^2}\right) = 10\log_{10}\left(\frac{I}{I_0}\right) = \text{SIL} \tag{2.9}$$

Where:

- p_{rms} is the root mean square pressure of the sound
- p_0 is the reference sound pressure, $20\mu Pa$

- I is the sound intensity
- I_0 is the reference sound intensity, $10^{-12} W/m^2$
- Z_o is the acoustic impedance with units $N \cdot s/m^3$

In order to gain insight into the dBA order of magnitude, figure 2.3 presents typical dBA levels and their corresponding sound power sources

190 dBA	Heavy weapons, 10 m behind the weapon (greatest level)
180 dBA	Toy pistol fired close to ear (greatest level)
170 dBA	Slap on the ear, fire cracker explodes on shoulder, small arms at a distance of 50 cm (greatest level)
160 dBA	Hammer stroke on brass tubing or steel plate at 1 m distance, airbag deployment very close at a distance of 30 cm (greatest level)
150 dBA	Hammer stroke in a smithy at 5 m distance (greatest level)
130 dBA	Loud hand clapping at 1 m distance (greatest level)
120 dBA	Whistle at 1 m distance, test run of a jet at 15 m distance
	Threshold of pain, above this fast-acting hearing damage in short action is possible
115 dBA	Take-off sound of planes at 10 m distance
110 dBA	Siren *) at 10 m distance, frequent sound level in discotheques and close to loudspeakers at rock concerts, violin close to the ear of an orchestra musicians (greatest level)
105 dBA	Chain saw at 1 m distance, banging car door at 1 m distance (greatest level), racing car at 40 m distance, possible level with music head phones
100 dBA	Frequent level with music via head phones, jack hammer at 10 m distance
95 dBA	Loud crying, hand circular saw at 1 m distance
90 dBA	Angle grinder outside at 1 m distance
	Over a duration of 40 hours a week hearing damage is possible
85 dBA	2-stroke chain-saw at 10 m distance, loud WC flush at 1 m distance
80 dBA	Very loud traffic noise of passing lorries at 7.5 m distance, high traffic on an expressway at 25 m distance
75 dBA	Passing car at 7.5 m distance, un-silenced wood shredder at 10 m distance
70 dBA	Level close to a main road by day, quiet hair dryer at 1 m distance to ear
65 dBA	**Bad risk of heart circulation disease at constant impact is possible**
60 dBA	Noisy lawn mower at 10 m distance
55 dBA	Low volume of radio or TV at 1 m distance, noisy vacuum cleaner at 10 m distance
50 dBA	Refrigerator at 1 m distance, bird twitter outside at 15 m distance
45 dBA	Noise of normal living; talking, or radio in the background
40 dBA	**Distraction when learning or concentration is possible**
35 dBA	Very quiet room fan at low speed at 1 m distance
25 dBA	Sound of breathing at 1 m distance
0 dB	Auditory threshold

Figure 2.3: Typical dBA Levels and Their Corresponding Sound Power Sources. Source: [37]

2.3.4 Conversion Between Sound Intensity, Sound Intensity Level, Sound Pressure, and Sound Pressure Level

In this section the formulas needed in order to convert sound intensity to sound pressure are exposed. To achieve the conversion between SI and SP, the procedure will be as follows.

Convert Sound Intensity Into Sound Pressure

If SP is needed from SI:

- Convert sound intensity into sound intensity level
- As explained beforehand: SIL = SPL when using $Z_0 = 400N \cdot s/m^3$
- Convert sound pressure level into sound pressure

Therefore, the equation 2.10 (source: [41]) may be used in order to convert from SI to SP:

$$p = 2 \cdot 10^{-5} \cdot 10^{\frac{10 \cdot \log\left(\frac{I}{10^{-12}}\right)}{20}} \tag{2.10}$$

Where:

- I is the sound intensity
- p is the sound pressure

Convert Sound Pressure Into Sound Intensity

If SI is needed from SP:

- Convert SP into SPL
- As explained beforehand: SPL = SIL when using $Z_0 = 400N \cdot s/m^3$
- Convert SIL into SI

Therefore, the equation 2.11 (source: [41]) may be used in order to convert from SP to SI:

$$i = 10^{-12} \cdot 10^{\frac{20 \cdot \log\left(\frac{p}{10^{-5}}\right)}{10}} \tag{2.11}$$

Where:

- I is the sound intensity
- p is the sound pressure

Using the Acoustic Impedance

There is another method to achieve the conversion between SI and SP and its using the acoustic impedance formula seen in equation 2.7, with this method the levels will not be computed. Knowing that the acoustic impedance that will be used throughout the entirety of the project is $400N \cdot s/m^3$ the equation 2.12 (source: [41]) may be used in order to get the sound pressure from the sound intensity and equation 2.13 (source: [41]) may be used in order to get the sound intensity from the sound pressure.

An alternative approach to the conversion between SI and SP is available, which utilizes the acoustic impedance formula presented in equation 2.7. This method does not involve the computation of levels. Given that the acoustic impedance to be employed throughout the entirety of the project is $400N \cdot s/m^3$, the equation 2.12 (source: [41]) may be utilized to derive the sound pressure from the sound intensity. Conversely, equation 2.13 (source: [41]) may be employed to obtain the sound intensity from the sound pressure.

$$p = \sqrt{I \cdot Z} = \sqrt{I \cdot 400} \tag{2.12}$$

$$I = \frac{p^2}{Z} = \frac{p^2}{400} \tag{2.13}$$

Where:

- I is the sound intensity
- p is the sound pressure
- Z is the acoustic impedance

2.4 Microphone Calibration

This section will present the principal concepts regarding the calibration of microphones prior to setup.

2.4.1 Calibration Procedure

It is of paramount importance to calibrate a microphone in order to utilize it successfully. Otherwise, if the calibration is not performed, the units and magnitudes will not correspond to reality. Therefore, the calibration procedure relies on the calibrator, which is explained in section 3.2.4. This calibrator is used to project a known signal into the microphone, and then the microphone's output values are compared to the known input calibrator signal. If the two signals are equal, the microphone would the be calibrated. On the other hand, if the two signals differ, the microphone's sensitivity, in other words, the transfer factor, must be adjusted in order to match the calibrator signal.

The calibrator signal is designated as the reference signal and is known to have a specific frequency and sound pressure level. These values are used to compare the signal captured by the microphone. The reference value for the sound pressure level is typically 94 dBSPL, while the signal frequency is 1 kHz. This 94 dB SPL is established as an international standard, with one Pascal (Pa) equaling 94 dB SPL, as cited in [42]. It is notable that the calibrator employs the decibel sound pressure level (dBSPL) unit. In this project, the decibel on the A-weighted scale (dBA) will be utilized. However, as 1 kHz is utilized, the A-weight filter exhibits a response of 0 dB at 1 kHz. Consequently, the filter does not attenuate or amplify the signal in that frequency range. Therefore, in this instance, 94 dBSPL is equivalent to 94 dBA.

2.4.2 Sensitivity and Transfer Factor

The primary objective of microphones is to convert sound pressure captured from the ambient noise into a digital electric signal in order to facilitate the study, utilization, and measurement of required levels. The measurement that relates the sound pressure (Pa) and the electric signal generated (V) is the sensitivity and the transfer factor. Both expressions represent the same relation, Pa-V. However, the sensitivity is expressed in decibels with a reference of 1 V/Pa, while the transfer factor is expressed as V/Pa. The transfer factor is the direct relation between these two units.

The sensitivity must be expressed in negative decibels (dB), as expressed in [42]:

> *Because all microphones produce a voltage level which is less than 1 volt ≡ 0 dB for the reference sound pressure level of 94 dB or 74 dB.*

In order to transition from one to the other, equation 2.14 (source: [42]) illustrates the calculations required to derive the transfer factor from a given sensitivity. Conversely, equation 2.15 (source: [42]) may be employed to obtain the sensitivity from a given transfer factor.

$$\text{transfer factor} = 10^{\text{sensitivity}/20} \tag{2.14}$$

$$\text{sensitivity} = 20 \cdot \log\left(\text{transfer factor}\right) \tag{2.15}$$

2.5 AASHTO T 360-16

Throughout this paper, the AASHTO T360 standard will be referenced and discussed in detail. This section will provide a comprehensive description of the standard. The reference used for the standard is cited in [28], which is the official document from the *American Association of State Highway and Transportation Officials (AASHTO)*, the title is: *Standard Method of Test for Measurement of Tire/Pavement Noise Using the On-Board Sound Intensity (OBSI) Method*. The AASHTO designation is *T 360-16 (2020)*. It was technically revised in 2016, reviewed but not updated in 2020 and editorially revised in 2021. The technical subcommitte is 5a, pavement measurement. The entirety of the information and details has been obtained from this reference.

It is important to note that this test method standard describes the procedure, requirements, and validation of both hardware devices and software computations. This section will focus on the software scope, as OBSIDAS is emphasized in the signal acquisition, processing, graphical representation, and file creation of the results, rather than the physical attributes, such as the tire inspection, pavement surface, or weather limitations prior to the test. Consequently, this project, and particularly this section, will analyze the software and processing signal requirements and procedure.

The significance and use of this test method to quantify the noise associated with a road or pavement in order to facilitate comparative analysis. Furthermore, when a change or improvement to the pavement is to be implemented, this standard is an ideal instrument for determining whether the upgrade has resulted in a reduction in noise.

2.5.1 Terminology

The standard establishes the technical terminology that will be used throughout the entirety of the document. A greater number of concepts are delineated in the standard, yet the principal concepts that require explanation are as follows:

- **Coherence;** as quoted from [28], *"mathematically, it is the magnitude of the cross-spectrum between two signals squared divided by the product of the auto spectrum of both signals"*. This metric quantifies the linear dependency

between two signals. If the coherence is close to or equal to zero, it can be concluded that both signals are not linearly dependent. Conversely, if the coherence is close to or equal to one, it can be inferred that the signals are highly linearly dependent.

- **Energy averaging of levels;** in order to calculate this average, the levels should be transformed into energy quantities (normally intensity or pressure units), average these, and then converting back to level units (normally dB). To achieve the average of N number of L_i levels, the following equation 2.16 may be used (source: [28]):

$$L_{E-avg} = 10 \cdot \log_{10} \left(\frac{1}{N} \sum_{i=1}^{N} \left(10^{L_i/10} \right) \right) \tag{2.16}$$

- **Measurement period;** is the time elapsed for the vehicle to cross the test road.

- **Standard Reference Test Tire (SRTT);** tire used to perform the OBSI method, as it adheres to the standard ASTM F2493 [29].

- **Overall OBSI level;** it is the final OBSI measurement that will expose the quantification of the tire-pavement noise. This measure is used to compare different pavements. The goal of the OBSI method is to get this indicator. It is defined as: (quoted from [28]) *"the energy sum of the A-weighted sound intensity within the one-third octave bands ranging from 400Hz to 5000Hz"*. The equation 2.17 may be used (source: [28]):

$$OBSI\ level = 10 \cdot \log_{10} \left(\frac{1}{N} \sum_{400}^{5000} \left(10^{L_i/10} \right) \right) \tag{2.17}$$

  Where L_i is the A-weighted intensity level in the one-third octave band with center frequency i.

- **Pressure Intensity index (PI index);** this measurement is defined as the arithmetic average of the sound pressure level (SPL) in a given one-third octave band, minus the sound intensity level in the same band. The standard states that the unit should be dB rather than dbA due to the fact that it represents a difference in level.

- **Reference temperature;** is defined as the ambient temperature of 20 degrees Celsius.

- **Sound intensity;** quoted from [28]:

  > *Average rate of sound energy transmitted n a specified direction at a point through a unit area normal to this direction at the point. Units are watt per square meter (W/m^2). As such, it is a vector quantity with magnitude and direction. The direction can be associated with the direction of sound propagation or the direction of the orientation of a probe for measuring sound intensity. (Reference ANSI S1.1, Acoustical Terminology.)*

- **Sound Intensity Level (SIL);** is defined as ten times the logarithm to the base 10 of the ratio of the sound intensity of a given sound in a stated direction to the reference sound intensity of 1 pW/m^2. For a detailed illustration of the computation of SIL, refer to equation 2.18 (source: own elaboration). The units are decibels.

$$SIL[dB] = 10 \cdot \log_{10}\left(\frac{sound\ intensity[W/m^2]}{1 \cdot 10^{-12}[W/m^2]}\right) \tag{2.18}$$

- **Sound Pressure Level (SPL);** is defined as ten times the logarithm to the base 10 of the ratio of the time mean square pressure of a given sound in a stated frequency band, to the reference sound pressure of 20 μPa. For a detailed illustration of the computation of SPL, refer to equation 2.19 (source: own elaboration). The units are decibels.

$$SPL\,[\mathrm{dB}] = 10\log_{10}\left(\frac{p^2_{\mathrm{rms}}\,[\mathrm{Pa}^2]}{p^2_0\,[\mathrm{Pa}^2]}\right) = 20\log_{10}\left(\frac{p_{\mathrm{rms}}\,[\mathrm{Pa}]}{p_0\,[\mathrm{Pa}]}\right) \tag{2.19}$$

- **Standard Reference Test Tire;** 16 inches (P225/60R16) standard tire as defined in ASTM F2493 [29].

- **Test Section;** the longitudinal dimension of the pavement upon which the OBSI method is performed. This dimension is 134 ±3 meters (440 ±10 feet) in length.

2.5.2 Standard Test Speed and Averaging Time

This section will present an overview of the test requirements and operating conditions. The standard test speed is 97 km/h (60 mph), unless circumstances warrant a different speed. In such cases, the highest speed from the following list should be used.

- 72 km/h (45 mph)

- 56 km/h (35 mph)
- 40 km/h (25 mph)

Test speed during the trial should be constant, however a variation of ±1.6 km/h (±1 mph) is considered valid. The length of the test road is the one depicted in section 2.5.1 (134 meters), therefore given the speed test as 97 km/h (60 mph), the time test is 5.0 ±0.1 seconds. When a slower velocity is used, an increment on the test time should be performed, following:

- 72 km/h (45 mph) corresponds to 6.7 ±0.2 seconds
- 56 km/h (35 mph) corresponds to 8.6 ±0.2 seconds
- 40 km/h (25 mph) corresponds to 12 ±0.3 seconds

2.5.3 Requirements for a Valid Test Run

This section will define the conditions that must be met for the final measurements to be considered a valid test run.

Pressure Intensity Index

In order to consider the pressure intensity index valid, it must conformance with the maximum and minimum levels depicted in figure 2.4.

Coherence

As quoted from the official standard documentation [28]

> *The coherence of sound pressure between the two microphones of each sound intensity probe shall be equal to or greater than 0.8(rounded to the nearest tenth) for each one-third octave band with a center frequency between 400 and 4000 Hz. (Note there is no coherence requirement for the 5000 Hz one-third octave band.) Any test run in which these criteria are not satisfied shall be considered invalid and not used in any averages of the test run.*

One-Third Octave Band Center Frequency (Hz)	Minimum *PI* Index (dB)	Maximum *PI* Index (dB)
400	–1.0	5.0
500	–1.0	4.5
630	–1.0	3.5
800	–1.0	2.5
1000	–1.0	2.5
1250	–1.0	3.0
1600	–1.0	3.0
2000	–1.0	3.0
2500	–1.0	3.5
3150	–1.0	4.0
4000	–1.0	4.5
5000	–1.0	5.0

Figure 2.4: Pressure Intensity Requirements for Each One-Third Octave Band. Source: [28]

Normalization of Sound Intensity Levels to a Reference Temperature

An average 20ºC reference temperature should be assumed. But changes in this tmperature will be likely to happen. Therefore, a linear adjustment of 0.072 dB/ºC shall be implemented in order to normalize the measured sound intensity levels to the reference temperature of 20ºC. To achieve this, the following equation 2.20 shall be used (source: [28]):

$$IL_{Normalized}[dBA] = IL_{Measured}[dBA] + 0.072 \cdot (AirTempC - 20C) \quad (2.20)$$

Where:

- $IL_{Measured}$ refers to the sound intensity level measured by the system (original SIL).
- $AirTemp$ refers to the ambient air temperature at the time of the trial.
- $IL_{Normalized}$ refers to the final sound intensity level normalized to the reference temperature.

It is worth noting that the standard specifies that the test be limited to an ambient air temperature range of 4 to 38 ºC.

2.5.4 Reported Data

This section will provide an overview of the data that should be included in the test report. As a general guideline, AASHTO has established that all levels should be rounded to the nearest 0.1 dB. Furthermore, the frequency spectra of the sound intensity, sound pressure, and sound pressure level will be A-weighted one-third octave bands with center frequencies between 400 and 5000 Hz. The octave band filters must be compliant to the IEC 61260 [43] [44] [45] or ANSI S1.11 [31] standards.

It is noteworthy that the standard provides clarification regarding the averaging procedure to be performed with the various microphones in order to obtain the final value of the test trial. In order to obtain the final OBSI values, each A-weighted one-third octave band measurement will be arithmetically averaged, as explained in section 2.5.1. Once the final probe value (leading or trailing) has been computed, the across probe value (final test value) is obtained by calculating an energy average (as explained in section 2.5.1) of the leading and trailing values. Therefore, between the same probe microphone measures, an arithmetic average must be performed, and between different probe measures, an energy average must be calculated.

Required Data for Inclusion in the Test Report

Quoted from [28]:

- Roadway route number, designation, and direction;
- Time and date of measurement;
- Air temperatures taken during the measurement period;
- Organization and operators responsible for the measurement;
- Organization and operators responsible for the measurement;
- Pavement lane in which the measurements were taken;
- GPS coordinates or official agency distance mark and identifying the beginning od the test sections; accuracy should be ±3m (10 feet);
- Test speed, including a clear designation if the speed selected for testing differs from the standard tst speed of 97 km/h (60 mph);
- Most recent measurement result of tire hardness and the date it was measured; and

- Average (of all valid test runs) overall A-weighted sound intensity levels (SIL) in one-third octave bands with center frequencies from 400 to 5000 Hz (OBSI level). Both measured and normalized levels shall be reported.

It is noteworthy that the last needed data, the overall sound intensity levels is the measurement that the program OBSIDAS will obtain. The rest measurements are physical attributes, such as the tire inspection, pavement surface, or weather related, which OBSIDAS does not consider.

Optional Data for Inclusion in the Test Report

The standard exposes that there are certain measurements which should be recorded but may be optionally reported. The following list considers this optional data that the OBSIDAS program will report (quoted from [28]).

- Overall sound intensity levels for each valid test run. Measured and normalized levels shall both be reportes.
- Sound intensity levels in A-weighted one-third octave bands with center frequencies from 400 to 5000 Hz for each valid test run. Measured and normalized levels shall both be reported.
- Ranges of the overall sound intensity and A-weighted one third-octave-band levels with center frequencies between 400 and 5000 Hz.
- Pressure intensity index, and coherence check results.

2.5.5 Installation and Hardware Configuration

As previously stated, the OBSIDAS program is not designed to focus on the hardware setup but rather on the signal acquisition and processing. However, this section will provide an overview of the setup that the OBSIDAS is designed to work with. The sections 2.5.5 and 2.5.5 are directly quoted from [28].

Probe Position

The position of the sound intensity probe is defined to be the location of the midpoint between the two microphone diaphragms. The sound intensity probes shall have a fixed position relative to the tire as illustrated in Figure 2.5. The horizontal distance from the probes to the closest point on the tire sidewall is 4 in. (101.6 mm), measured with the tire loaded and in the presence of the bulge in the tire sidewall due to the loading. The vertical distance above ground level is 3 in. (76.2 mm). The spacing between the probes shall be 8.25 in. (209.6 mm),

centered about the hub [4.125 in. (104.8 mm) on each side of hub centerline]. All of these positions shall have a tolerance of ±0.25 in. (±6.3 mm). Figures 2.6, 2.7, and 2.8 show typical arrangements.

Microphone Spacing

The pair of microphones on any sound intensity probe shall be positioned in a side-by-side configuration and separated such that the center-to-center spacing between the two diaphragms is 0.063 ±0.04 in. (16 ±1 mm).

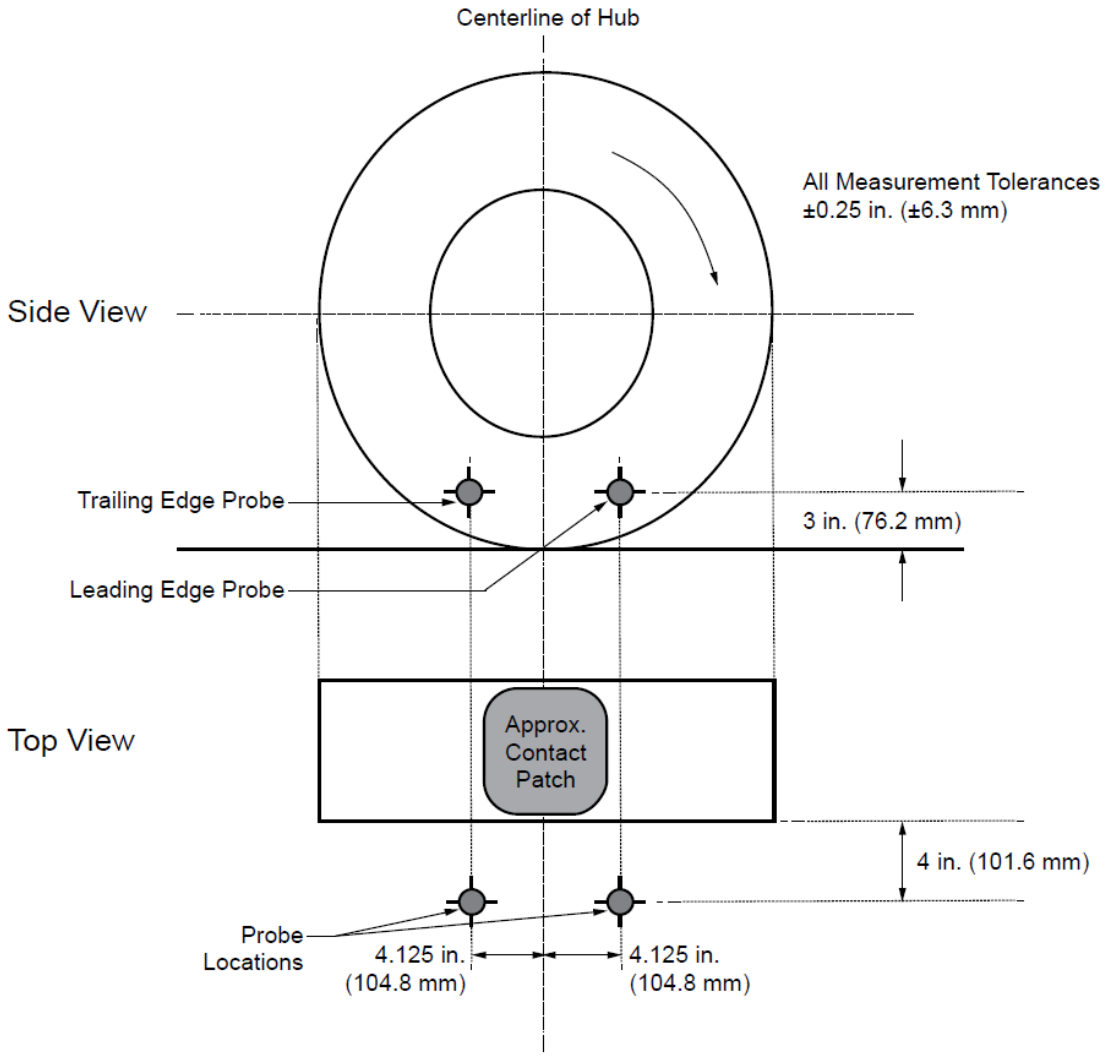


Figure 2.5: Sound Intensity Probe Positions Showing Leading and Trailing Edge Measurements. Source: [28]



Figure 2.6: Dual-Probe Vertical OBSI Setup. Source: [28]



Figure 2.7: Dual-Probe Vertical OBSI Setup (shown without required windscreens). Source: [28]



Figure 2.8: Close-Up of Dual-Probe Vertical OBSI Setup (shown without required windscreens). Source: [28]

Chapter 3

Technologies Involved

In this section the technologies, environment, and tools employed for the successful development of the project will be explained and outlined. Throughout the entire duration of this project the main technology being used has been the visual programming environment LabVIEW, acronym of Laboratory Virtual Instrument Engineering Workbench, which was developed by the American company National Instruments in the mid-1970s. As will be detailed in the following sections, three primary LabVIEW libraries were used: Sound and Vibration Toolkit [30], NI-DAQmx [46], and "File Input/Output" [47].

3.1 Software: LabVIEW

Laboratory Virtual Instrument Engineering Workbench, widely known as LabVIEW is a graphical environment that has been designed with a focus on the development of hardware and software systems, in addition to design of data acquisition, data communication, instrumentation, or signal processing in simulated, real, or embedded systems. With regard to the utilization of the program, there are two different windows: "Block Diagram" and "Front Panel". This first one corresponds to the structure, and logic of the main functionality of the program, while the "Front Panel" represents the graphical user interface. Both windows are connected through the same file, and a change on one would represent a modification on the other. In summary, a LabVIEW programmer must implement the code and logic structure in the"Block Diagram", and arrange the graphic controls created in accordance with the tools generated in the "Block Diagram". The final user of the program will only interact with the "Front Panel". For the sake of comparison, "Bock Diagram" may be considered the integrated development environment (IDE), while the "Front Panel" represents the graphic user interface. The file type utilized and generated by the program is ".vi" (visual instrument).

Therefore, in the subsequent sections of this paper, the designation "VI" will be used to refer to both the "Front Panel" and the "Block Diagram".

Within this project the version utilized was 2024-Q1 (64-bit). The selection of this version was made based on its compatibility with the requisite toolkits. As it will be discussed in the following sections, the primary libraries or toolkits employed are: "Sound and Vibration Toolkit (SVT)", and "NI-DAQmx". It was necessary to ascertain the compatibility of the three software programs (SVT, DAQmx and LabVIEW). Accordingly, version 2024-Q1 met all the requirements of compatibility.

It is important to note the existence of "subVIs". While there are no fundamental differences between subVIs and classic VIs, subVIs can be incorporated into other VIs, allowing a VI to call subVIs. This approach enables the implementation of repetition structures within a VI into a subVI, which can then be called. Moreover, the primary concept underlying this strategy is often referred to as the "Divide and Conquer Algorithm" [48], which essentially implies that a complex problem will be addressed by dividing it into sub-problems of a more manageable nature.

3.1.1 Sound and Vibration Toolkit

As one of the primary functions of the OBSIDAS program is signal processing, the toolkit "Sound and Vibration" provides a wide variety of options for programmers in this domain. This toolkit is a collection of VIs designed to perform sound and vibration analysis, as the author of [30] asserts:

> Use the Sound and Vibration Toolkit to perform the following sound and vibration measurements:
>
> - Scaling a signal to engineering units (EU)
> - Calibrating a measurement channel
> - Applying weighting filters
> - Integrating time-domain signals
> - Performing level measurements
> - Performing swept-sine measurements
> - Performing single-tone measurements
> - Performing limit and mask testing
> - Performing fractional-octave analysis
> - Performing frequency analysis

- Performing transient analysis
- Performing distortion analysis
- Displaying results

Refer to figure 3.1 for an overview image of the different parts and states of a complete acquisition-visualization program. This toolkit has been designed to function in conjunction with other software packages and/or hardware devices, such as acquisition sources and microphones (e.g., DAQmx).

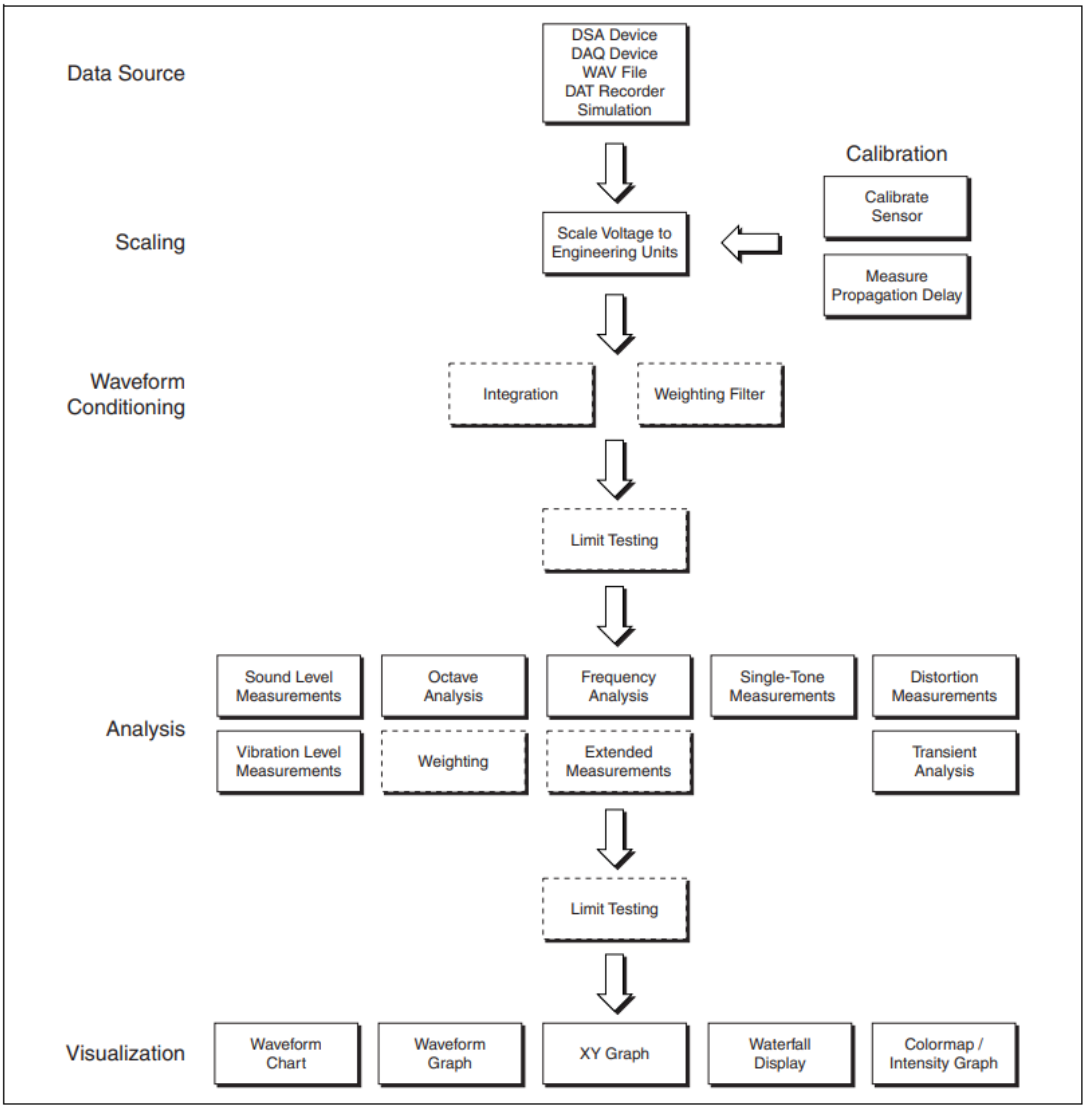


Figure 3.1: Sound and Vibration Toolkit overview, Source: [30]

Refer to figure 3.2 for an overview of the functions palette available in the VI's block diagram. Among the extensive range of mechanisms offered by this library, the most prevalent will be:

- "Scalling and Calibration": as its name implies, these resources assist the programmer to successfully scale the acquired signal to engineering units (EU), which may be expressed in terms of volts, Pascals, Newtons, grams, etc. In addition, they enable the automatic calibration of the selected microphone.
- "Weighting and Filtering": this tool is used to A-weight the acquired signal, as outlined in the AASHTO T 360-16 standard (OBSI method) [28].
- "Sound Level": with these mechanisms equivalent sound level (Leq), running Leq, and exponential Leq can be computed.
- "FFT": this section contains the functionality of studies in the frequency domain, which is the computation of the Fast Fourier Transform, Power Spectral Density, or Cross-Spectrum, in baseband, subset or zoom FFT.
- "Octave": the instruments included in this part focus on developing octave band frequency division. This division is of paramount importance for the purpose of segmenting the spectrum in order to perform a more specific analysis.
- "Sound Quality": this palette offers a range of measurements related to sound quality, including Stationary Loudness, Aures Sharpness, Aures Roughness, or Fluctuation Strength.

3.1.2 DAQmx Toolkit

The DAQmx toolkit emphasized on one of the key points of the OBSIDAS software: the acquisition of the tire-pavement noise-signal. Employing this toolkit, the programmer could not only successfully capture the signal, but also create a simulated sinusoidal signal as if it were a real input. Typically, associated with this toolkit is the program "NI Measurement and Automation Explorer", which is also referred to as "NI MAX". This program, which is automatically downloaded with the installation of the "NI-DAQmx" driver, assists the developer to manage the NI hardware. This software can create and edit tasks and channels [50]. In the NI world, a task is composed of a variety of channels, each of which could represent an input signal. Therefore, a task could be the acquisition of the trial, while

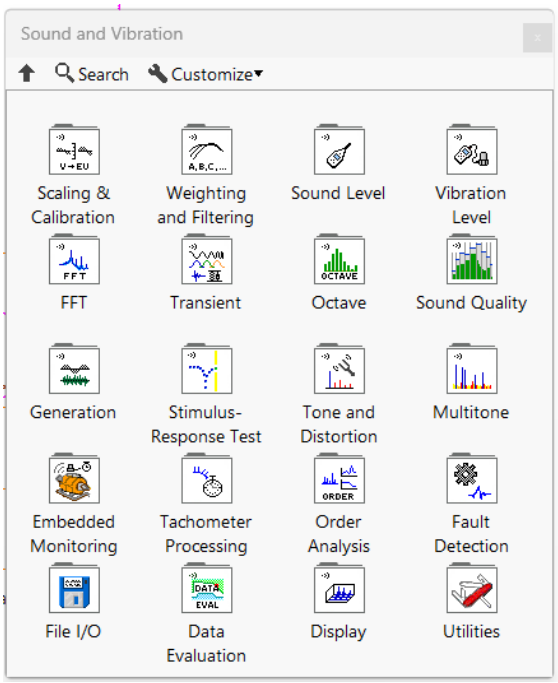


Figure 3.2: Sound and Vibration Toolkit function palette, Source: [49]

each task would collect each of the microphone's signals. As previously stated, the software allows the developer to create simulated signals, which it entails on the creation of simulated channels and tasks.

National Instruments (NI) offers an extensive amount of data acquisition hardware devices. These hardware devices are classified according to the type of measurements they perform, including voltage, current, temperature, sound and vibration, or board-level controllers [51]. The NI-USB4431 was utilized to acquire the sound pressure signals for OBSIDAS, as will be explained in the following section. The driver DAQmx could be then considered a bridge between the real-physical device and the programming environment of LabVIEW. It acquires the signal and, through the implementation of tasks with the appropriate channels, enables the signal to be observed and computed in LabVIEW.

3.1.3 File Input/Output (I/O)

Another library within the LabVIEW framework that will be widely utilized in the OBSIDAS project is the "File Input/Output (I/O)" library. This library is responsible for facilitating communication with files. It has been employed to read past trials from a CSV file and to write the current trial to a CSV file. Furthermore, this resource enables formatting into files, scanning fom files, building paths, or

stripping paths, among other functions. Consequently, the initial (reading of the file), and final (writing to the file) stages of the OBSIDAS program are fulfilled through the employment of this "File I/O" implementation.

3.2 Hardware

This section will analyze and detail the hardware used for the trials and testing of the system. It is worth noting that the entire hardware setup was provided by the *John A. Volpe National Transportation Systems Center*, which is part of the *United States Department of Transportation* and is located in Cambridge, Massachusetts. The overall hardware is as follows:

- NI USB-4431; is a versatile data acquisition device designed for dynamic signal acquisition. It is the bridge between the analog microphones and the computer, and is therefore necessary for the digitalization of the signal.

- Four GRAS 40AI 1/2" intensity microphones (leading in, leading out, trailing in and trailing out); which are specifically designed for the accurate measurement of sound intensity.

- Four GRAS 26AK 1/2" preamplifier, in order to achieve a higher range of magnitude, minimize the noise and distortion, and match the impedance between the microphone and the data acquisition system.

- Bruel and Kjaer 4231 calibrator, in order to calibrate the microphones. The calibrator frequency is 1 kHz and the calibration pressure is 94 dB SPL.

- Two GRAS 12AQ 2-Channel Power Module, to provide the necessary, stable, and clean power to the microphones and preamplifiers. Additionally, the modules include gain adjustment and filtering capabilities, which enhance the quality of the acquired signal.

- Cabling, in order to connect all components, the most commonly used cables are the GRAS 7-pin Lemo and the BNC.

- Main computer with LabVIEW OBSIDAS software implemented, it is important to note that the OBSIDAS software program must be accessible on the primary computer, which will be used to connect the NI USB-4431.

3.2.1 NI USB-4431

The National Instruments USB-4431 [52] is a 24-bit dynamic signal acquisition device utilized to facilitate communication between analog microphones and digital computers. In order for the computer to recognize the device as an acquisition device, the driver *DAQmx*, detailed in section 3.1.2, must be installed. Figure 3.3 depicts a representation of this acquisition device.

The device has four analog input channels (A0, A1, A2, and A3) and one analog output (AO), which is not utilized in this project but may be employed in a stimulus-response test. The four input channels incorporate an IEPE, integrated electronic piezoelectric, to facilitate processing by this measurement device.

The following section presents the main specifications of the device and the analog inputs. The analog output specifications are not included here, as they are not relevant to this project. However, they can be obtained from the source [53]. The rest of the specifications were obtained from the same source [53].

- Power Consumption: 2.5 W max.
- PC Communication Interface: USB 2.0.
- ADC Type: Delta-Sigma, refer to [54] and [55] to gain further insight into this specific type of ADC.
- Sample Rate Range: 1kS/s (samples per second) to 102.4 kS/s.
- Input Range: ±10 Vpk (volts peak).
- Dynamic Range; with a signal that is 60 dBFS (60 dB bellow full scale, which refers to the maximum input level that the ADC can handle), 1kHz tone and sampling rate being 102.4 kS/s the typical dynamic range value (under normal conditions) is 100 dB, and the minimum guaraneed dynamic range (under less optimal condition) is 98 dB.
- AC Coupling -3 dB Cutoff Frequency: 0.8 HZ.
- Operating Temperature: -30 ^{o}C to 70 ^{o}C.
- Relative Humidity: 0% to 95% RH, non-condensing.



Figure 3.3: Dynamic Signal Acquisition NI USB-4431. Source: [52]

3.2.2 GRAS 40AI 1/2" Intensity Microphone

The GRAS 40AI 1/2" Intensity Microphone is a IEC 61094 compliant, WS2F standard externally polarized sound-intensity microphone, as referenced in [56]. Figure 3.4 depicts a representation of this microphone. It is one of the primary components of the project, as it is responsible for acquiring the sound pressure signal of the noise. The following section presents the main specifications of the device, which were obtained from the source [56].

- This microphone operates with zero volts polarization and uses Constant Current Power (CCP)for powering the microphone.
- Frequency Range; accurate within ±1 dB from 50 Hz to 8 kHz, and within ±2 dB from 10 Hz to 20 kHz.
- Dynamic Range; can measure from 20 dBA (A-weighted dB) up to 152 dBA.
- Sensitivity: 25 mV/Pa (or -32 dB with reference 1V/Pa) at 250 Hz.
- Operational and Storage Temperatures; operational from -40 to 150 °C, and storage from -40 to 85 °C.

3.2.3 GRAS 26AK 1/2" Preamplifier

The GRAS 26AK 1/2" Preamplifier is a IEC 61094 compliant preamplifier as referenced in [58]. Figure 3.5 depicts a representation of this preamplifier. The preamplifiers are needed in order to achieve a higher range of magnitude, minimize



Figure 3.4: GRAS 40AI 1/2" Intensity Microphone. Source: [57]

the noise and distortion, and match the impedance between the microphone and the data acquisition system. The following section presents the main specifications of the device, which were obtained from the source [58].

- Frequency Range; along 2.5 Hz to 200 kHz the device maintains an accuracy of ±0.2 dB when using an 18 pF dummy microphone.
- Slew Rate; 20 V/µs, rapid voltage change capability.
- Input and Output Impedance; High input impedance ($20G\Omega$ with 0.4 pF) and low output impedance (75 Ω).
- Output Voltage Swing; up to 12 Vp with ±14V/+28V power, and up to 50 Vp with ±60V/+120V power.
- Noise Levels (20 Hz – 20 kHz): Low noise with typical A-weighted values of 1.8 µV and linear values of 3.5 µV.
- Gain: slight negative gain of -0.25 dB (almost negligible).
- Temperature and Humidity; operates between -30 to 70°C and can be stored between -40 to 85°C; suitable for non-condensing environments up to 95% relative humidity.

3.2.4 Bruel and Kjaer 4231 calibrator

The Bruel and Kjaer 4231 calibrator is a sound level calibrator designed for precise calibration of sound measurement devices (such as sound level meters or microphones). This calibrator is compliant with IEC/EN 60942 (2017) Class 1, and



Figure 3.5: GRAS 26AK 1/2" Preamplifier. Source: [59]

ANSI S1.40–2006 [60]. The calibrator frequency is 1 kHz and the calibration pressure is 94 dB. Figure 3.6 depicts a representation of the calibrator: figure 3.6a represents solely the calibrator, while figure 3.6b depicts how the microphone and preamplifier need to be assembled to perform the calibration.



(a) Bruel and Kjaer 4231 Calibrator. Source: [60]



(b) Calibration Setup. Source: own elaboration

Figure 3.6: Bruel and Kjaer 4231 Calibrator

The following section presents the main specifications of the calibrator, which

were obtained from the source [60].

- Sound Pressure Levels; provides twin sound pressure levels: 94 dB SPL and 114 dB SPL (in noisy environments). In this project, the calibration was conducted using a sound pressure level of 94 dB SPL.
- Calibration Accuracy; precision of ±0.2 dB.
- Frequency; 1 kHz tone.

3.2.5 GRAS 12AQ 2-Channel Power Module

The GRAS 12AQ 2-Channel Power Module s a versatile tool designed for general-purpose dual-channel precision measurements, intensity measurements, standard and custom signal conditioning, and both manual and digital control. It supports CCP and traditional power supplies with 0 V or 200 V polarization options, as referenced in [61]. Figure 3.7 depicts a representation of this power module. The following section presents the main specifications of the device, which were obtained from the source [61].

- Voltage supply; ±15 V or ±60 V for powering up to two microphone preamplifiers.
- Polarization voltage; 200 V or 0 V for two condenser microphones.
- Constant current supply; 4 mA sourced at 28 V DC for two CCP microphone preamplifiers.
- Independent Channel Setup: Each channel can be independently configured for gain, filter, input, output, and polarization voltage.
- Gain: adjustable from -20 dB to +70 dB in discrete 10 dB steps.
- Filters:
  - Linear response with high-pass filter of 0.2 Hz.
  - High-pass filter of 20 Hz.
  - A-weighted response.
  - Customizable external signal-response network.
- Frequency Response:
  - ±0.1 dB at gain ≤50 dB: 10 Hz to 100 kHz

  - ±0.2 dB at gain ≤50 dB: 2 Hz to 200 kHz
  - ±0.1 dB at gain >50 dB: 10 Hz to 20 kHz

- Operation temperature range; -10^{o}C to +50°C.



Figure 3.7: GRAS 12AQ 2-Channel Power Module. Source: own elaboration.

3.2.6 Cabling and Final Setup

In this section an overview of the final setup and the cabling will be presented. The three types of cables that were used in this project are:

- GRAS 7-pin lemo; between preamplifiers and power modules.
- BNC plug; between power modules and NI USB-4431 analog inputs.
- USB type A - USB type B connector; between NI USB-4431 and main computer.

Figure 3.8 depicts an overview of the hardware setup and its connections. As can be observed, the setup shown in the figure 3.9 is not mounted on the vehicle, the final setup to perform the OBSI method according to the AASHTO T360 standard has to be implemented using the same connections but the microphones mounted as shown in figure 1.1 and 2.5.

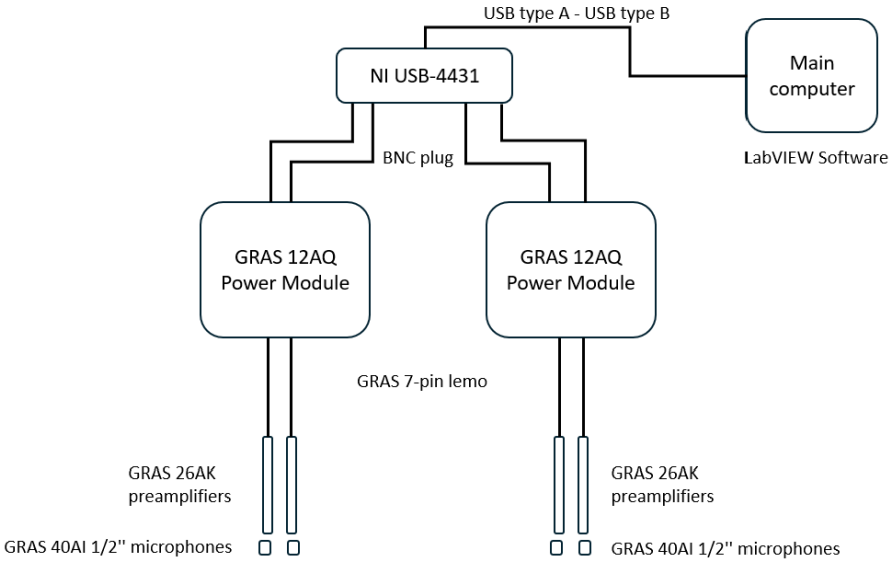


Figure 3.8: Block Diagram of the Hardware Setup and Its Connections. Source: own elaboration.

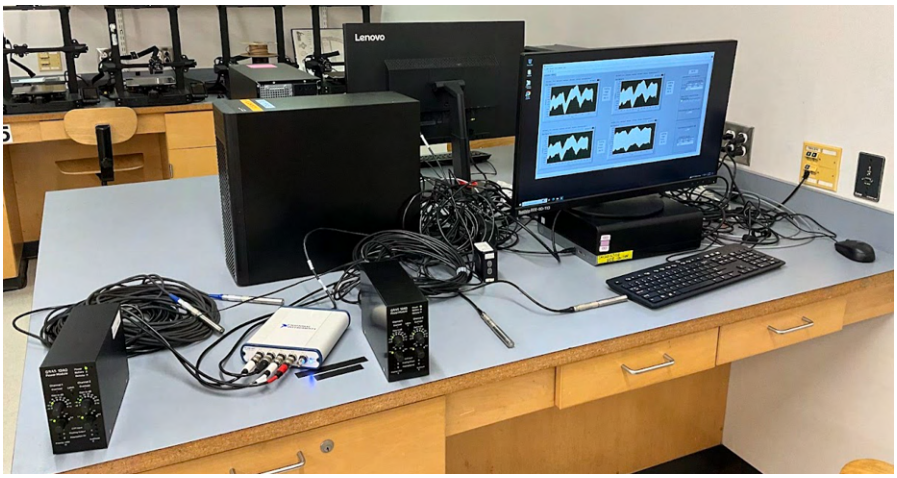


Figure 3.9: Real Hardware Setup and Its Connections. Source: own elaboration.

Chapter 4

State of the Art

This section will present an overview of the current software and systems products available in the market for measuring tire-pavement noise using the OBSI method. The primary technologies are as follows: *"OBSI Module: Real-Time On Board Sound Intensity measurement and post-processing"* [62] from the *SoftdB* company and the *"OBSI System"* [63] from *AVEC (Acoustical and Vibration Engineering Consultants)* will be analyzed in greater detail. Additionally, technologies and systems not based on the OBSI method but related to sound intensity measurements will be examined.

4.1 *SoftdB*

The *SoftdB* product: "OBSI Module: Real-Time On Board Sound Intensity measurement and post-processing", is a hardware-software package -available in [64]- that performs tire-pavement noise according to the AASHTO TP 76-10 standard method. The product consists of a configured laptop, referred to as the conductor, with the appropriate hardware connections to link to the microphones. The laptop also incorporates the OBSI software for the sound pressure measurements and results. The following sections will provide a detailed explanation of each component of the product.

4.1.1 Hardware

The compatible hardware for the OBSI Module are the conductor, "Alto-6 SoftdB", and "Alto-Lite SoftdB".

Conductor

The term "conductor" refers to the laptop on which the software is installed. The user interacts with the program through this tool. This laptop is also connected to the DSP (Digital Signal Processing) analyzers (and these to the microphones). Figure 4.1 illustrates the different connections and sides of the hardware.

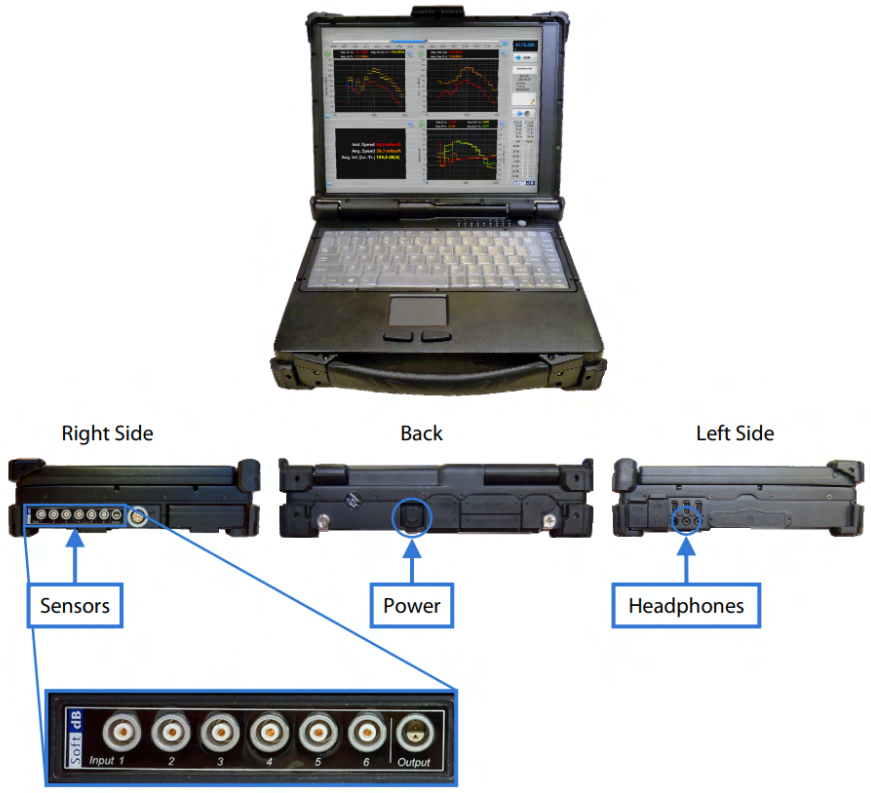


Figure 4.1: SoftdB OBSI Module Conductor Connections, Source: [62]

As stated in the user manual [62],the following connection should be implemented in order to utilize appropriate cabling:

- Channel 1: leave unconnected
- Channel 2: tachometer (optional)
- Channel 3: Leading Edge Probe – Mic A
- Channel 4: Leading Edge Probe – Mic B
- Channel 5: Trailing Edge Probe – Mic A

- Channel 6: Trailing Edge Probe – Mic B

Alto-6i and Alto-Lite

The *Alto* series [65] of *SoftdB* is a high-performance DSP analyzer that, once linked to the conductor and microphones, performs sound and vibration measurements. The Alto-6i has six 24-bit asynchronous inputs and 2 outputs, while the type Alto-Lite has four 24-bit asynchronous inputs and 2 outputs. Both types are equipped with a LEMO 0B Series connection cable and a maximum sampling rate of $48kHz$.

4.1.2 Software

User Interface

TThe manner in which users interact with the program is through the user interface depicted in figure 4.2. This figure will now be detailed in order to provide a comprehensive overview of its various components.

- In the "Measure Controls" area, located in the top right corner of the display, the user may initiate the trial by pressing the "RUN" button. An indicator that displays the elapsed time is presented.
- The "File Info" cluster contains fundamental data about the current file calculator.
- In the section labeled "Audio Controls" part, the user is able to listen to the recorded file of the trial.
- The user is provided with a visual representation of the entire trial measurement time range in the "Min Time Axis".
- In the "Display Area, Legend, display Settings and Rescale Axis", the graphical representation of the results is presented in the following ways: time history graph, one-third octave band graph, FFT graph, and numeric indicators.

In order to calibrate the microphones and to prepare the setup of the sensors in the tab "Setup" there is the "Input Setup" interface, shown in the figure 4.3. In this menu, the user can select all information before starting the test: input channel, input range, input type (AC, DC or ICP), input sensitivity or microphone distance. If a tachometer is added to the configuration, it can also be initialized in this menu.

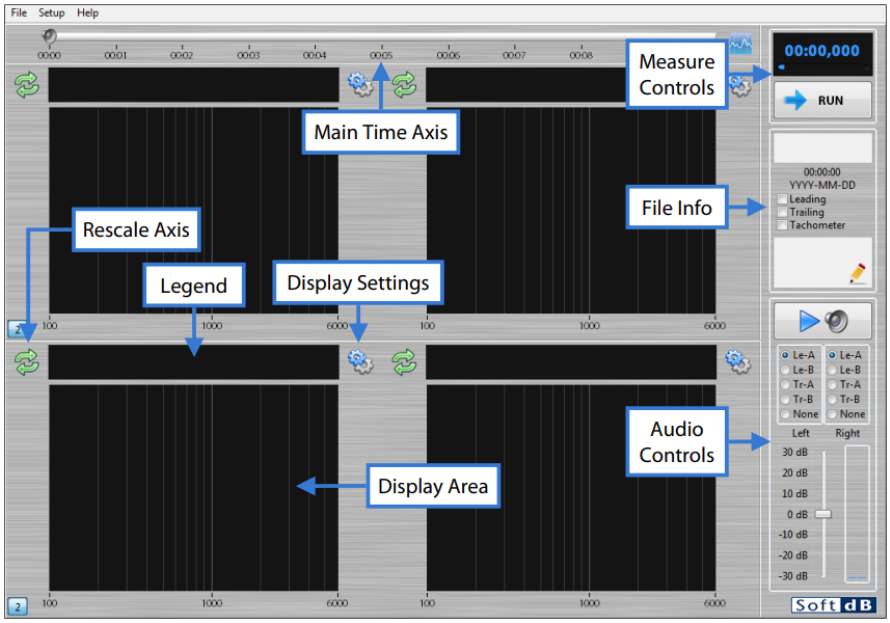


Figure 4.2: SoftdB OBSI Module User Interface, Source: [62]

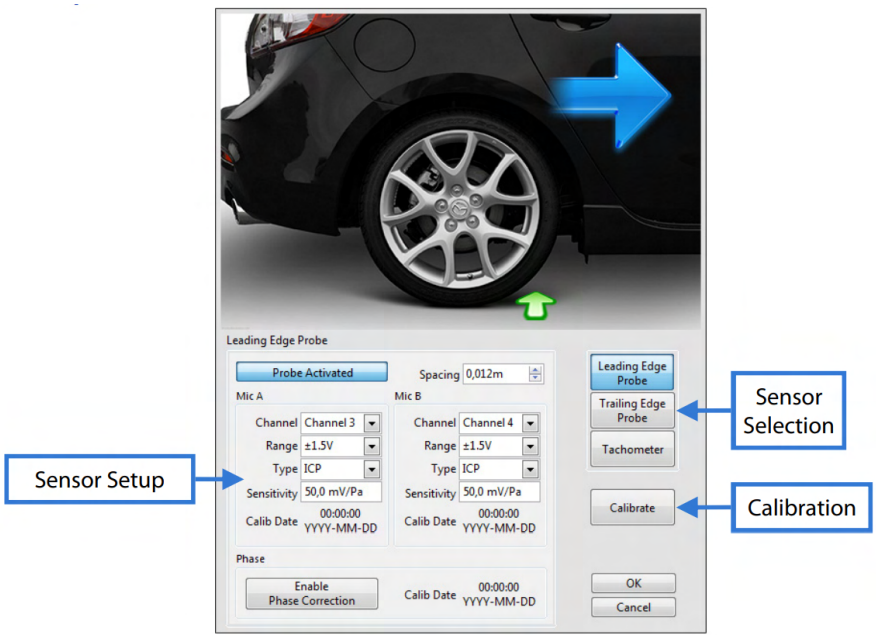


Figure 4.3: SoftdB OBSI Module Input Setup User Interface, Source: [62]

Record Setup

The program includes a menu in which the user is able to adjust and configure the record setup. The user can select whether to save the measurements automatically, if so, the measurement will be saved in the user's desired path (being the default "My Documents/OBSI/").

The data can be saved and exported as a tab-delimited file, compatible with spreadsheet software like Microsoft Excel. The graphs displayed in the main interface are the data which will be saved. Consequently, data that is not visible in the user interface will not be saved.

Measurements and Processing

As previously stated in the first part of section 4.1.2, the user interface (UI) is capable of handling different display types. In order to configure the display and select which measurement and processing signal the user wants to be shown, there is the "Display Area Setup Interface". This menu, depicted in figure 4.4, allows the user to select which available data the program will show. Each of the four graphs can be configured to display up to four plots. These plots can be either intensity and pressure or coherence and pressure intensity index. The offered data are as follows, as quoted from the source text in reference [62]:

- Intensity (dB, dBA, W/m^2 or $W/m^2(A)$) – (Instantaneous or Time averaged)
  - Leading edge probe
  - Trailing edge probe
  - Average leading edge / trailing edge
- Pressure (dB, dBA, $Pa^2(rms^2)$ or $Pa^2(rms^2)(A)$) – (Instantaneous or Time averaged)
  - Leading edge probe (mic A)
  - Leading edge probe (mic B)
  - Leading edge probe (average mic A / mic B)
  - Trailing edge probe (mic A)
  - Trailing edge probe (mic B)
  - Trailing edge probe (average mic A / mic B)
  - Average leading edge / trailing edge
- PI Index (dB, or $linear$) – (Instantaneous or Time averaged)

  - Leading edge probe
  - Trailing edge probe

- Coherence (dB, or $linear$) – (Instantaneous or Time averaged)
  - Leading edge probe
  - Trailing edge probe

- Speed (m/s, km/h, $mile/h$ or ft/s) – (Instantaneous or Time averaged)
- Distance (m, km, $mile$ or ft) – (Instantaneous or Time averaged)

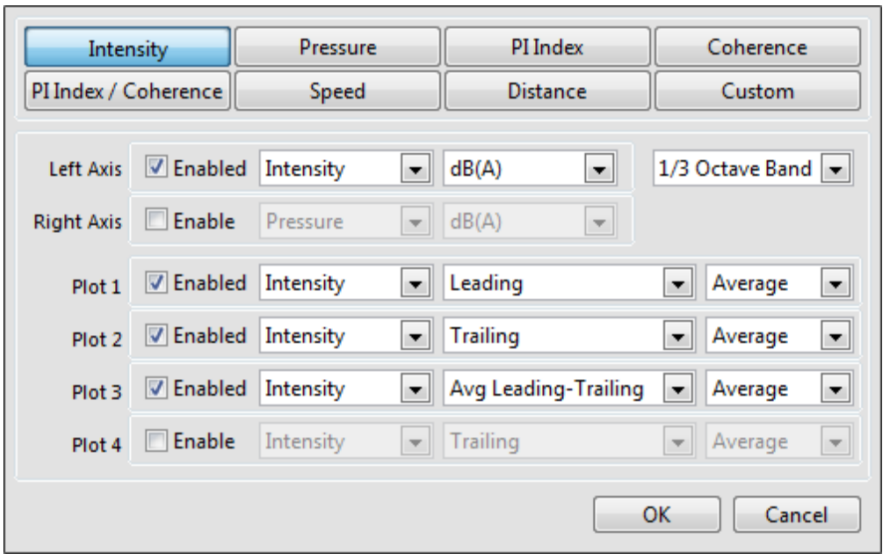


Figure 4.4: SoftdB OBSI Module Display Area Setup Interface, Source: [62]

As a final reference, figure 4.5 shows the final displays and data of a completed trial where the user could retrieve the OBSI results. In this example, the top left graph shows the instantaneous intensity levels of the trailing, leading, and average trailing/leading, while on the bottom left graph shows the time-averaged intensity levels of the trailing, leading, and average trailing/leading. The top right graph shows the instantaneous pressure levels of the leading and trailing, while the bottom right graph shows the time-averaged pressure levels of the trailing and leading.

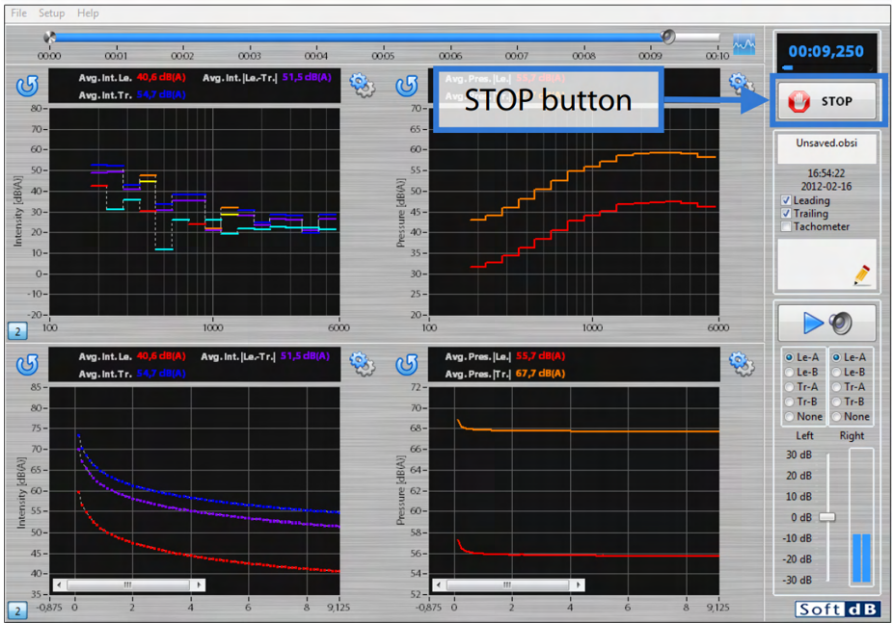


Figure 4.5: SoftdB OBSI Module Final Results, Source: [62]

4.2 *AVEC*

The American company *AVEC, Acoustical and Vibrations Engineering Consultants* [66], has developed an OBSI System for the measurement of the tire-pavement noise. The hardware and software development is based on the current AASHTO T 360 standard, which is explained in section 2.5. The company offers the software and hardware for sale separately. The full product includes the following items, as quoted from [67]:

- AVEC's OBSI software with USB remote for easy software control
- Two GRAS 1/2" intensity probe sets (40GI, four total microphones)
- Two GRAS 1/2" CCP pre-amplifier sets (26CA)
- National Instruments 4-channel DAQ (cDAQ-9171, USB-9234)
- GRAS 42AG sound level calibrator
- Semi-rugged laptop (with software pre-installed)
- Garmin GPS unit and GPS software module for OBSI
- Plastic carrying case for OBSI system

- Wind screens and cables for intensity probes.
- DC/AC converter.

Additionally, there are optional items, described in [67]:

- Continuous mode OBSI software module
- Tachometer
- Accelerometers and in-cabin microphones
- Standard Reference Test Tire (SRTT)
- Car mount for laptop

4.2.1 Software

The software offered by AVEC includes a user interface that is designed to be intuitive and user-friendly. Figure 4.6 provides a representation of such user interface. It allows users to initiate trials in different road sections. As previously stated, this software adheres to the latest AASHTO standard, T 360. Once the trial has been completed, the software will inform the user if the current test is valid based on the AASHTO standard.

Another significant feature of AVEC's software is its compatibility with *Office 365* programs. Automatic reports can be generated in *Microsoft Word*. Furthermore, figures and results may be exported to *Microsoft Excel* or *Microsoft Word*. It is also noteworthy that the software is provided with an unlimited license, as opposed to an annual subscription.

Furthermore, the product includes an USB remote control, which allows the user to control and adjust the trial during the test. The measurements of the sound pressure levels (SPL) may be displayed in a graphical format as a function of distance or time. These levels will be available in full spectrum or divided into one-third octave bands, in accordance with the AASHTO standard.

The software also provides interfaces through which the user can configure the trial. Figure 4.7 illustrates the channels setup, where the user can adjust the microphones' sensitivities or gain, and enable automatic calibration of these microphones. Figure 4.8 illustrates the configuration of weather and test data, where the user can select various parameters, including temperature, barometric pressure, air density, wind speed and direction, and road temperature.

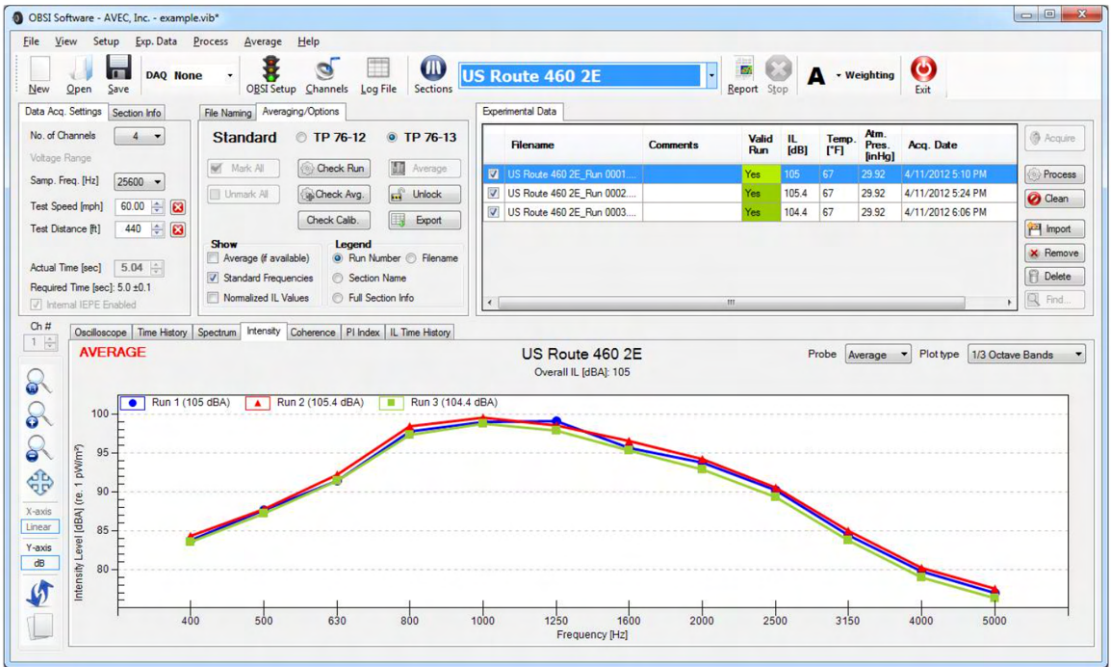


Figure 4.6: AVEC OBSI System User Interface, Source: [63]

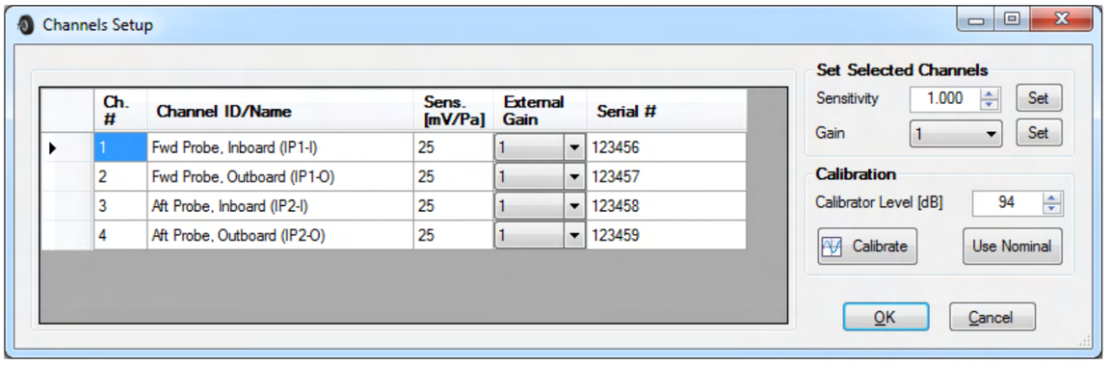


Figure 4.7: AVEC OBSI System Channels Setup Interface, Source: [63]

4.3 Additional Technologies and Systems

A variety of general programs can be adapted to perform the OBSI method. However, these technologies are limited by the absence of the processing required by the AASHTO T360 standard. REW (Room EQ Wizard) and sound level meters are two examples of these technologies. REW can be used for wide sectors of signal processing, among them the predominant one is room acoustic processing [68]. Sound level meters are employed to measure ambient sound pressure levels,

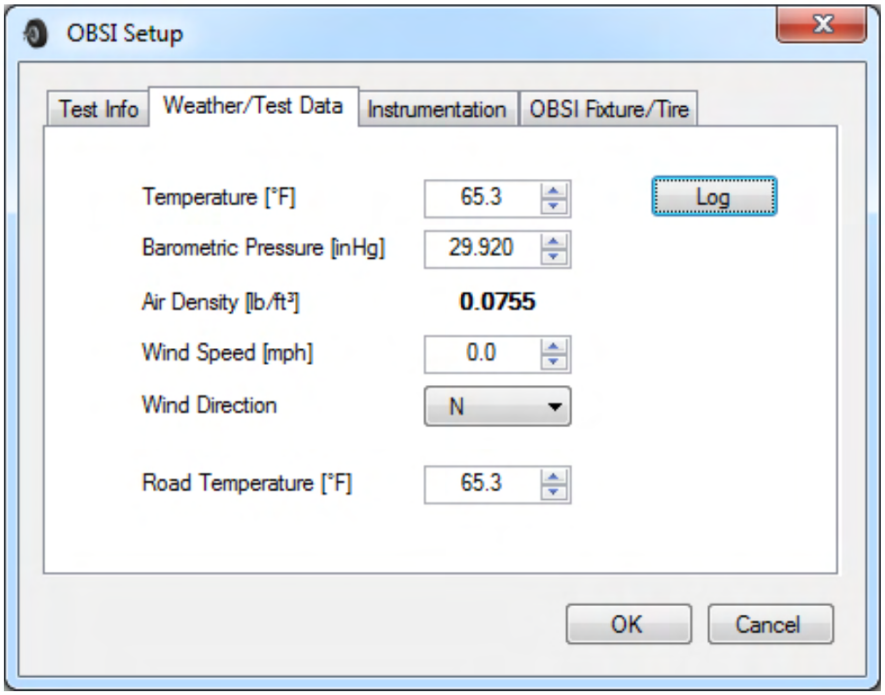


Figure 4.8: AVEC OBSI System Weather Interface, Source: [63]

without any division into bands or normalization (in contrast to the standard requirements). Consequently, these other technologies are not specifically designed for the OBSI method, but can be employed as a proxy for it. The following sections will explain these technologies in greater detail.

4.3.1 Room EQ Wizard

The Room EQ Wizard is a free software program utilized to measure the acoustic properties of rooms, speakers, and audio products. This software is designed to generate a study of the room in question, whether that be a cinema or an office. Its purpose is to identify the optimal configuration for the speakers, to measure the responses of the room to sound, and to counter the effects of room modal resonances [69].

This software is capable of measuring sound pressure levels, impedance, and distortion, as well as computing frequency and impulse responses. It is also able to generate phase, spectral decay, and group delay graphs, as well as real-time analyzer (RTA) graphs [68]. Furthermore, it can implement equalizer measurements, including EQ window, EQ filter panels, and room simulation.

4.3.2 Sound Level Meters

A variety of sound level meters are currently available on the market. These products are capable of measuring the sound pressure levels in a given location. As an illustration of this, one may cite the Testo Sound Level Meter [70] or the TopTes TS-501B Decibel Meter [71].

The Testo Sound Level Meter, depicted in Figure 4.9, is designed to capture sound pressure levels in accordance with the current IEC 61672-1 standard, as referenced in [72]. The meter allows the user to select a time weighting option, either "slow" for signals exhibiting gradual changes in sound level (1 second) or "fast" for signals exhibiting rapid fluctuations in sound level (125 msec.). A selection of a A or C filter weighting may be made in order to more accurately measure the sound in question.

As demonstrated by the REW software, these types of sound pressure level meters do not align with the AASHTO standard for the OBSI method. However, they may be utilized to create an approximation of the method.



Figure 4.9: Testo 816-1 Sound Pressure Lever Meter, Source: [70]

Chapter 5

Project Definition

5.1 Justification

This section will elucidate the rationale behind the OBSIDAS project, which provides the market with a robust, updated, and reliable piece of software that not only adheres to the last AASHTO T360 standard but also offers an innovative and user-friendly program to perform this standard. Furthermore, in addition to this standard, the program can perform sound level, sound quality, and frequency domain measurements.

This section will present a detailed account of the previous work conducted in the field of tire-pavement noise measurement, as outlined in chapter 4. It will also highlight the enhancements and additional functionalities that the OBSIDAS software has incorporated, in comparison to the aforementioned programs.

5.1.1 SoftdB OBSI

Outdated Standard

The primary improvement that OBSIDAS has over the software detailed in section 4.1 is that this last adheres to the standard AASHTO TP-76, which is a provisional standard antecedent to the AASHTO T360. In other words, *SoftdB* is based on an outdated standard. The present OBSIDAS software is based on the most recent, revised standard, which can be accessed via [28]. Consequently, the primary improvement OBSIDAS has made in comparison to SoftdB OBSI is the incorporation of the latest AASHTO standard.

Machine-Bound

Moreover, the *SoftdB OBSI* program is subject to a significant disadvantage that the OBSIDAS program addresses: machine-bound or device-specific. The *SoftdB* program is designed to operate exclusively within the conductor, as detailed in section 4.1.1. In contrast, the OBSIDAS software is capable of running on any machine, including laptops and personal computers. This program is implemented into a type .EXE file, which can be installed on any machine. This second improvement that OBSIDAS performs in comparison to *SoftdB* makes OBSIDAS more accessible to the general public, as no machine is required.

5.1.2 AVEC OBSI

Compatibility

Similar to what happens with the *SoftdB OBSI*, the AVEC OBSI program is subject to a significant disadvantage that the OBSIDAS program addresses: compatibility. The AVEC program is designed to operate exclusively within the operatin system Mirosoft Windows 7 version. In contrast, the OBSIDAS software is capable of running on any machine and version, including laptops, and latest Microsoft Windows version. This program is implemented into a type .EXE file, which can be installed on any machine. This improvement that OBSIDAS performs in comparison to AVEC OBSI makes OBSIDAS more accessible to the general public, as no machine or version is required.

5.1.3 Additional Features

In addition to the improved features that OBSIDAS offers in comparison to other software previously described, the OBSIDAS software incorporates supplementary processing and measurements that extend beyond the scope of the AASHTO T360 standard. This provides users a broader spectrum of measurements to quantify the tire-pavement noise. It is noteworthy that neither of the two programs under comparison in this section incorporates these additional capabilities. The following is a list of additional measurements provided by the OBSIDAS program.

- Visual and graphical representation of the one-third octave band of sound pressure and sound pressure level.
- Frequency domain measurements including: two different FFT graphical analysis (using the Sound and Vibration Toolkit [30], and secondly using LabVIEW native programming), and graphical representation of the power spectral density.

- Sound level measurements including:
  - Equivalent continuous sound level (L_eq).
  - Exponential averaged sound level, additionally the user could select the exponential mode between: slow (1 second), fast (125 msec.), impulse (35 ms rising and 1.5 s falling), or custom as the user will input the time desired.
  - Peak sound level.
- Sound quality measurements including:
  - Stationary loudness (sone) and loudness level (phon). Additionally, the specific loudness (sones/bark) is graphically represented in critical band rate (bark).
  - Sharpness from the sound pressure signal waveform. Additionally, a graphical representation of the aures sharpness is performed.
  - Roughness of the sound pressure signal according to Aures' method [30].

Furthermore, the OBSIDAS software also allows the user to save the measurements and the raw data in a CSV type file. Additionally, this software has four possible inputs:

1. Real device acquisition
2. Simulated device acquisition (using the LabVIEW task and channel simulator)
3. CSV file input
4. Wav file input

In addition, the OBSIDAS software provides a real-time graphical interface to see what the microphone (or simulated or file sources) is capturing in real time, as detailed in figure 5.1.

5.2 Objectives

This project: OBSIDAS, is a software program developed in LabVIEW that is designed to acquire the noise signal, compute the necessary sound level measurements, and graphically display the result in order to perform the AASHTO (American Association of State Highway and Transportation Officials) [27] standard T360 [28].

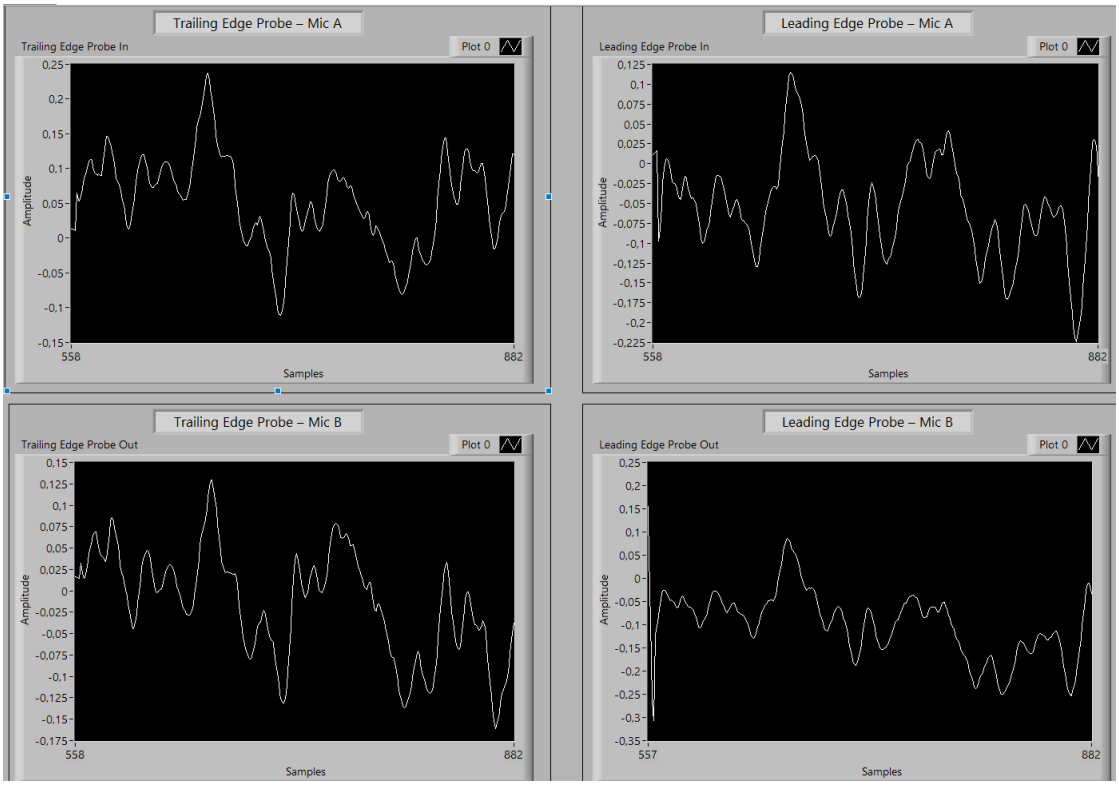


Figure 5.1: Acquisition Interface of the OBSIDAS Program, Source: own elaboration

OBSIDAS not only adheres to all AASHTO T360 procedures, but also presents the results in graphs and schemes in a highly intuitive user interface. Furthermore, it performs additional sound quality and level measurements, as well as frequency analysis, thereby offering the user more information about the OBSI test.

Therefore, the following is a list of goals that OBSIDAS must achieve.

1. Develop a user interface where the end user could navigate and interact with the software. Such an interface must have two tabs: an acquisition tab, where the user will edit and add all functions related to the acquisition step prior to signal processing; and a results tab, where the user could collect and view the final results.

2. The program must be able to acquire the signal from four different sources:

   - Real device acquisition
   - Simulated device acquisition (using the LabVIEW task and channel simulator)

- CSV file input
- Wav file input

3. The program must allow the user to select any feature related to microphone calibration (sensitivity, end units, dB reference, pregain, or weighting filter).

4. The program must be able to save the raw collected data and the OBSI results to a CSV type file.

5. The program must have a real time graph to show the user the data that this program is collecting.

6. The program must calculate and graphically display for each microphone the following:

   - Acquired raw data
   - A-weighted one-third octave band sound pressure RMS
   - A-weighted one-third octave band sound pressure level
   - FFT
   - Power spectral density
   - Exponential sound level L_{eq}
   - Loudness (sones/bark)
   - Aures Sharpness
   - Aures Roughness

7. The program must calculate and numerically display for each microphone the following: (Note that when across-probe is used, it means that two measurements are obtained: one for the leading probe and one for the trailing probe. Across-Probe means that the measurements for the two microphones of a probe are averaged to obtain a single measurement for each probe.) (On the other hand, overall means that a single measurement is taken for the four microphones. Overall means that the measurements for each across-probe are averaged to obtain a single measurement for the walk test.)

   - A-weighted one-third octave band sound pressure RMS
   - A-weighted one-third octave band sound pressure RMS total band power
   - A-weighted one-third octave band sound pressure level
   - Normalization of the A-weighted one-third octave band sound pressure level to the current temperature

- A-weighted one-third octave band sound pressure level total band power
- FFT peaks
- Linear sound level L_{eq}
- Peak sound level L_{eq}
- Maximum exponential sound level L_{eq}
- Loudness in each critical band rate (Bark)
- Loudness (Sone)
- Loudness level (Phon)
- Aures Sharpness in each critical band rate (Bark)
- Aures Roughness in each critical band rate (Bark)
- Individual OBSI level for each microphone
- Across-probe arithmetic average SPL
- Across-probe OBSI level
- Individual pressure intensity index for each one-third octave band of each microphone
- Energy averaged overall run test OBSI level
- Energy averaged overall SPL

8. The program must be able to save the computed measurements and the raw data in a CSV type file

5.3 Methodology

The agile method [73], which is based on the collaborative effort between the end user and the developers, adaptive planning to such feedback from the end users, evolutionary development, early delivery, and continuous improvement, will be employed throughout the work process. Therefore, since John A. Volpe National Transportation Systems is the main customer of this software, their feedback and proposals for improvement will be taken into account. These inputs will be integrated into the project during its production phase, rather than being incorporated into the latest version. This methodology offers benefits to both parties, as the developer is able to idenitfy the user's needs and preferences, while the user is able to provide feedback and assessment of the initial software versions.

The primary procedure employed throughout the project to address an issue or implement a feature has been as follows.

1. Establish a plan for the necessary time frame to achieve the objective.

2. Review the theoretical background of the new feature or issue; by establishing the theoretical basis, every new objective can be addressed with complete knowledge of this objective. It is noteworthy that the webpage `https://sengpielaudio.com/` by the Grammy Award-winning German sound engineer Eberhard Sengpiel has been of great value and guidance in order to establish the fundamental theoretical knowledge.

3. Implementing the feature in LabVIEW; it is of note that the *Sound and Vibration Toolkit User Manual* by LabVIEW has been of great value and guidance in establishing the fundamental knowledge regarding signal and sound processing measurements that is required to use LabVIEW effectively.

4. Test the new feature.

5. Contact the customer to obtain feedback on the newly added feature.

5.4 Planning and Budgeting

5.4.1 Gantt Chart

This section will present an overview of the planning that was undertaken in order to achieve the objectives. The entire year has been divided into weeks, and each of the goals has been established as a target for completion by a specific week.

A Gantt chart has been employed for the administration of the task. The diagrams presented in the following figures represent the most recent version, as they have been modified throughout the project due to external influences or tasks that have exceeded the anticipated time frame.

Note that the final paper is due on the 17^{th} of June, while the oral presentation of the project is due on the 18^{th} of June.

Figure 5.2 depicts the Gantt chart of the hardware tasks scheduled from October 23, 2023, to June 18, 2024.

Figure 5.3 depicts the Gantt chart of the user interface tasks scheduled from January 1, 2024, to February 26, 2024.

Figure 5.4 depicts the Gantt chart of the user interface tasks scheduled from March 4, 2024, to April 29, 2024.

Figure 5.5 depicts the Gantt chart of the user interface tasks scheduled from May 6, 2024, to June 18, 2024.

Figure 5.6 depicts the Gantt chart of the LabVIEW processing tasks scheduled from January 1, 2024, to February 26, 2024.

Figure 5.7 depicts the Gantt chart of the LabVIEW processing tasks scheduled from March 4, 2024, to April 29, 2024.

Figure 5.8 depicts the Gantt chart of the LabVIEW processing tasks scheduled from May 6, 2024, to June 18, 2024.

Figure 5.9 depicts the Gantt chart of the documentation tasks scheduled from October 23, 2023, to December 25, 2023.

Figure 5.10 depicts the Gantt chart of the documentation tasks scheduled from January 1, 2024, to February 26, 2024.

Figure 5.11 depicts the Gantt chart of the documentation tasks scheduled from March 4, 2024, to April 29, 2024.

Figure 5.12 depicts the Gantt chart of the documentation tasks scheduled from May 6, 2024, to June 18, 2024.

It is noteworthy that during the first semester, a theoretical investigation and study was performed about the AASHTO T360 standard [28], LabVIEW Sound and Vibrations Toolkit [30], and sound level measurements [74].

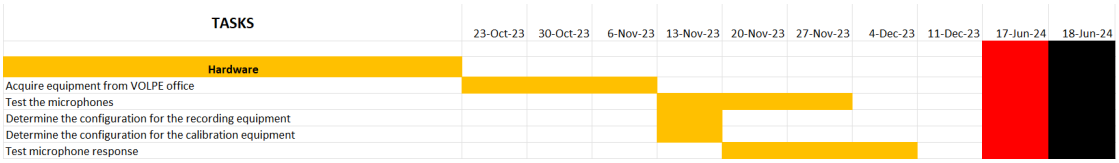


Figure 5.2: Gantt chart of the hardware tasks scheduled from October 23, 2023, to June 18, 2024. Source: own elaboration

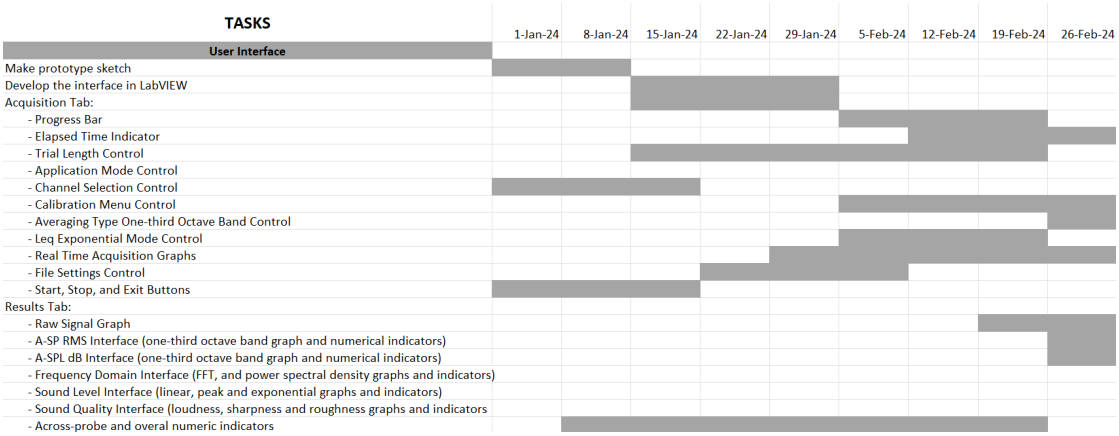


Figure 5.3: Gantt chart of the user interface tasks scheduled from January 1, 2024, to February 26, 2024. Source: own elaboration

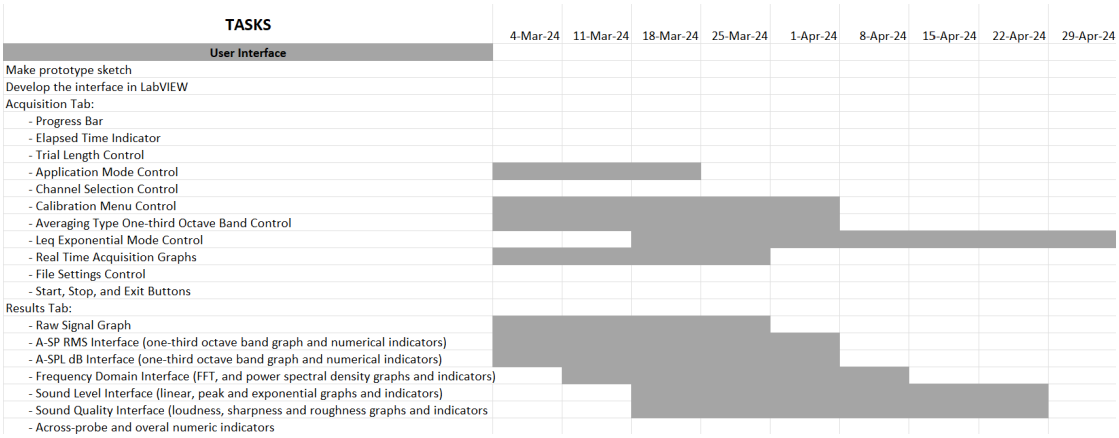


Figure 5.4: Gantt chart of the user interface tasks scheduled from March 4, 2024, to April 29, 2024. Source: own elaboration

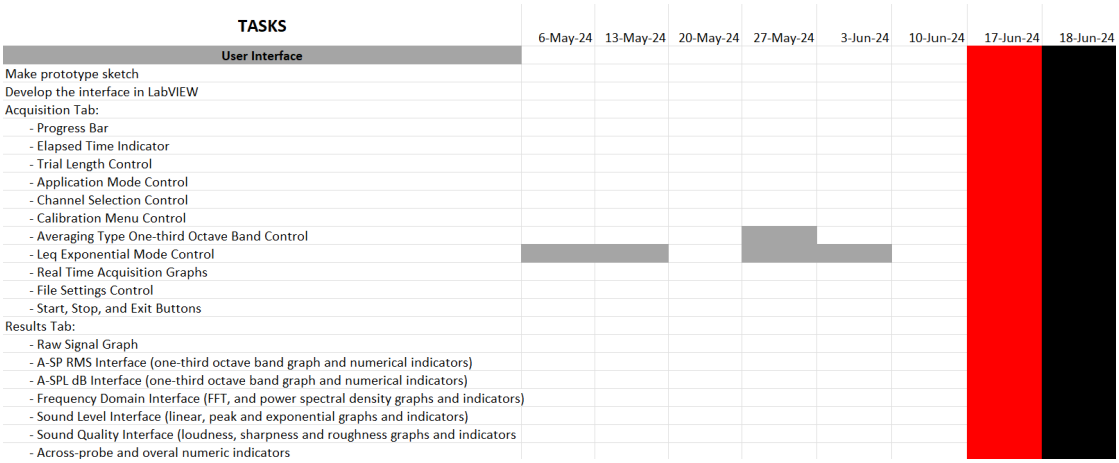


Figure 5.5: Gantt chart of the user interface tasks scheduled from May 6, 2024, to June 18, 2024. Source: own elaboration

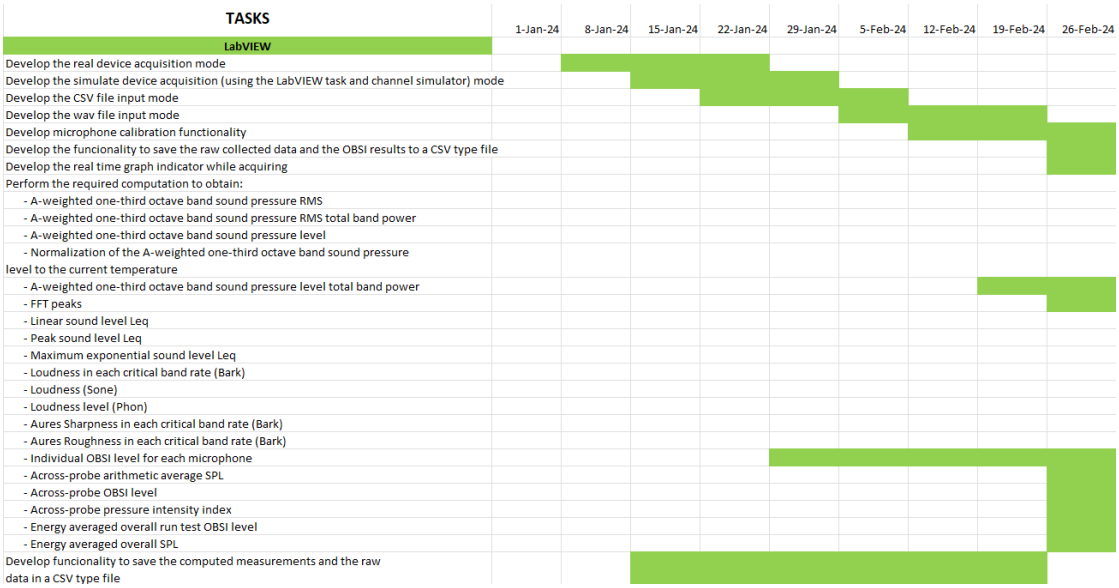


Figure 5.6: Gantt chart of the LabVIEW processing tasks scheduled from January 1, 2024, to February 26, 2024. Source: own elaboration

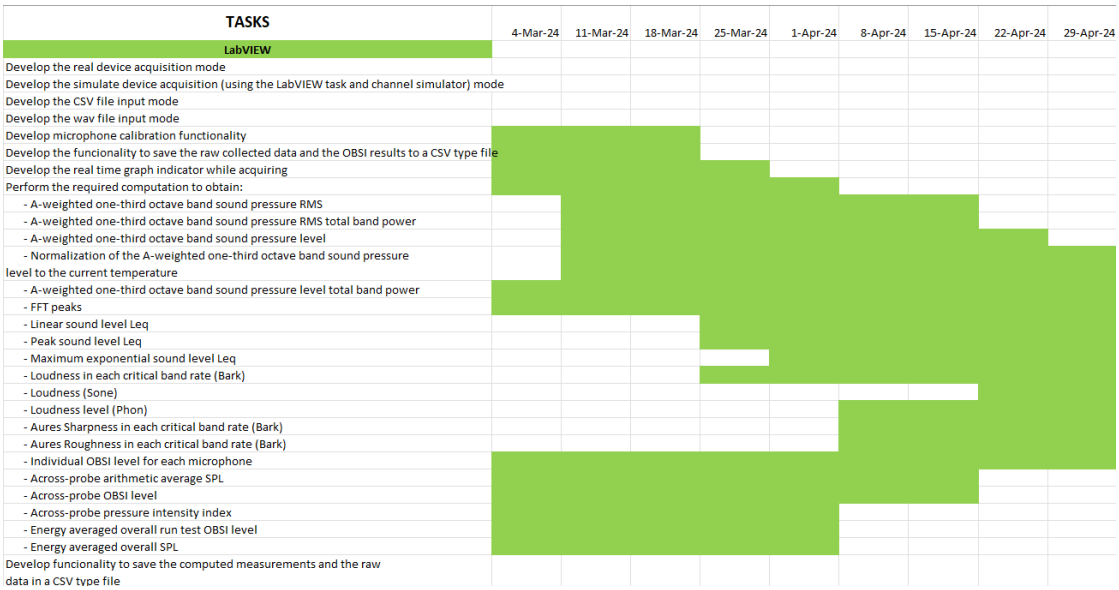


Figure 5.7: Gantt chart of the LabVIEW processing tasks scheduled from March 4, 2024, to April 29, 2024. Source: own elaboration

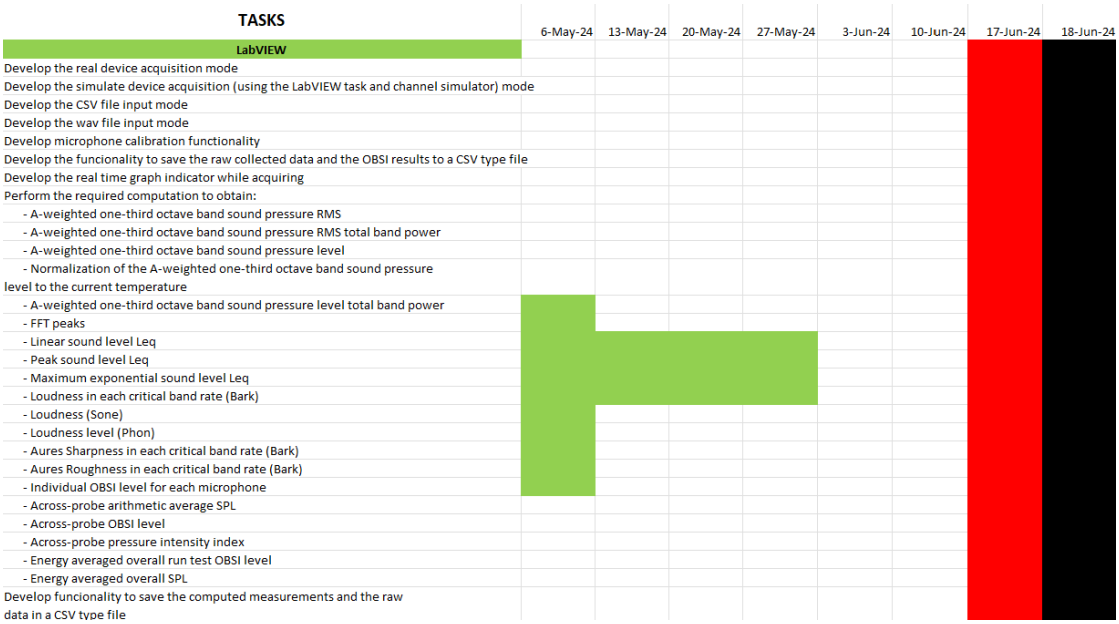


Figure 5.8: Gantt chart of the LabVIEW processing tasks scheduled from May 6, 2024, to June 18, 2024. Source: own elaboration



Figure 5.9: Gantt chart of the documentation tasks scheduled from October 23, 2023, to December 25, 2023. Source: own elaboration



Figure 5.10: Gantt chart of the documentation tasks scheduled from January 1, 2024, to February 26, 2024. Source: own elaboration

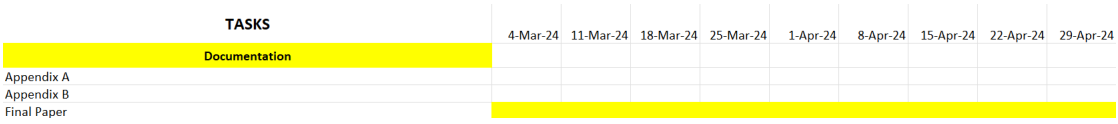


Figure 5.11: Gantt chart of the documentation tasks scheduled from March 4, 2024, to April 29, 2024. Source: own elaboration

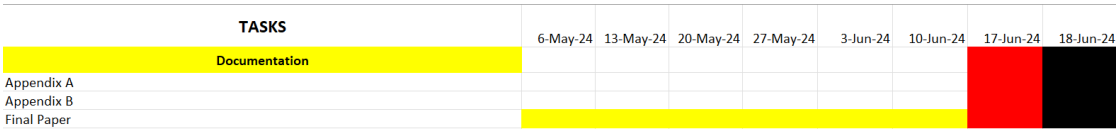


Figure 5.12: Gantt chart of the documentation tasks scheduled from May 6, 2024, to June 18, 2024. Source: own elaboration

5.4.2 Budget management

As commented beforehand, all hardware items have been provided by John A. Volpe National Transportation Systems Center. The other financial impacted part of the project are the LabVIEW licenses and toolkits. Table 5.1 depicts the financial impact of the project.

As previously noted, all hardware items have been provided by the John A. Volpe National Transportation Systems Center. The other financial aspect of the project that requires consideration is the cost of LabVIEW licenses and toolkits. Table 5.1 depicts the financial impact of the project.

Item	**Price**
LabVIEW Full	$ 1,664.00/year [75]
LabVIEW Sound and Vibration Toolkit Full	$ 1,952.00/year [76]
NI-DAQ mx	Free [77]
NI USB-4431	$ 4,466.00 [52]
GRAS 40AI 1/2" Ext. Polarized Intensity Microphone Pair	upon request [56]
GRAS 26AK 1/2" preamplifier	upon request [58]
Bruel and Kjaer 4231 calibrator	upon request [60]
GRAS 12AQ 2-Channel Power Module	upon request [61]
Laptop	$ 1000 aprox
GRAS 7-pin Lemo x4	Upon request in official webpage [78] $ 45.00/each = $ 180.00 from [79]
BNC cable x4	$ 14.48/each = $ 57.92 total [80]
USB type A - USB type B connector	$ 7.79 [81]
Total	**$ 9,327.71**

Table 5.1: Equipment Costs

Chapter 6

System Developed Overview

This chapter will expose an overview of the key features and elements of the LabVIEW system: the OBSIDAS program. It will provide a foundation for understanding and integrating the various components of the program. These components will be analyzed in greater detail in subsequent chapters. This section can be seen as an introduction to the OBSIDAS program.

The general structure of the program's block diagram is encapsulated in a while loop, in order to prevent the program from exiting unless the user explicitly requests it to do so. Inside this while loop, the program is divided into seven subdiagrams of a flat sequence structure. This flat sequence structure is defined in reference to [82] as:

> *Consists of one or more subdiagrams, or frames, that execute sequentially. Use the Flat Sequence structure to ensure that a subdiagram executes before or after another subdiagram. Data flow for the Flat Sequence structure differs from data flow for other structures. Frames in a Flat Sequence structure execute from left to right and when all data values wired to a frame are available. The data leaves each frame as the frame finishes executing. This means the input of one frame can depend on the output of another frame.*

The utilization of a flat sequence structure has enabled the sequential selection of which subdiagram will run. The seven subdiagrams that comprise the OBSIDAS program will be explained in the subsequent sections, in order they are run. Additionally, a list of these subdiagrams or frames is provided below.

1. Idle While Loop;
2. Selection of the Application Mode;

3. Timestamp;
4. Acquisition;
5. Division of the Signal, Get Temperature, and Error Handler
6. Signal Processing;
7. Save to file

The figure 6.1 is a state machine that illustrates the manner in which the program will execute each subdiagram. Each state represents a single subdiagram. As previously noted, at the conclusion of each state, the program will automatically transition to the next one.

The user will only be able to interact with the program in the following states:

- In the first state, subdiagram 1: idle while loop, tthe user can make a selection from the front panel menus, and start a new trial.
- In the fourth state, subdiagram 4: acquisition, the user is afforded the option to terminate the current trial acquisition, at which point the program will transition to the next subdiagram.
- Upon completion of the subdiagram 6: signal processing, the program will prompt the user to indicate whether the current trial should be saved as a CSV file. The user may select either "YES" or "NO" as the response.
- In each subdiagram, the user is afforded the option of terminating the program by selecting the "EXIT" button.

6.1 Subdiagram 1: Idle While Loop

This subdiagram will run unless the user selects to start a trial, and when this trial is finished the program will automatically come back to this subdiagram. In addition, the subdiagram depicts five distinct menus that the user may select from as needed. The menus are as follows:

1. Menu for the selection of the exponential mode of the sound level measurements.
2. Menu for the selection of the averaging type for the one-third octave band analysis.

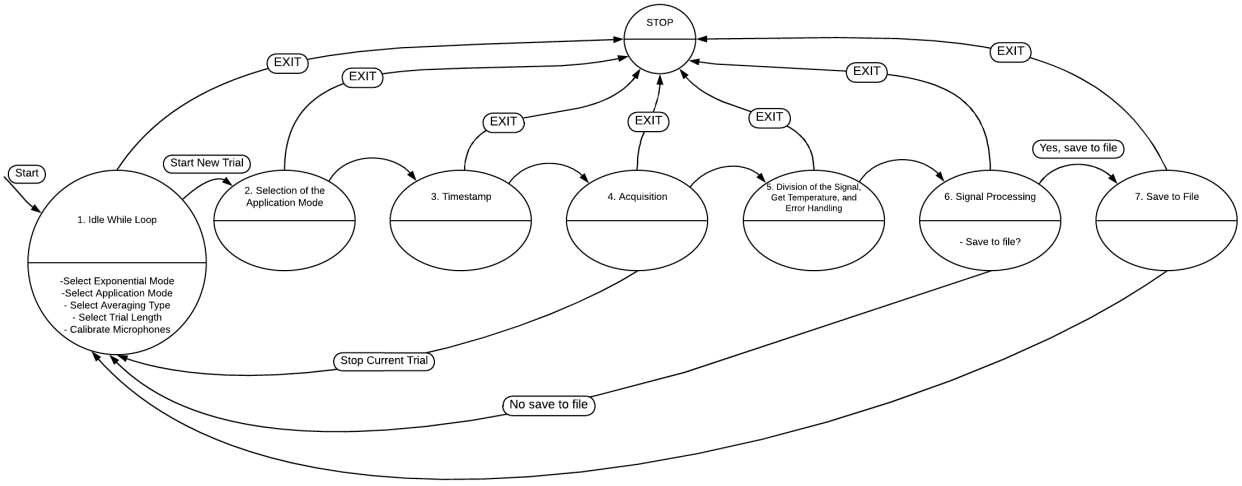


Figure 6.1: OBSIDAS Program State Machine. Source: own elaboration

3. Menu for the selection of the OBSIDAS application mode. Which allows the user to choose which source the application will use to collect the signal. The available sources are:
    - Real Device
    - Simulated Device
    - Type .wav File
    - Type CSV File
4. Menu for the microphones calibration. Where the user can select for each microphone:
    - Sensor sensitivity
    - dB reference
    - Weighting filter
    - Engineering units
    - Pregain
5. Menu for the selection of the trial length (in seconds)

Furthermore, this subdiagram incorporates the labels of the user interface. These labels inform the user of the microphone from which the graph is generated. During the execution of this subdiagram, the user is permitted to select menus,

initiate a trial, or terminate the application. It should be noted that upon completion of a trial, the application will return to this subdiagram, displaying the results of the previous trial. Consequently, during the execution of this subdiagram, the user will be able to review the previous results.

The user, after selecting the preferred options in the menus just seen, could start the trial and the program will go through the next subdiagram: 2 Selection of the application mode.

The complete description and analysis of this subdiagram can be found in chapter 8.

6.2 Subdiagram 2: Selection of the Application Mode

During this subdiagram, the program will establish the requisite values and constants regarding the selected application mode. The application mode permits the user to select the source from which the application will collect the signal. The available sources are:

- Real Device
- Simulated Device
- Type .wav File
- Type CSV File

Throughout this subdiagram, the program will open the necessary files, or create the necessary tasks. It is the first step for the acquisition of the signal.

The complete description and analysis of this subdiagram can be found in chapter 9.

6.3 Subdiagram 3: Timestamp

Throughout this subdiagram, the program will obtain the current timestamp prior to acquiring the signal waveform. This will be designated as t_0, the start time of the acquired signal waveform. Additionally, this timestamp will be prompted to the user via the user interface, which will be explained in the chapter 7.

The complete description and analysis of this subdiagram can be found in chapter 10.

6.4 Subdiagram 4: Acquisition

Throughout this subdiagram, the application acquires the signal through the channel specified by the user in the application mode. During the acquisition process, the signal is displayed in real time through graphs in the user interface. This allows the user to visualize in real-time what the program is acquiring.

As the source of the acquisition changes with respect to the application mode, a distinct acquisition approach will be developed for each mode. The program employs a case structure to differentiate between modes. Additionally, a while loop is required to traverse the entirety of the signal.

During this subdiagram, the user may select to finish the acquisition by pressing the button "Stop Current Trial". If so, the program will go on to the next subdiagram and start the division, the signal processing, etc. of the signal acquired until the button was pressed.

In this subdiagram, the user may select to terminate the acquisition process by pressing the button labeled "Stop Current Trial" If this is done, the program will proceed to the next subdiagram and initiate the division, signal processing, and other operations on the signal acquired up to the point when the button was pressed.

While the program is acquiring the signal, the user interface displays a progress bar and an elapsed time indicator. These allow the user to know the percentage of the signal that has been acquired at that time and the number of seconds that have elapsed during the acquisition process.

The complete description and analysis of this subdiagram can be found in chapter 11.

6.5 Subdiagram 5: Division of the Signal, Get Temperature, and Error Handler

In this subdiagram, the signal acquired in the previous frame will be divided into the four microphones' signals, and four distinct local variables will be created in order to facilitate the selection of the signals in the subsequent subdiagrams.

With the timestamp obtained in the subdiagram 3, the signal acquired in the previous subdiagram and the inverse of the sample frequency, each of the microphone's waveform will be created.

In this subdiagram, the program will obtain the current temperature through an API request in order to compute the normalization of sound intensity levels to a reference temperature.

Furthermore, during this subdiagram, the program will address all the errors

that were previously prompted. Here, the user will receive a notification that explains which error caused the program to terminate.

The complete description and analysis of this subdiagram can be found in chapter 12.

6.6 Subdiagram 6: Signal Processing

In this subdiagram the primary focus of the OBSIDAS program will be developed: the sound-noise measurements. During this subdiagram the numerical and graphical results will be displayed in the user interface. This subdiagram will use the signal divided in the previous subdiagram and will perform the following signal processing computations (as stated in section 5.2):

- Graphical measurements:
  - Acquired raw data
  - A-weighted one-third octave band sound pressure RMS
  - A-weighted one-third octave band sound pressure level
  - FFT
  - Power spectral density
  - Exponential sound level L_{eq}
  - Loudness (sones/bark)
  - Aures Sharpness
  - Aures Roughness
- Numerical measurements:
  - A-weighted one-third octave band sound pressure RMS
  - A-weighted one-third octave band sound pressure RMS total band power
  - A-weighted one-third octave band sound pressure level
  - Normalization of the A-weighted one-third octave band sound pressure level to the current temperature
  - A-weighted one-third octave band sound pressure level total band power
  - FFT peaks
  - Linear sound level L_{eq}
  - Peak sound level L_{eq}

- Maximum exponential sound level L_{eq}
- Loudness in each critical band rate (Bark)
- Loudness (Sone)
- Loudness level (Phon)
- Aures Sharpness in each critical band rate (Bark)
- Aures Roughness in each critical band rate (Bark)
- Individual OBSI level for each microphone
- Across-probe arithmetic average SPL
- Across-probe OBSI level
- Energy averaged overall run test OBSI level
- Energy averaged overall SPL

The complete description and analysis of this subdiagram can be found in chapter 13.

6.7 Subdiagram 7: Save to File

This subdiagra,, as explained in the previous sections, will be run only if the user respond "YES" to the prompt message, which will be displayed after the signal processing, asking whether to save the data in a CSV type file or not.

Therefore, if the user does not agree to save the trial to a CSV file, this subdiagram will not be run. On the other hand, the current experiment is saved to a file. The program will create two files: one where the raw acquired signal will be stored, and the second where the resulting computed data will be placed.

In the prompt message, the user can select the name of the file where the data will be saved.

The complete description and analysis of this subdiagram can be found in chapter 14.

Chapter 7

User Interface

Throughout this chapter, the User Interface will be deeply analyzed and explained. This user interface will be the LabVIEW front panel.

The interface is divided into two main tabs:

1. Acquisition Tab; contains menus to select the application mode, control trial length, and access buttons and the progress bar, among other features. Figure 7.1 shows the representation of this acquisition tab.

2. Results Tab; reflects all the graphical and numerical computations through graphs and indicators. Figure 7.2 and 7.3 show the representation of this results tab.

It should be noted that the user interface is represented throughout the entire block diagram. In other words, there is no subdiagram that has been designed with the user interface in mind. However, in each subdiagram, there are graphs, indicators, or constants that contribute to the user interface.

In this section, the front panel will be the sole focus of analysis. The section of the interface which is located along the block diagram, will be explained in detail throughout the rest of the chapters, with each subdiagram being discussed individually.

In the following sections, each tab will be analyzed.

7.1 Acquisition Tab

This tab is divided into three primary sections:

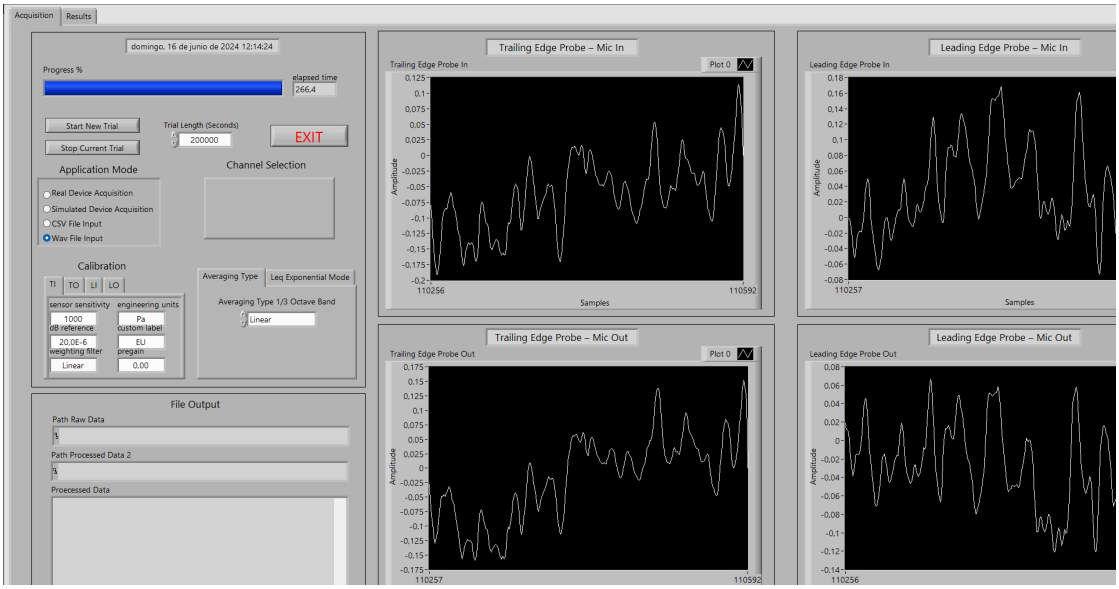


Figure 7.1: Acquisition Tab of the OBSIDAS User Interface. Source: own elaboration

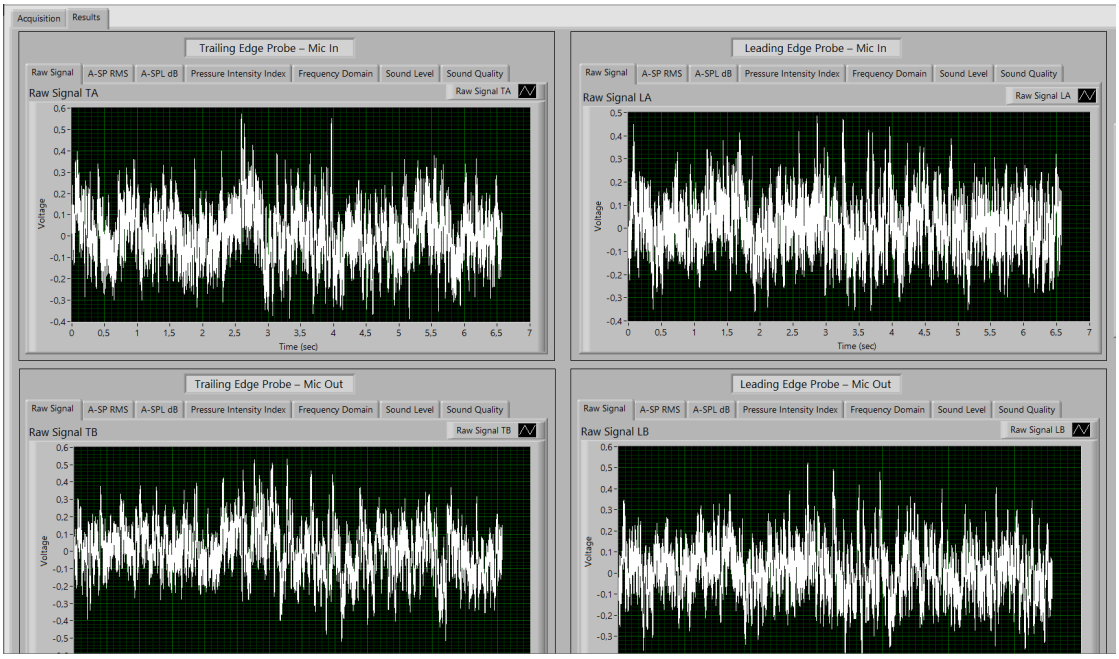


Figure 7.2: Results Tab of the OBSIDAS User Interface Part I. Source: own elaboration

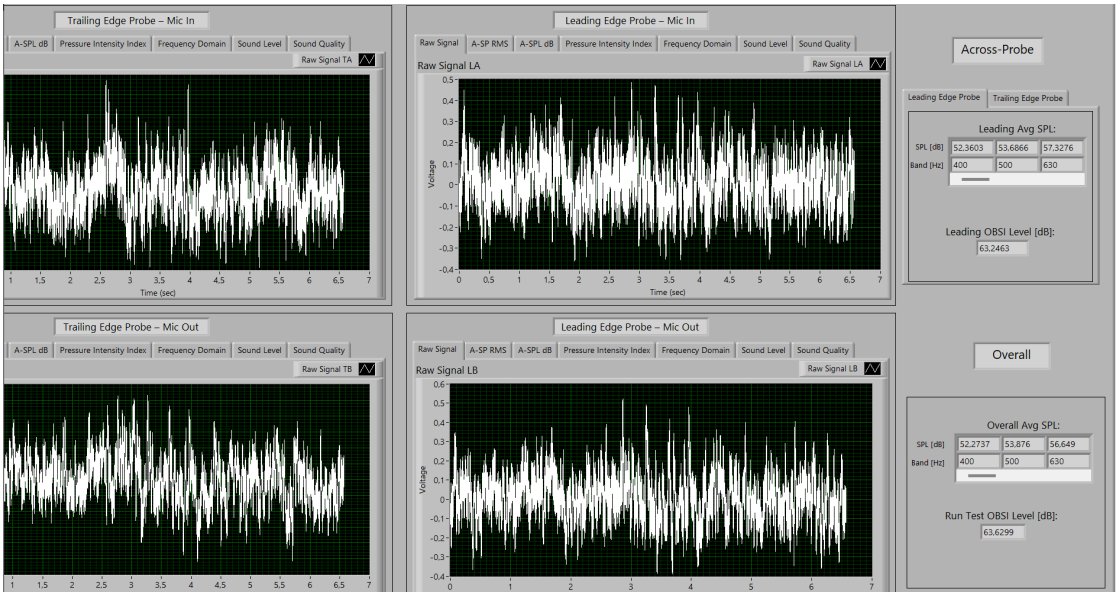


Figure 7.3: Results Tab of the OBSIDAS User Interface Part II. Source: own elaboration

- Acquisition Settings
- File Output
- Real Time Graphs

7.1.1 Acquisition Settings

In this section the user could select all the options prior to acquire the signal and regarding this acquisition. Figure 7.4 depicts a representation of this acquisition settings.

Indicators

In this menu the following indicators are displayed:

- Date and time indicator, the program will automatically capture the laptop's date and time and display it in this indicator.
- Percentage % progress bar, which represents the current stage of the acquisition evolution.

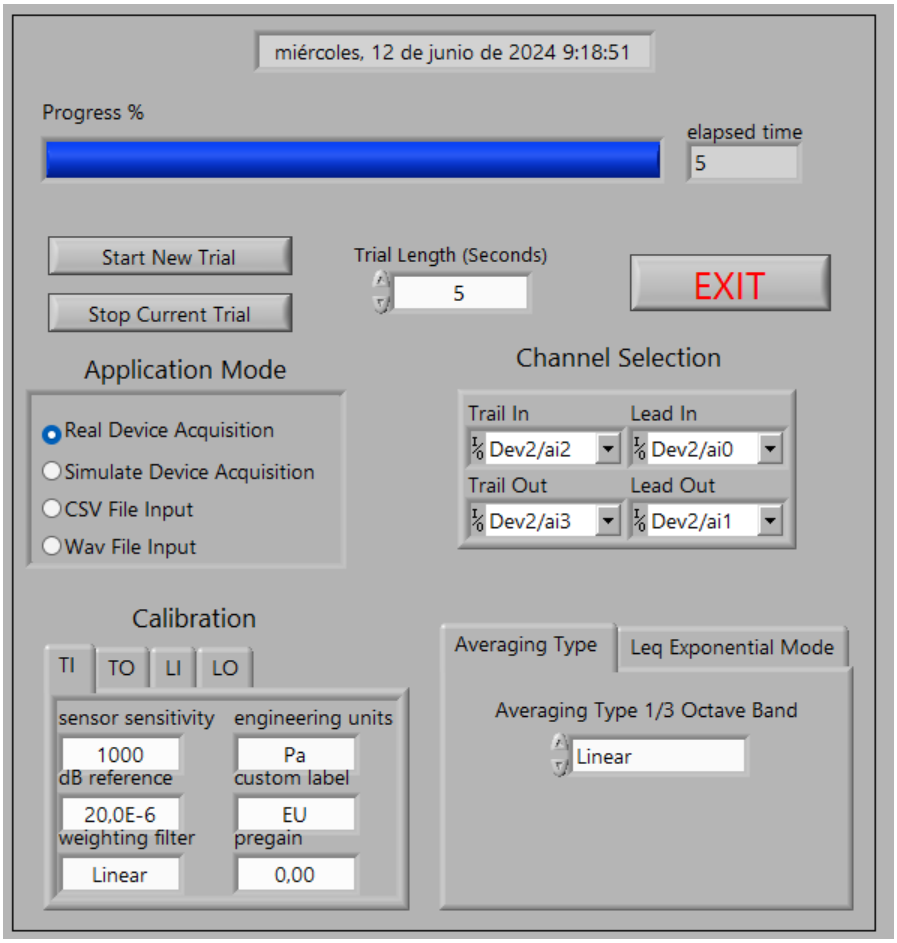


Figure 7.4: Acquisition Settings of the Acquisition Tab of the OBSIDAS User Interface. Source: own elaboration

- Elapsed time indicator, which displays the number of seconds that have elapsed since the program initiated the acquisition process.

Buttons

Additionally, the menu also includes buttons to decide whether to start the trial, stop the trial or exit the program. The functions of these buttons are described in the following list.

- Start New Trial, when clicked, the program will start signal acquisition in accordance with the specified acquisition settings.
- Stop Current Trial, when clicked, the acquisition procedure will be aborted

and the signal processing will be done with the signal acquired up to that point.

- Exit, when clicked, the program exits, this means that the user is done using OBSIDAS. The distinction between the "Exit" button and the "Stop Current Trial" button is that when the user selects the "Stop Current Trial‘" button, a new trial can be initiated immediately by selecting the "Start New Trial" button. In contrast, when the "Exit" button is pressed, the program will terminate.
- Trial Length control, the user can select the number of seconds to be acquired in this control. The program will automatically stop the trial when this time has elapsed.

Application Mode and Channel Selection

The "Application Mode" is the menu where the user could select which acquisition source to use in the trial. As preciously commented, the options are:

1. Real Device; if selected, the channel selection menu displays four drop-down menus that allow the user to select the physical device to be used for each of the microphones.
2. Simulated Device, if selected, the channel selection menu displays a drop-down menu that allow the user to select the simulated device to be used for the microphones.
3. Type .wav File, if selected, four pop-up windows will prompt the user to select a WAV file for each microphone's data.
4. Type CSV File, if selected, a pop-up window will prompt the user to select a CSV file containing all four microphones' data.

Calibration Settings

The calibration settings are displayed in the lower portion of the Acquisition Settings, as illustrated in Figure 7.4. This menu comprises four entries, one for each microphone. In each entry, the calibration options are selected. The following options are available:

- Sensor Sensitivity, which will be expressed in units of $mV/engineering\ Units$.
- Engineering Units, including volts, pascals, newtons, and grams, among others. For this project, it is necessary to select the unit of *Pascal*.

- dB reference, this will be the reference used in the calibration function. For this project, it is necessary to select the reference of 20μ.

- Custom label: engineering units (EU).

- Weighting filter, the user could decide which filter to use in order to calibrate the microphones. For this project, it is necessary to select the linear weighting.

- Pregain. For this project, it is necessary to select the pregain of $0,00$.

Averaging Type and Leq Exponential Mode

The menu comprises two tabs: the first allows the user to select the averaging type for the one-third octave band, while the second enables the user to choose the exponential mode for sound level measurement. Such menu settings are displayed in the lower portion of the Acquisition Settings, as illustrated in Figure 7.4.

Regarding the averaging type for the one-third octave band, the available options are:

1. Linear

2. Exponential

3. Equal Confidence, if selected, an additional menu will be displayed, allowing the user to add the equal confidence level (in dB) manually.

4. Peak

With regard to the sound level exponential mode, it should be noted that an indicator is displayed, indicating the number of seconds each mode will perform. The available options are as follows:

1. Slow, 1 second.

2. Fast, 125 milliseconds.

3. Impulse, 35 milliseconds rising and 1.5 milliseconds falling.

4. Custom, if selected, an additional menu will be displayed, allowing the user to input the number of milliseconds manually.

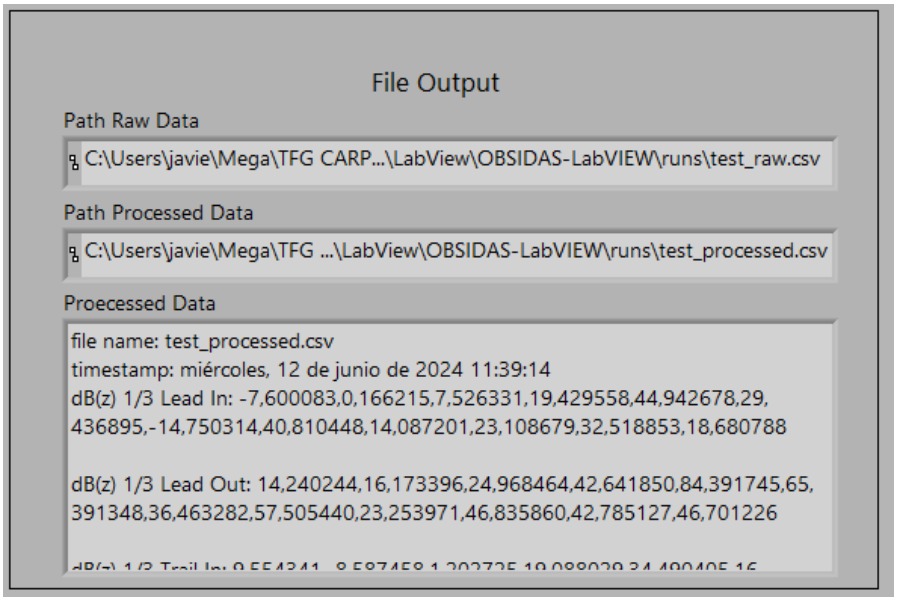


Figure 7.5: File Output section of the Acquisition Tab of the OBSIDAS User Interface. Source: own elaboration

7.1.2 File Output

If the user has selected to save the trial into a file, the output created data could be reviewed in this section by the user. Figure 7.5 depicts a representation of the file output section of OBSIDAS.

In this section, If the user has selected to save the trial into a file, three different piece of information will be displayed:

1. Path of the CSV document with the raw data (acquired data).
2. Path of the CSV document with the processed data (file with the OBSI and OBSIDAS measurements).
3. Review of the CSV document with the processed data (file with the OBSI and OBSIDAS measurements).

7.1.3 Real Time Graphs

This section presents a real-time graphical representation of the acquired signal. It comprises four graphs, each representing the data from a different microphone. During the acquisition of the signal, the aforementioned graphs will be updated

automatically, thereby allowing the user to observe the signal in real time as it changes. Figure 7.6 depicts a representation of the OBSIDAS real time graphs.

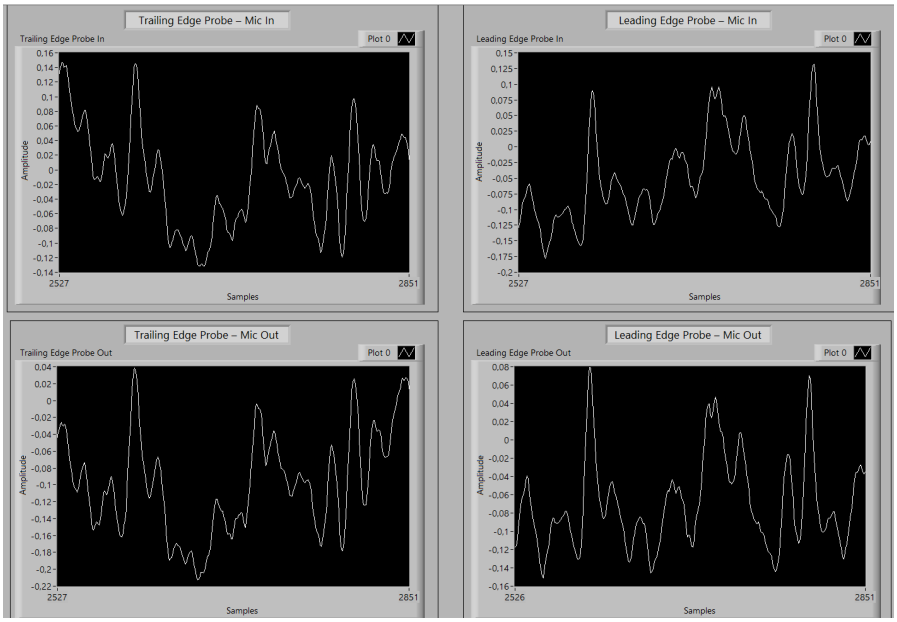


Figure 7.6: Real Time Graphs of the Acquisition Tab of the OBSIDAS User Interface. Source: own elaboration

7.2 Results Tab

This tab is divided into two primary sections:

- Individual Microphone Performance Analysis
- Across-Probe and Overall Performance Analysis

7.2.1 Individual Microphone Performance Analysis

Figure 7.7 depicts a representation of the individual microphone performance analysis section of the results tab of the OBSIDAS user interface.

This section is divided into four parts, each representing one of the four microphone results. Each part is equivalent, differing only in which microphone is the source. Each part reflects the graphical and numerical results of the OBSIDAS

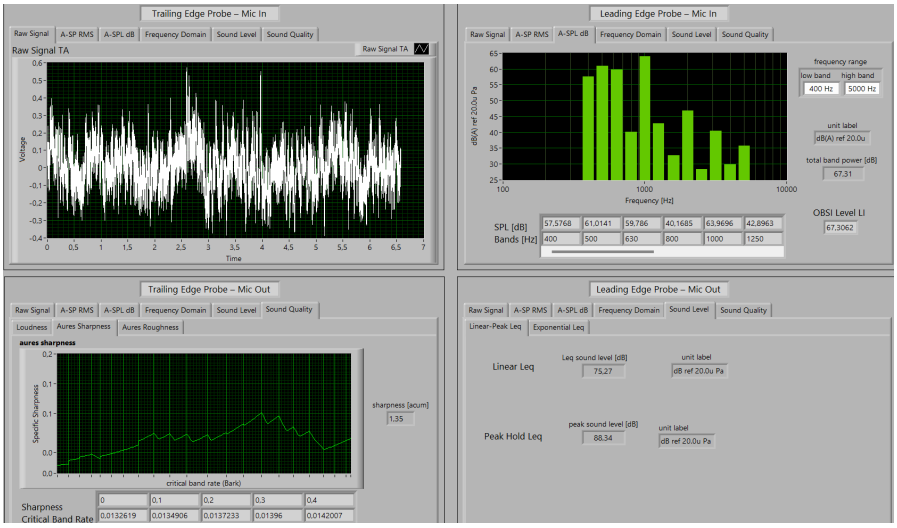


Figure 7.7: Individual Microphone Performance Analysis section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

signal processing. These results are divided into tabs. Some of these tabs have additional tabs within them, allowing for the organization of the interface. The primary tabs and subtabs are as follows:

- Raw signal, graphical representation of the acquired signal (time-amplitude).
- A-SP RMS, sound pressure (SP) RMS A-weighted one-third octave band results.
- A-SPL dB, sound pressure level (SPL) dB A-weighted one-third octave band results.
- Pressure Intensity Index (dB), sound pressure intensity index dB A-weighted one-third octave band results.
- Frequency Domain, divided into the following subtabs:
  - FFT analysis, using LabVIEW native.
  - FFT, using the Sound and Vibration Toolkit.
  - Power Spectral Density, using the Sound and Vibration Toolkit.
- Sound Level, divided into the following subtabs:
  - Linear-Peak Leq, Equivalent Continuous Sound Pressure Level
  - Exponential Leq, Equivalent Continuous Sound Pressure Level

- Sound Quality, divided into the following subtabs:
  - Loudness
  - Aures Sharpness
  - Aures Roughness

The following sections will analyze each of the tabs and subtabs. Note that the figures will be from the Trailing Edge Probe microphone in (or micophone A).

Raw signal

Figure 7.8 represent the tab where the raw acquired signal is displayed. It consists of a graph in which the Y axis corresponds to the voltage transformed by the microphone. While the X axis is time in seconds.

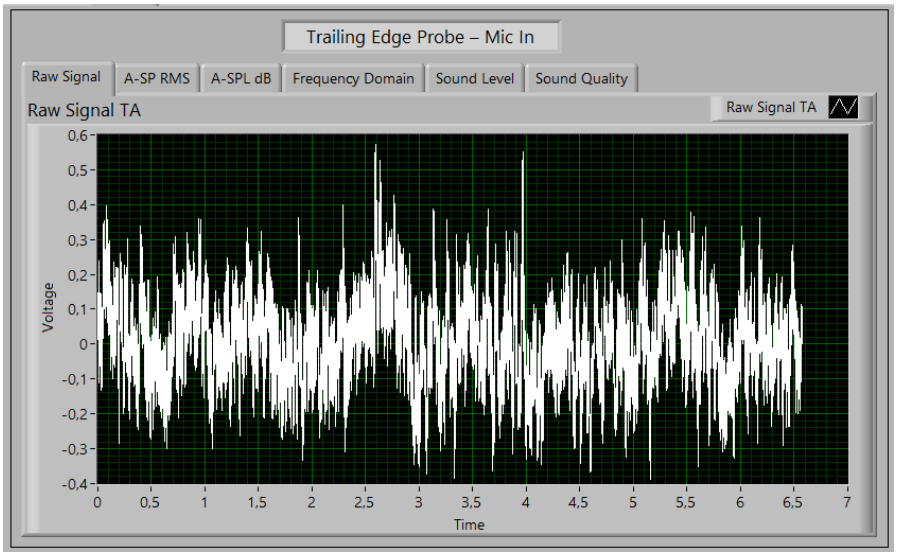


Figure 7.8: Tab "Raw Signal" of the Trailing Edge Probe Microphone In, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

A-SP RMS

Figure 7.9 represents the tab where the sound pressure (SP) RMS A-weighted one-third octave band is displayed. It consists of a graph in which the Y axis corresponds to the A-weighted sound pressure in Pa^2RMS. While the X axis is the frequency bands, ranging from 400 Hz to 5000 Hz. Furthermore, a numerical

indicator of the graph is presented, which is divided into the twelve one-third octave bands. Additionally, the total band power of the whole spectrum is displayed, as well as the frequency range (400Hz to 5000Hz) and the unit label (Pa^2RMS).

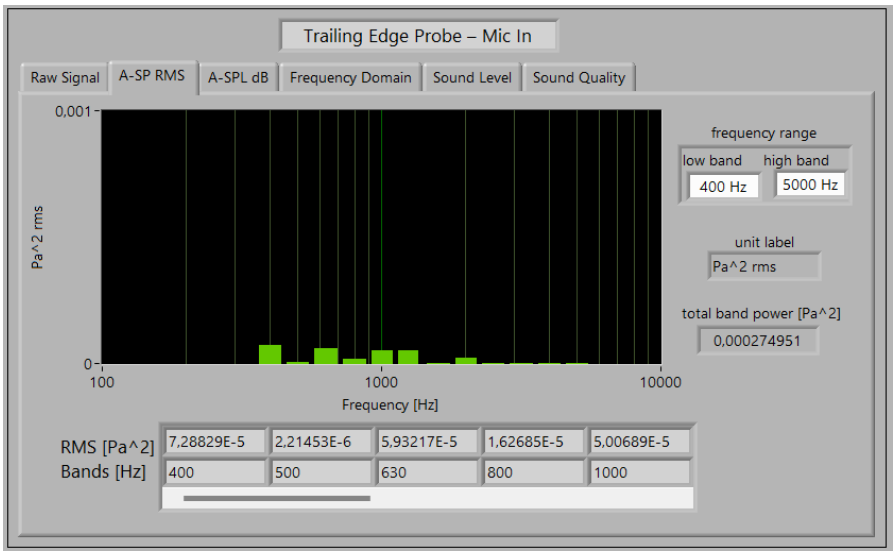


Figure 7.9: Tab "A-SP RMS" of the Trailing Edge Probe Microphone In, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

A-SPL dB

Figure 7.10 represents the tab where the sound pressure level (SPL) dB A-weighted one-third octave band is displayed. It consists of a graph in which the Y axis corresponds to the A-weighted sound pressure level in dB. While the X axis is the frequency bands, ranging from 400 Hz to 5000 Hz. Furthermore, a numerical indicator of the graph is presented, which is divided into the twelve one-third octave bands. Additionally, the total band power of the whole spectrum is displayed, as well as the frequency range (400Hz to 5000Hz), the unit label (Pa^2RMS), and the OBSI level of the microphone.

Pressure Intensity Index

Figure 7.11 represents the tab where the sound pressure intensity index dB A-weighted one-third octave band is displayed. It consists of a numerical indicator

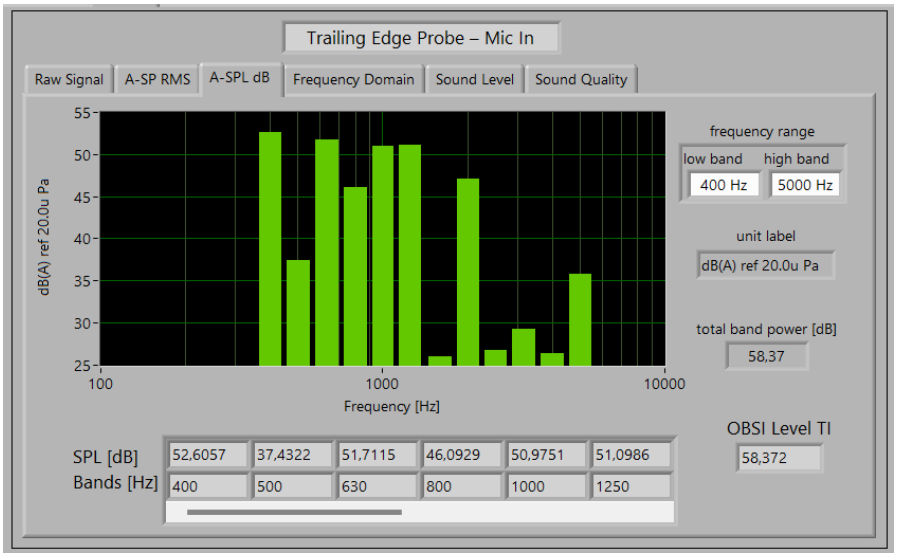


Figure 7.10: Tab "A-SPL dB" of the Trailing Edge Probe Microphone In, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

of the pressure intensity index in dB divided into twelve one-third octave bands from 400 Hz to 5000 Hz.

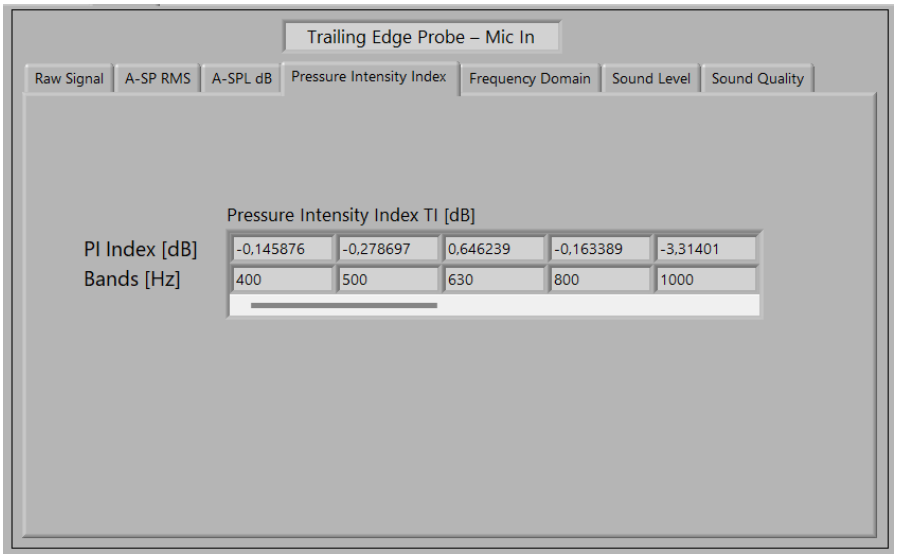


Figure 7.11: Tab "Pressure Intensity index" of the Trailing Edge Probe Microphone In, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

Frequency Domain

This tab is divided into three additional subtabs: "FFT Analysis", "FFT", and "Power Spectral Density". It should be noted that the figures presented in this FFT domain study are not derived from the same trial as the other results tabs. In order to obtain a more accurate representation of the frequency, a 1 kHz sinusoidal signal was utilized. Additionally, the leading Edge Probe Microphone Out was employed.

FFT Analysis Figure 7.12 represents the subtab "FFT Analysis" from the tab "Frequency Domain". Where the FFT graph of the acquired signal is displayed. This FFT is obtained by LabVIEW native computations, as explained in chapter 13.

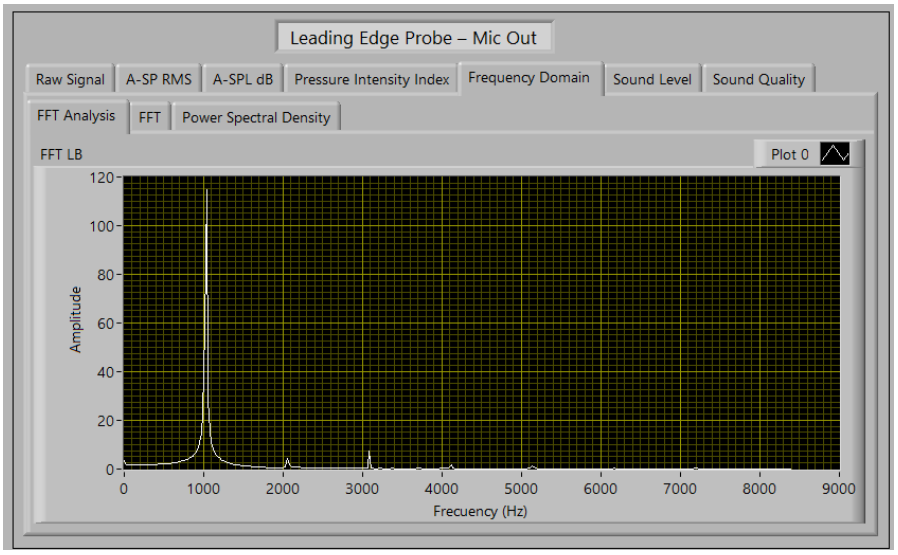


Figure 7.12: Subtab "FFT Analysis", from the tab Frequency Domain of the Leading Edge Probe Microphone Out, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

FFT Figure 7.13 represents the subtab "FFT" from the tab "Frequency Domain". Where the FFT graph of the acquired signal is displayed. This FFT is obtained by the Sound and Vibration Toolkit resources, as explained in chapter 13. A numerical FFT peak search display is also represented.

Power Spectral Density Figure 7.14 represents the subtab "Power Spectral Density"' from the tab "Frequency Domain". Where the power spectral density graph across the whole spectrum of the acquired signal is displayed.

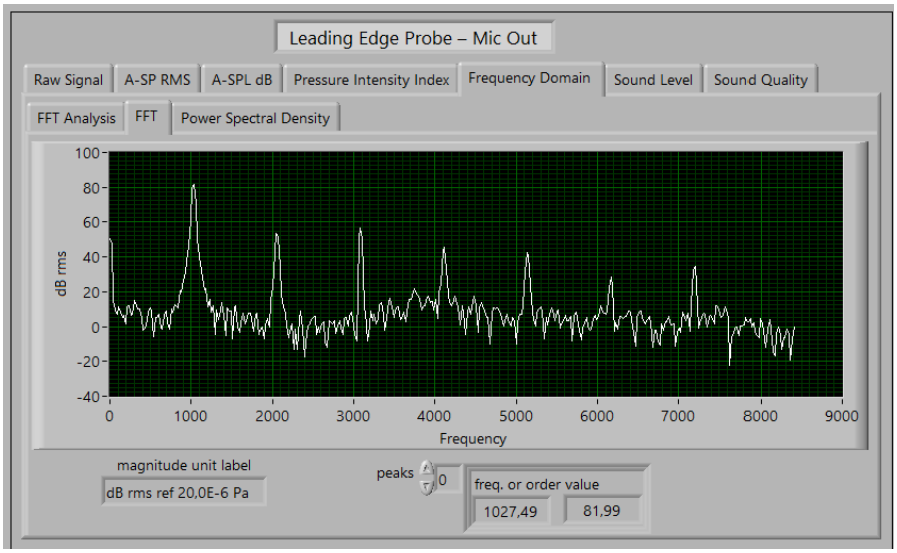


Figure 7.13: Subtab "FFT", from the tab Frequency Domain of the Leading Edge Probe Microphone Out,, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

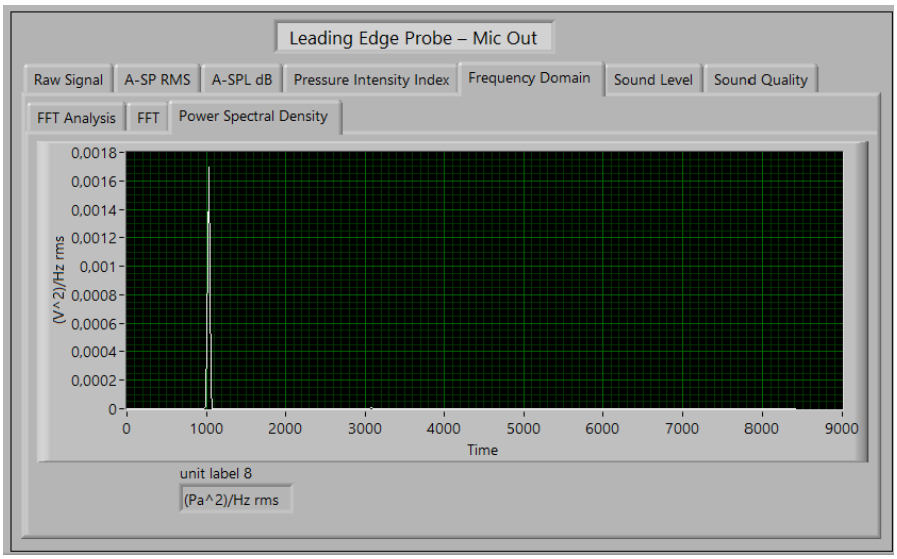


Figure 7.14: Subtab "Power Spectral Density", from the tab Frequency Domain of the Leading Edge Probe Microphone Out, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

Sound Level

This tab is divided into two additional subtabs: "Linear-Peak Leq", and "Exponential Leq".

Linear-Peak Leq Figure 7.15 represents the subtab "Linear-Peak Leq" part of the tab "Sound Level". Where the linear and peak hold equivalent continuous sound pressure level (L_{eq}) of the acquired signal is numerically displayed, as well as the units.

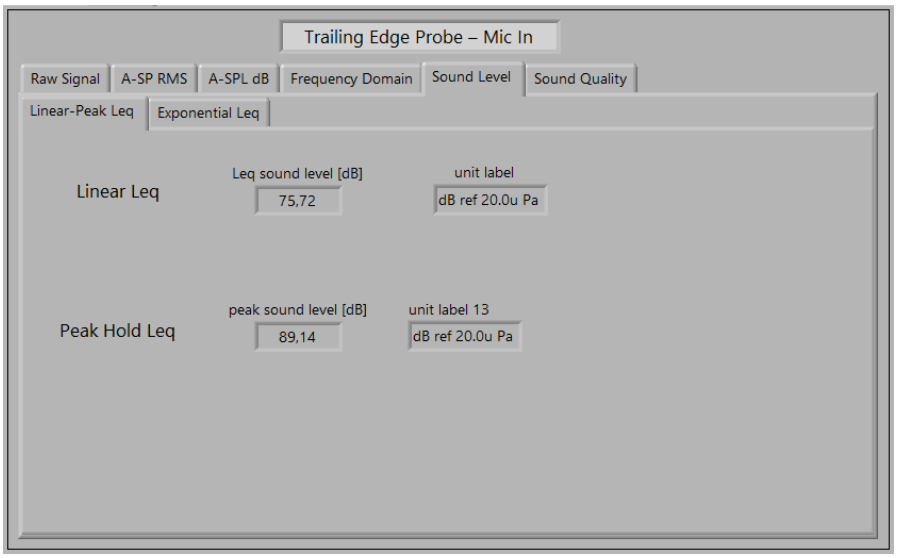


Figure 7.15: Subtab "Linear-Peak Leq", from the tab Sound Level of the Trailing Edge Probe Microphone In, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

Exponential Leq Figure 7.16 represents the subtab "Linear-Peak Leq" part of the tab "Sound Level". Where the exponential equivalent continuous sound pressure level (L_{eq}) of the acquired signal is displayed graphically (x axis time, while y axis sound level). Additionally, the maximum sound level (dB), the unit label and the actual time constant in milliseconds are shown.

Sound Quality

This tab is divided into three additional subtabs: "Loudness", "Aures Sharpness", and "Aures Roughness".

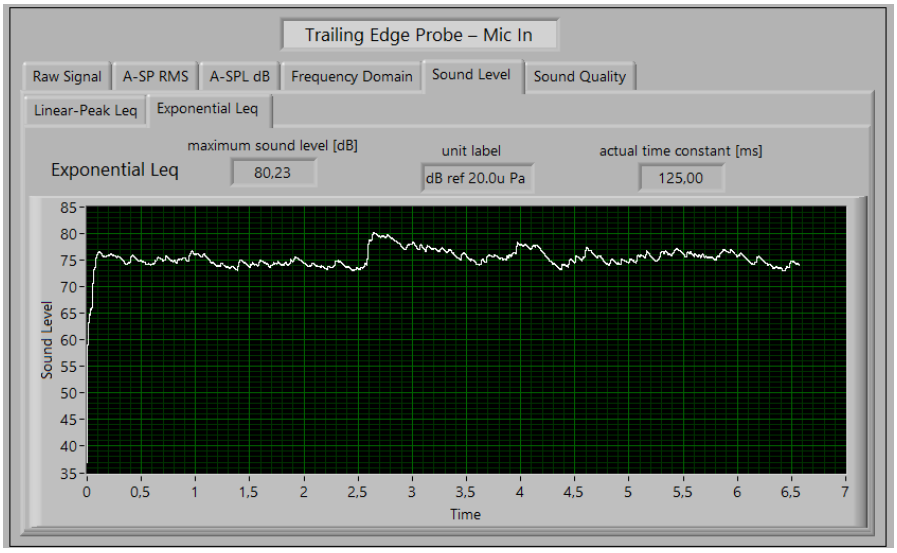


Figure 7.16: Subtab "Exponential Leq", from the tab Sound Level of the Trailing Edge Probe Microphone In, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

Loudness Figure 7.17 represents the subtab "Loudness" part of the tab "Sound Quality". Where the stationary loudness of the acquired signal is displayed graphically (x-axis critical band rate (Bark), while y-axis loudness (sones/Bark)). Furthermore, a numerical indicator of the graph is presented, which is divided into the critical band rate. Additionally, overall loudness (Sone) and loudness level (Phon) are shown.

Aures Sharpness Figure 7.18 represents the subtab "Aures Sharpness" part of the tab "Sound Quality". Where the sharpness of the acquired signal is displayed graphically (x-axis critical band rate (Bark), while y-axis specific sharpness). Furthermore, a numerical indicator of the graph is presented, which is divided into the critical band rate. Additionally, sharpness measurement in units of acum is shown.

Aures Roughness Figure 7.19 represents the subtab "Aures Roughness" part of the tab "Sound Quality". Where the roughness of the acquired signal is displayed graphically (x-axis critical band rate, while y-axis specific roughness). Furthermore, a numerical indicator of the graph is presented, which is divided into the critical band rate. Additionally, roughness measurement in units of asper is shown.

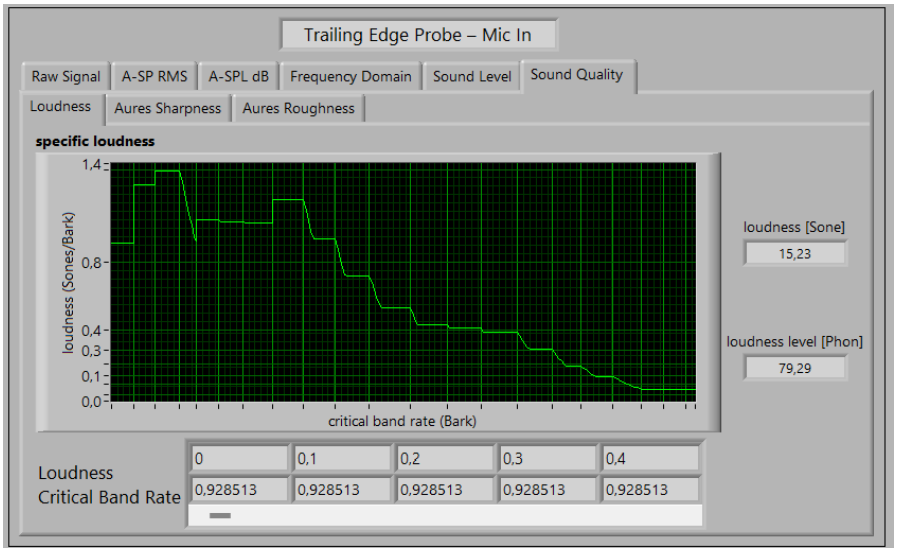


Figure 7.17: Subtab "Loudness", from the tab Sound Quality of the Trailing Edge Probe Microphone In, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

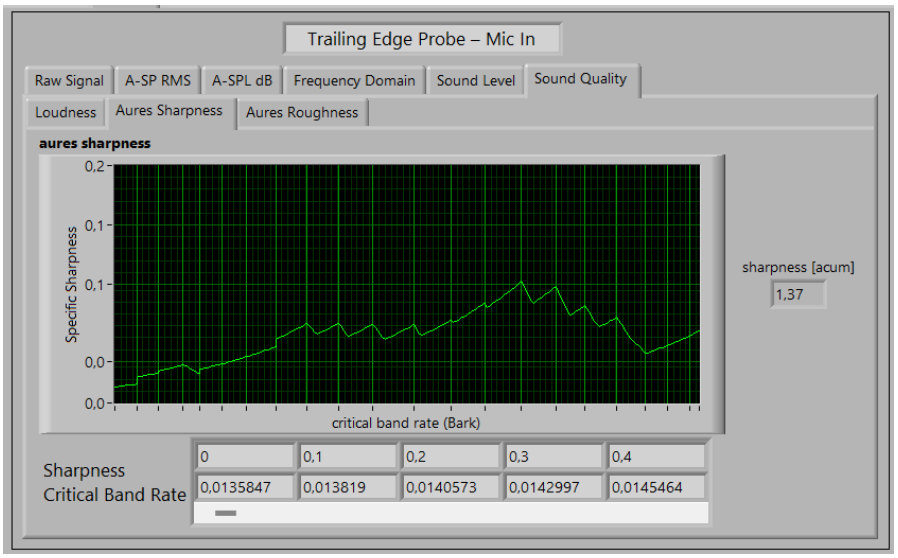


Figure 7.18: Subtab "Aures Sharpness", from the tab Sound Quality of the Trailing Edge Probe Microphone In, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

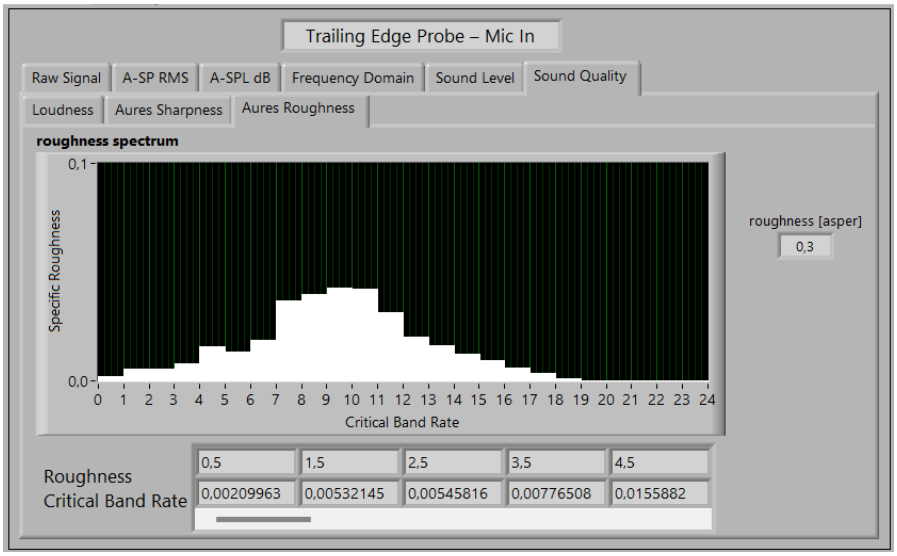


Figure 7.19: Subtab "Aures Roughness", from the tab Sound Quality of the Trailing Edge Probe Microphone In, Part of the Individual Microphone Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

7.2.2 Across-Probe and Overall Performance Analysis

Figure 7.20 depicts a representation of the across-probe and overall performance analysis section of the results tab of the OBSIDAS user interface. Where the across-probe and overall numerical results are displayed.

This section is divided into three parts:

- Leading Edge Probe, where across leading edge probe results are displayed.
- Trailing Edge Probe, where across trailing edge probe results are displayed.
- Overall, where the overall results, in other words, trial results are displayed.

Leading Edge Probe

Figure 7.21 represents the subtab where across leading edge probe results are displayed. The results shown are:

- Across leading edge probe arithmetic average SPL.
- Across leading edge OBSI level.

Trailing Edge Probe

Figure 7.22 represents the subtab where across trailing edge probe results are displayed. The results shown are:

- Across trailing edge probe arithmetic average SPL.
- Across trailing edge OBSI level.

Overall

Figure 7.23 represents the subtab where overall results are displayed. The results shown are:

- Energy averaged overall SPL.
- Energy averaged overall run test OBSI level.

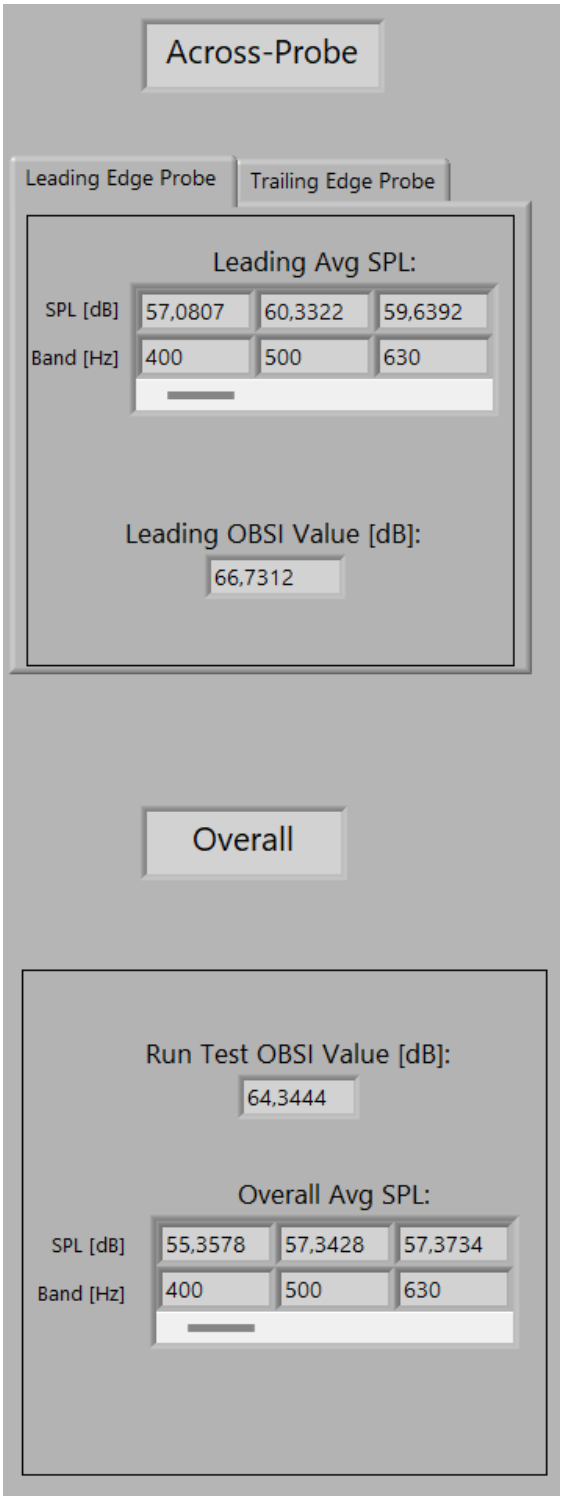


Figure 7.20: Across-Probe and Overall Performance Analysis section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

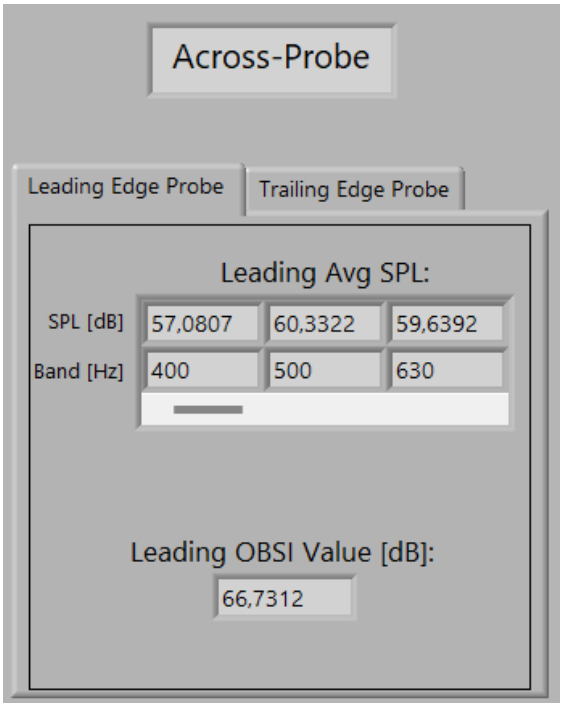


Figure 7.21: Subtab "Leading Edge Probe", Part of the Across-Probe and Overall Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

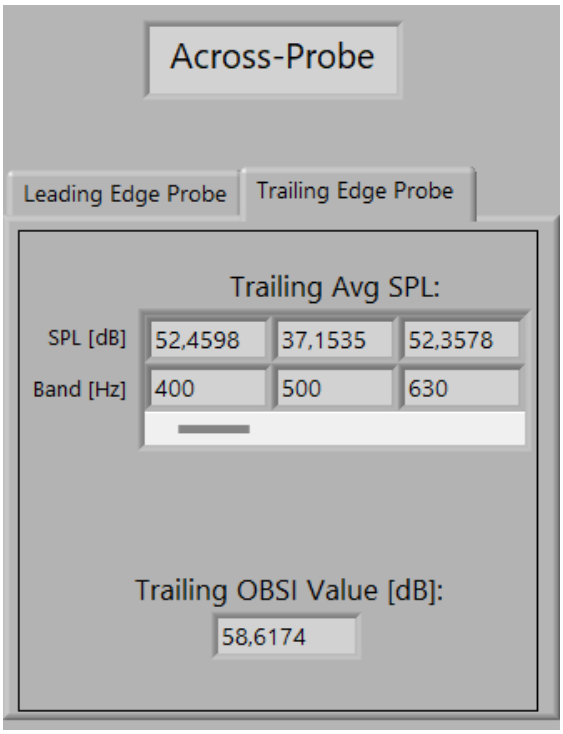


Figure 7.22: Subtab "Trailing Edge Probe", Part of the Across-Probe and Overall Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

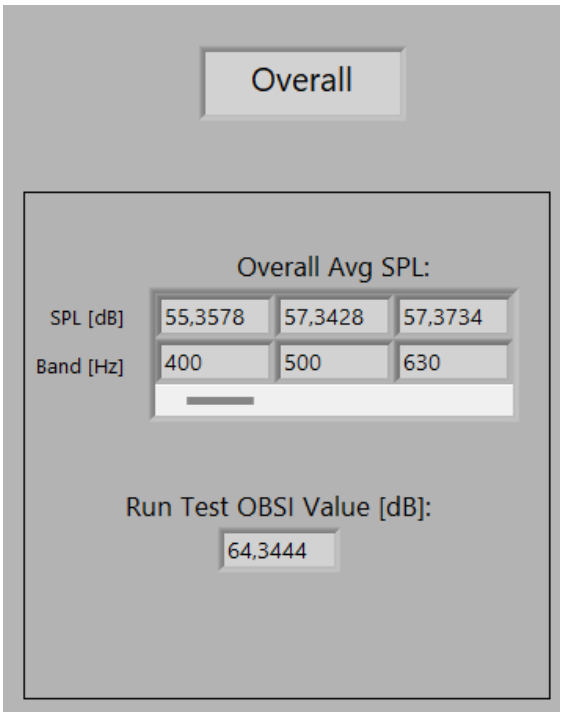


Figure 7.23: Subtab "Overall", Part of the Across-Probe and Overall Performance Analysis Section of the Results Tab of the OBSIDAS User Interface. Source: own elaboration

Chapter 8

Subdiagram 1: Idle While Loop

This chapter will present the block diagram of the first subdiagram, the Idle While Loop. As previously stated, this subdiagram represents the state in which the program will run indefinitely unless the user initiates a new trial. Following the completion of the pertinent trial process, the program will return to this state.

This block diagram consists of two main parts:

- Indicators Initialization
- While Loop

Each part will be explained in the following sections.

8.1 Indicators Initialization

Figure 8.1 depicts the block diagram of the initialization of these indicators. The indicators are string constants that will appear on the user interface to display the aforementioned string.

8.2 While Loop

Figure 8.2 depicts the block diagram of the while loop. This while loop is constantly checking the buttons "EXIT" and "Start New Trial". If one of these two buttons is pressed, the while loop will terminate, and the program will go on to the second subdiagram.

Additionally, the while loop checks the three menus from the user interface: "Exponential Mode", "Averaging Type one-third octave band", and "Application Mode".

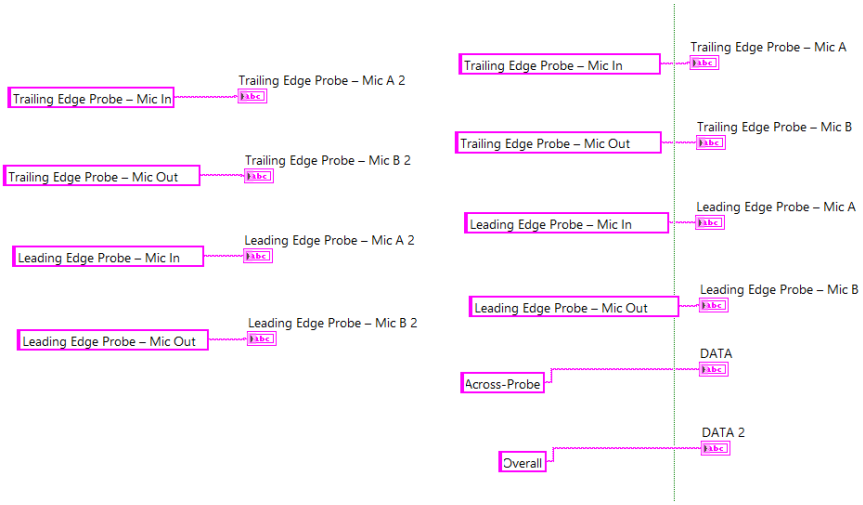


Figure 8.1: Subdiagram 1: Idle While Loop, Indicators Initialization Block Diagram. Source: own elaboration

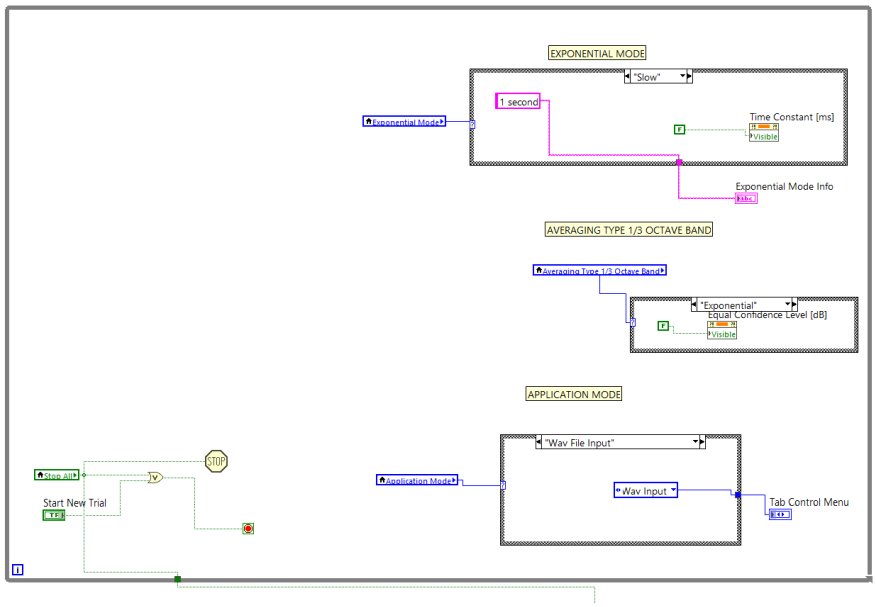


Figure 8.2: Subdiagram 1: Idle While Loop, While Loop Block Diagram. Source: own elaboration

8.2.1 Exponential Mode

This case structure checks the L_{eq} Exponential Mode menu, explained in section 7.1.1. The number of seconds is displayed, thought a string constant, in the block diagram according to the selected exponential mode. If the mode "Custom"

is selected, the "Time Constant" control is visible and changeable to the user; otherwise, it is hidden.

8.2.2 Averaging Type One-Third Octave Band

This case structure checks the Averaging Type One-Third Octave Band menu, explained in section 7.1.1. If the mode "Equal Confidence" is selected, the "Equal Confidence Level [dB]" control is visible and changeable to the user; otherwise, it is hidden.

8.2.3 Application Mode

This case structure checks the L_{eq} Exponential Mode menu, explained in section 7.1.1. If the mode "Real Device Acquisition" is selected, the "Channel Selection" menu will display four drop-down menus (thought a tab control that the tabs are hidden) that allow the user to select the physical device to be used for each of the microphones.

On the other hand, if the mode "Simulated Device Acquisition" is selected, the "Channel Selection" menu will display a drop-down menu that allow the user to select the simulated device to be used for the microphones.

The channel selection is implemented as a tab control, only two tabs present the drop-down channels. Consequently, the block diagram merely transitions between the tabs, despite the fact that the tabs themselves are not visible. This phenomenon gives the impression that the drop-down menus are appearing and disappearing.

Chapter 9

Subdiagram 2: Application Mode

his chapter will present the block diagram of the second subdiagram, the Selection of the Application Mode. As previously stated, this subdiagram represents the state in which the program will set up all the necessary components to initiate the acquisition process, including the creation of tasks, the initialization of constants, or the opening of files. This subdiagram consists of a case structure that depends on the Application Mode, resulting in four distinct possibilities (which could be seen in section 7.1.1).

9.1 Real Device Acquisition

In this mode, a real device, for example the National Instruments USB-4431 (explained in section 3.2.1) is utilized as the signal input. This is the mode in which the OBSI method will be performed if the hardware is fully functional. Figure 9.1 depicts the block diagram of the case structure if the real device acquisition is selected.

The program will proceed through the following steps and subVIs:

1. The program will read through the Channel Selection the names of the physical channels, where the device is connected. The program will automatically detect the device.

2. The Daqmx subVi "DAQmx Unflatten Channel String.vi" will convert these physical channels names into strings.

3. The four channels (each of the microphones) string names will be appended into one string.

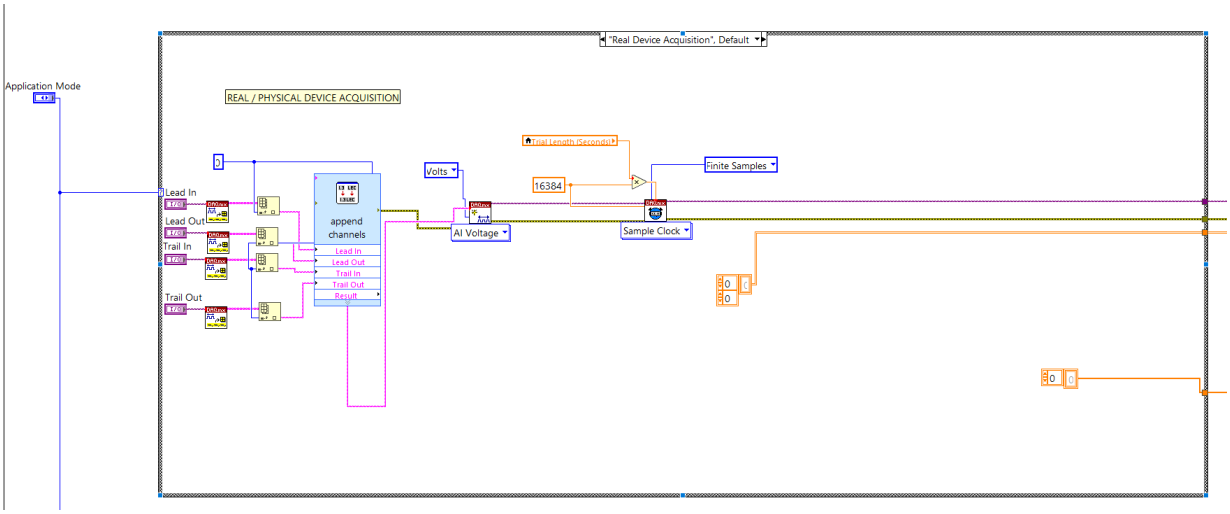


Figure 9.1: Subdiagram 2: Selection of the Application Mode, Real Device Acquisition Selected Block Diagram. Source: own elaboration

4. This appended string will be the input of the "DAQmx Create Virtual Channel.vi". Which will create a virtual channel, in other words, a DAQ task.

5. This task will be the input of the next DAQ subVI, "DAQmx Timing.vi". This subVI will also have the constant 16384 as the rate input (in Hz), and in the input samples per channel, the constant will be the trial length (as explained in section 7.1.1) times the rate frequency (16384 Hz). Additionally, the sample mode will be finite samples.

6. The output task and error signal will be passed outside the case structure.

9.2 Simulated Device Acquisition

This mode will be analogous to the "Real Device Acquisition" explained in the previous section (section 9.1). With the difference that in this mode the task has to be started manually. Figure 9.2 depicts the block diagram of the case structure if the simulated device acquisition is selected.

The program will proceed through the following steps and subVIs:

1. The program will read through the Channel Selection the simulated DAQmx task, where the signal will be simulated. This will create a task with the signal.

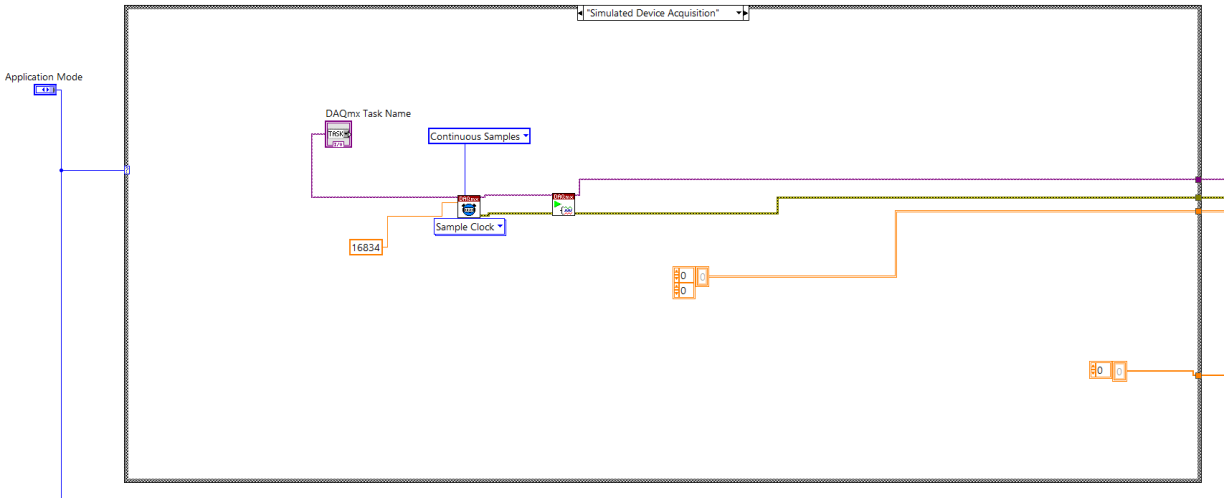


Figure 9.2: Subdiagram 2: Selection of the Application Mode, Simulated Device Acquisition Selected Block Diagram. Source: own elaboration

2. This task will be the input of the next DAQ subVI, "DAQmx Timing.vi". This subVI will also have the constant 16384 as the rate input (in Hz). Furthermore, the sample mode will be continuous samples, allowing the user to determine when to conclude this acquisition.

3. The output "DAQmx Timing.vi" task will be passed to the "DAQmx Start Task.vi" where the simulated task will be initialized.

4. The output task and error signal of the "DAQmx Start Task.vi" will be passed outside the case structure to the next subdiagram.

9.3 CSV File Input

In this mode, a CSV file input is utilized as tue acquired input. Figure 9.3 depicts the block diagram of the case structure if the CSV file acquisition is selected. It is notable that the program will require a CSV file created by the OBSIDAS program. Consequently, a trial can be conducted using a real device, after which the CSV file can be saved. The program will then be able to open the CSV file and replicate the original trial.

The program will proceed through the following steps and subVIs:

1. Firstly, the program will create a pop-up message to the user, which will be required to select the CSV file.

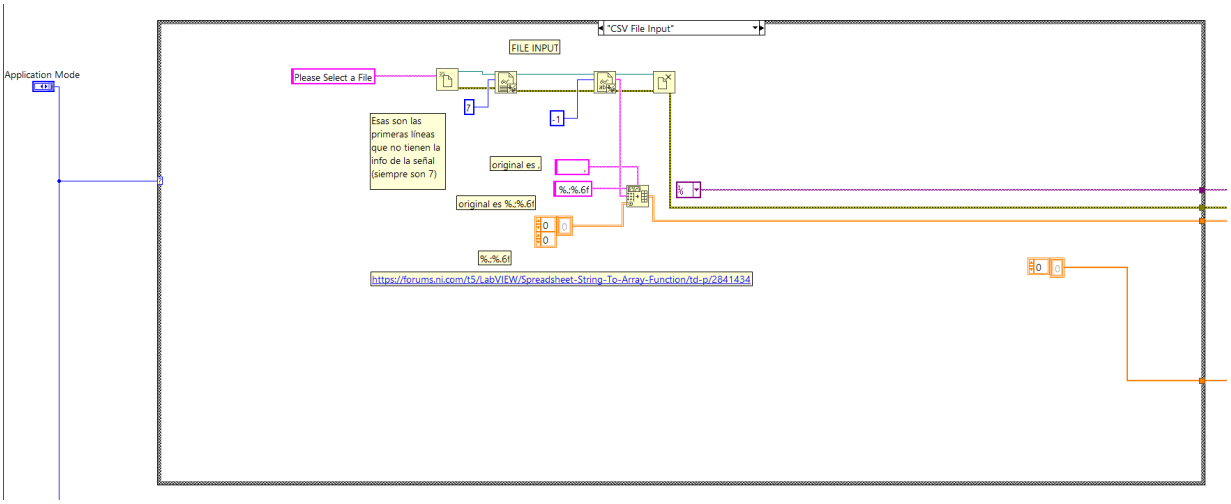


Figure 9.3: Subdiagram 2: Selection of the Application Mode, CSV File Input Acquisition Selected Block Diagram. Source: own elaboration

2. Once the user has selected this CSV file, the program will omit the first seven rows. As it is known that this first seven rows are only strings and data about the file and the trial, the signal data is from this row.

3. Subsequently, the program will proceed to read the remaining portion of the document, where the signal values are located. This CSV file contains six main columns:

   (a) Index column, where the values from 1 to n are numbered.

   (b) Time column, where the timestamp of the signal is represented.

   (c) Values leading edge probe microphone in.

   (d) Values leading edge probe microphone out.

   (e) Values trailing edge probe microphone in.

   (f) Values trailing edge probe microphone out.

4. The return value of the document will then be passed through the subVI: Spreadsheet String to Array Function. This will create a two-dimensional array consisting of the columns described in the previous step. The resulted array will be passed on to the next subdiagram.

5. The program will then proceed to close the CSV file used to read the values.

9.4 Wav File Input

This mode will be analogous to the "CSV File Input" explained in the previous section (section 9.3). In this mode, the signal is captured from a .wav file. Additionally, each microphone signal is stored in different and separate .wav files, so the program will ask the user to select four files in order to collect the values. Figure 9.4 depicts the block diagram of the case structure if the simulated device acquisition is selected.

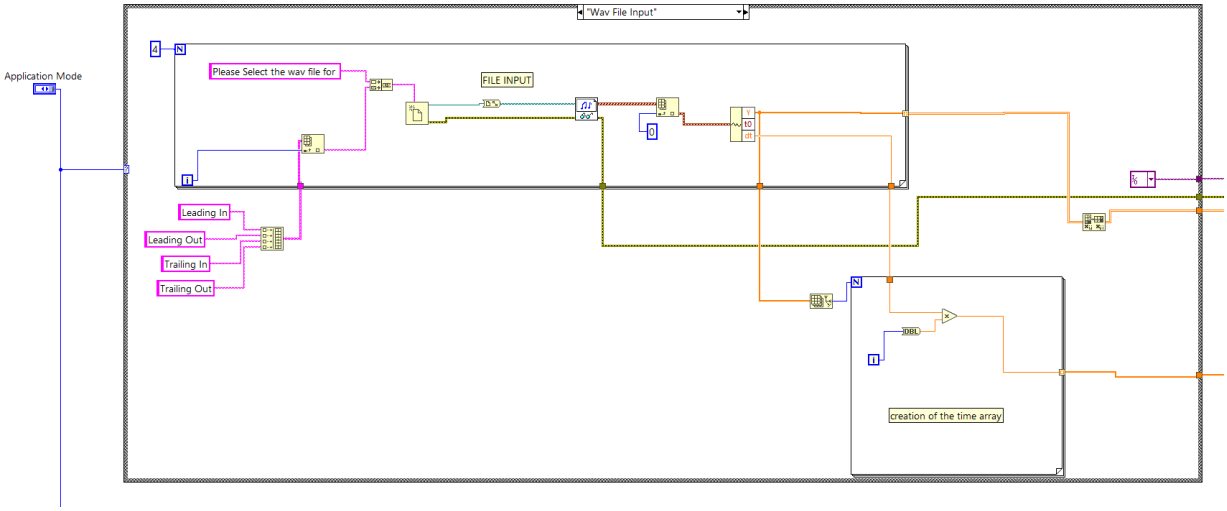


Figure 9.4: Subdiagram 2: Selection of the Application Mode, Wav File Input Acquisition Selected Block Diagram. Source: own elaboration

It should be noted that the primary structure of this mode is a for loop that is executed four times, once for each microphone. During this loop, the program prompts the user to select the .wav file, extract the values from the file, and create an array with the signal values. The program will proceed through the following steps and subVIs.

1. In the Foor loop:

   (a) Firstly, the program will generate a pop-up message prompting the user to select a WAV file for each microphone.

   (b) Once the user has selected this .wav type file, the program will extract the signal value (waveform) using the subVI: "Sound File Read Simple.vi".

   (c) The program will then obtain the waveform components: y, dt, and t0.

(d) The y component is transported outside the loop via an auto-indexed tunnel, which implies that in each iteration, this y component will be appended to an array.

(e) dt will be passed outside the loop through a normal connection.

2. Outside the Foor Loop, the array with the signal values will be transpose. As each microphone was distributed in rows, it is necessary to arrange them in columns.

3. Finally, the time array is created using a for loop, which iterates the size of the files. In each iteration, the loop multiplies the number of the iteration by dt. This array is then passed to the next subdiagram.

Chapter 10

Subdiagram 3: Timestamp

This chapter will present the block diagram of the third subdiagram, Timestamp. As previously stated, this subdiagram represents the state in which the program will capture the current timestamp to include in the signal processing. Figure 10.1 depicts the block diagram of this third subdiagram.

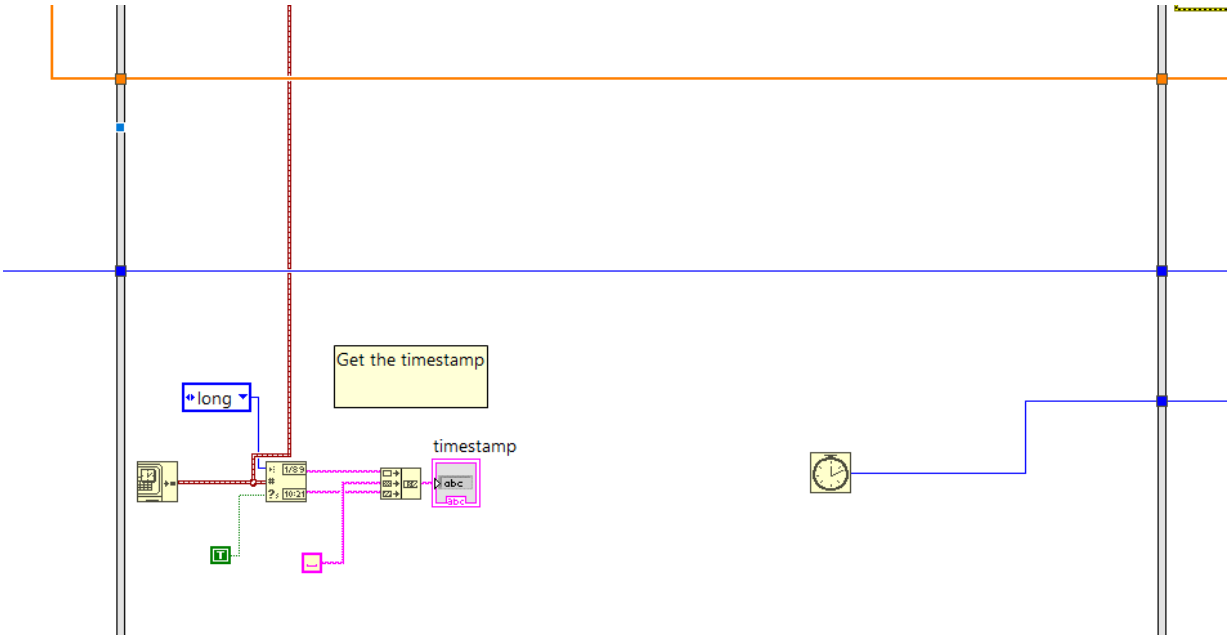


Figure 10.1: Subdiagram 3: Timestamp Block Diagram. Source: own elaboration

The program will proceed through the following steps and subVIs.

1. With the Get Date/Time In Seconds Function, the current time is captured.

2. Subsequently, this timestamp is converted into string and displayed to the user in the front panel of the user interface.

3. Note that there is a "Tick Count (msec)" function, which returns the value of the millisecond timer. This is done in order to compare the time elapsed in the following subdiagrams.

4. The time in timestamp units is passed to the next subdiagram.

It is important to note that every other signal that traverses this subdiagram will be transmitted to the subsequent one.

Chapter 11

Subdiagram 4: Acquisition

This chapter will present the block diagram of the fourth subdiagram, Acquisition. As previously stated, this subdiagram represents the state in which the program will acquire and graphically expose in real time to the user the signal. Figure 11.1 depicts the block diagram of this fourth subdiagram.

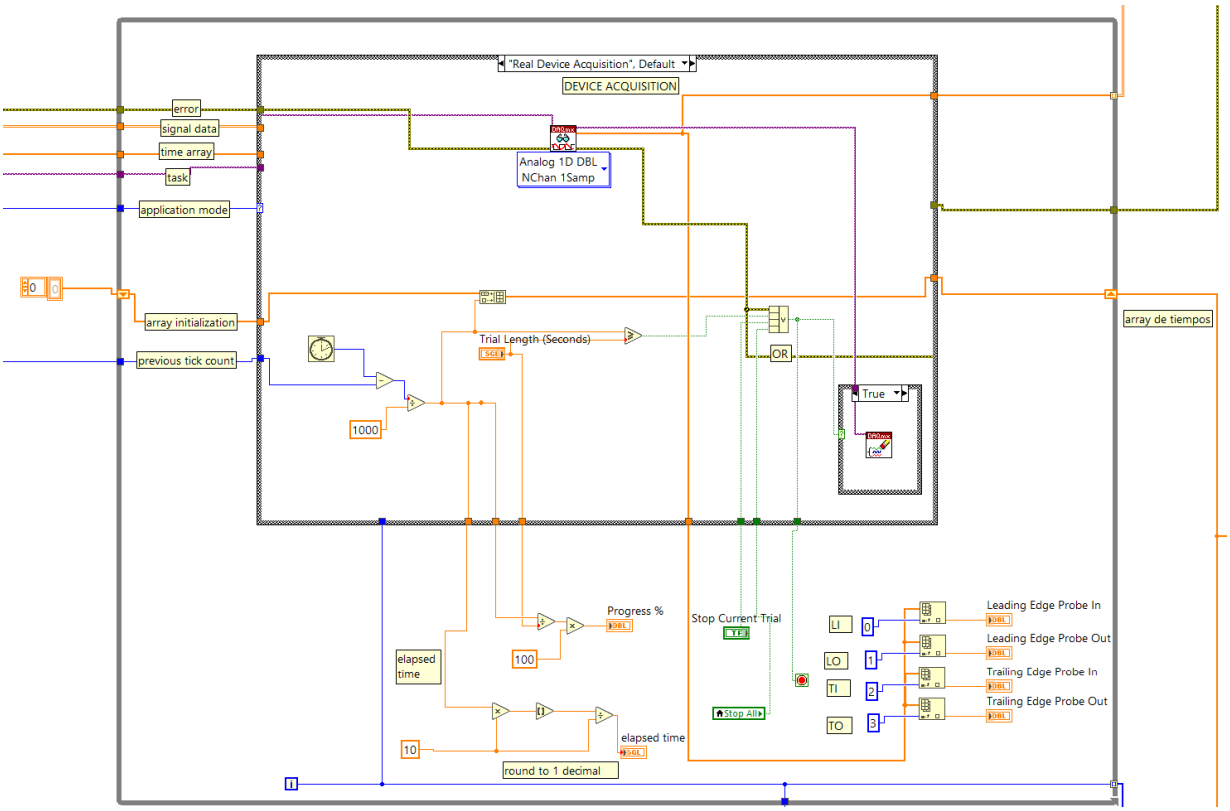


Figure 11.1: Subdiagram 4: Acquisition Block Diagram. Source: own elaboration

The subdiagram consists of a while loop, which will only terminate if any of the following actions occur:

1. An error
2. The "Stop Current Trial" button is pressed
3. The "Exit" button is pressed
4. The elapsed time is greater than or equal to the trial length selected by the user

This while loop is divided into two main sections:

1. Case Structure, which will acquire the signal depending on the application mode selected by the user.
2. Real Time Graphs, which will provides the user with a graphical representation of the signal being acquired by the case structure.

In the following sections, each part will be explained.

11.1 Case Structure

As displayed in the previous sections, this case structure will discern between the application mode. Each of which will be explained in the following sections.

11.1.1 Real Device Acquisition

Figure 11.2 depicts the block diagram of the case structure when the "Real Device Acquisition" mode is selected.

In this instance, the variables that will be incorporated into the structure as inputs are as follows:

- Error signal from subdiagram 2 selection of the application mode.
- Task signal from subdiagram 2 selection of the application mode.
- A blank initialized two dimensions array.
- Ticks from the subdiagram 3 timestamp.

The program will proceed through the following steps and subVIs.

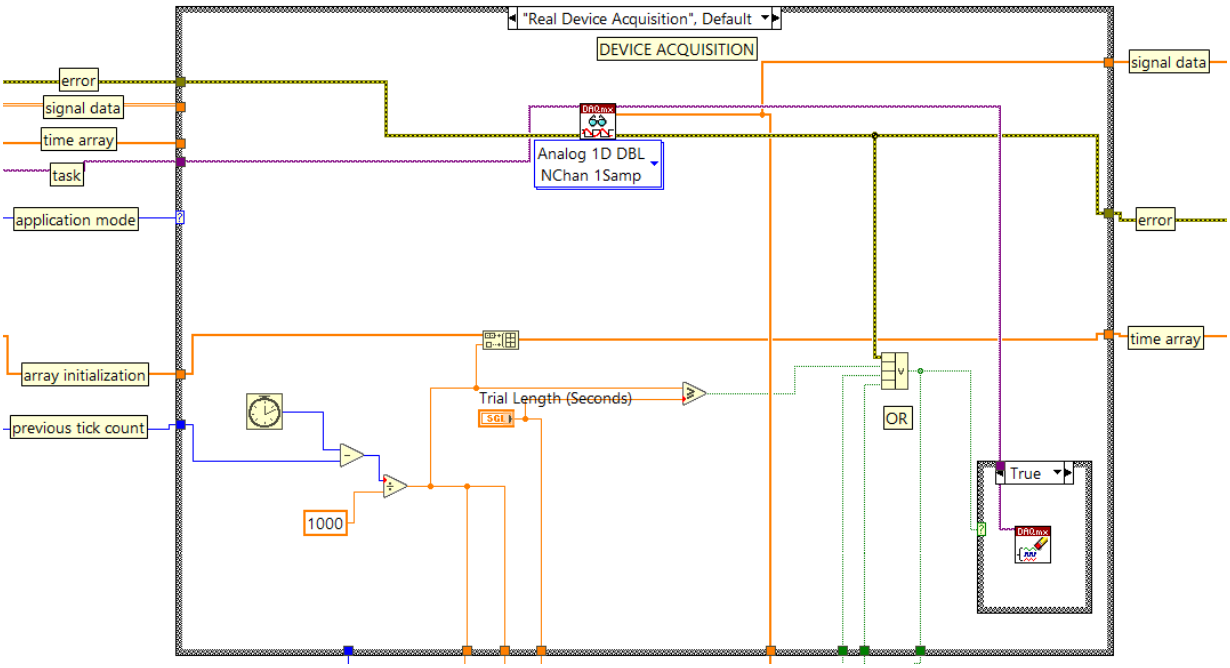


Figure 11.2: Subdiagram 4: Acquisition, Block Diagram of the Case Structure when Real Device Acquisition Mode Selected. Source: own elaboration

1. The task and error signal will be passed to the subVI "DAQmx Read.vi", where the signal will be read. This subVI must be configured as follows: analog, one-dimensional double, N channels, and 1 sample (it will acquire one sample per iteration). The subVI will return the acquired value of the four microphones, the error signal and the output task.

2. An analogous tick function will retrieve the milliseconds, which minus the value of the ticks from subdiagram 3 will lead into the trial elapsed time. This elapsed time will be compared to the trial length. Depending on this, the while loop will be terminated.

3. Each of the elapsed milliseconds will be appended to the blank initialized two dimensions array.

4. If the while loop is terminated, the output task will then be cleared by the subVI: "DAQmx Clear Task.vi".

This case structure will then yield the following outputs:

1. Error signal

2. Time one dimension array

3. One dimension array with one sample of signal from each microphone. It should be noted that the while loop has an auto-indexed array with this output, which means that in each iteration, each of the samples will be appended to the final two-dimensional array with the signals of the four microphones.

11.1.2 Simulated Device Acquisition

Figure 11.3 depicts the block diagram of the case structure when the "Simulated Device Acquisition" mode is selected.

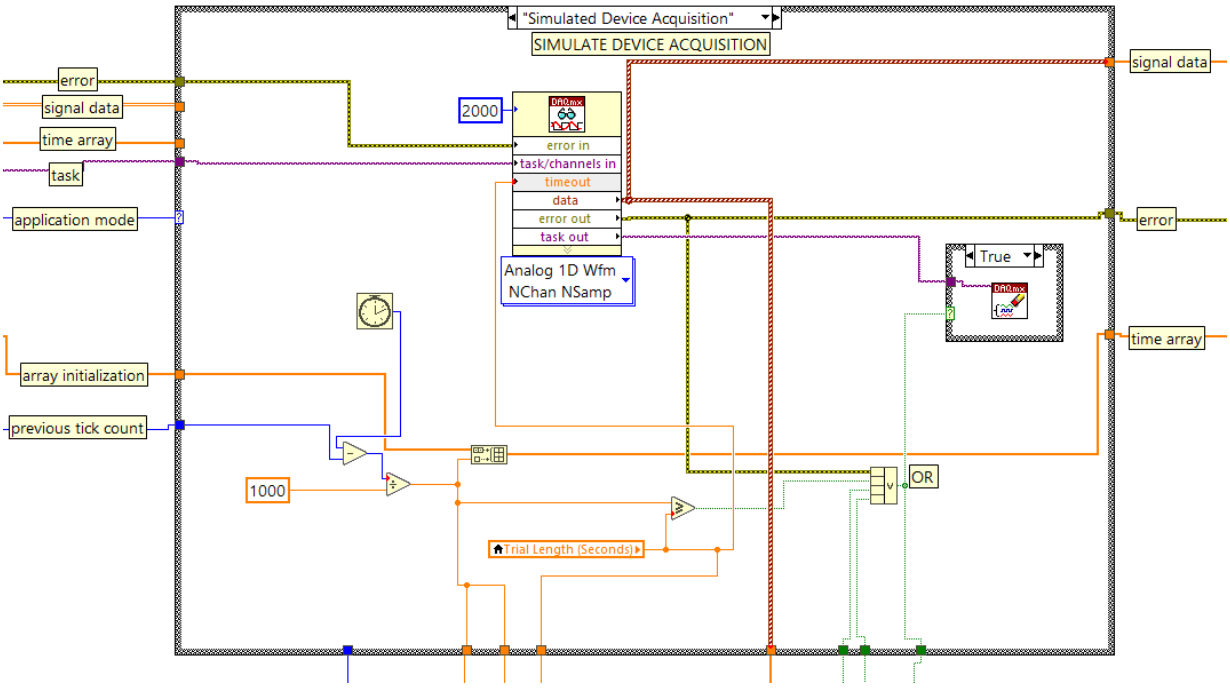


Figure 11.3: Subdiagram 4: Acquisition, Block Diagram of the Case Structure when Simulated Device Acquisition Mode Selected. Source: own elaboration

In this instance, the variables that will be incorporated into the structure as inputs are as follows:

- Error signal from subdiagram 2 selection of the application mode.
- Task signal from subdiagram 2 selection of the application mode.

- A blank initialized two dimensions array.
- Ticks from the subdiagram 3 timestamp.

The program will proceed through the following steps and subVIs.

1. The task and error signal will be passed to the subVI "DAQmx Read.vi", where the signal will be read. This subVI must be configured as follows: analog, one-dimensional waveform, N channels, and N sample (it will acquire n sample per iteration), 2000 number of samples per channel will be specified, and the timeout value, which is the trial length. The subVI will return the acquired value of the four microphones (waveform), the error signal and the output task.
2. An analogous tick function will retrieve the milliseconds, which minus the value of the ticks from subdiagram 3 will lead into the trial elapsed time. This elapsed time will be compared to the trial length. Depending on this, the while loop will be terminated.
3. Each of the elapsed milliseconds will be appended to the blank initialized two dimensions array.
4. If the while loop is terminated, the output task will then be cleared by the subVI: "DAQmx Clear Task.vi".

This case structure will then yield the following outputs:

1. Error signal
2. Time one dimension array
3. One dimension array with one sample of signal from each microphone. It should be noted that the while loop has an auto-indexed array with this output, which means that in each iteration, each of the samples will be appended to the final two-dimensional array with the signals of the four microphones.

11.1.3 CSV File Input

Figure 11.4 depicts the block diagram of the case structure when the "Simulated Device Acquisition" mode is selected.

In this instance, the variables that will be incorporated into the structure as inputs are as follows:

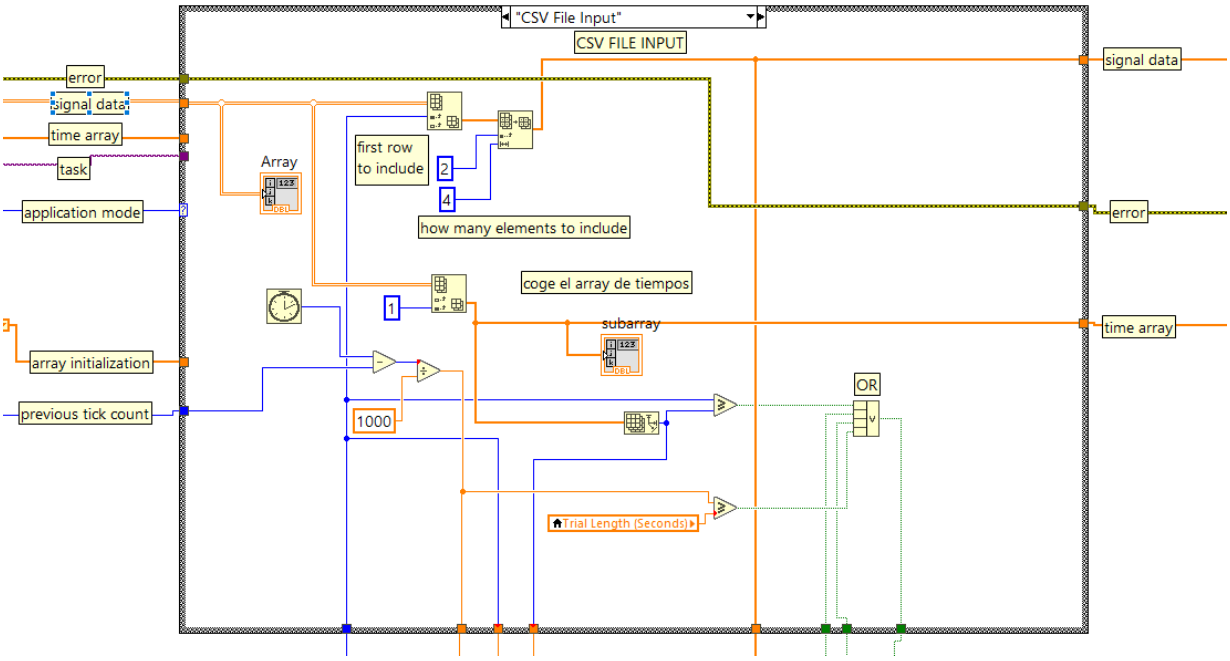


Figure 11.4: Subdiagram 4: Acquisition, Block Diagram of the Case Structure when CSV File Input Mode Selected. Source: own elaboration

- Error signal from subdiagram 2 selection of the application mode.
- Signal data from subdiagram 2 selection of the application mode.
- Ticks from the subdiagram 3 timestamp.

The program will proceed through the following steps and subVIs.

1. The error will be passed through the case structure and will be outputted, but will not be utilized.
2. The number of iterations of the while loop is used to access the pertinent row of the signal data. In this row, only positions 2 to 4 are computed (positions in which the microphone data are located). An array is created only with this row and these positions. This one-dimensional array with the four values of the microphone data at that time is one of the outputs of the case structure.
3. Similarly to the cases before, an analogous tick function will retrieve the milliseconds, which minus the value of the ticks from subdiagram 3 will lead into the trial elapsed time. This elapsed time will be compared to the trial length. Depending on this, the while loop will be terminated.

4. However, in this case, the time array will be the first column of the case input data array.

This case structure will then yield the following outputs:

1. Error signal

2. Time one dimension array.

3. One dimension array with one sample of signal from each microphone. It should be noted that the while loop has an auto-indexed array with this output, which means that in each iteration, each of the samples will be appended to the final two-dimensional array with the signals of the four microphones.

11.1.4 Wav File Input

Figure 11.5 depicts the block diagram of the case structure when the "Simulated Device Acquisition" mode is selected.

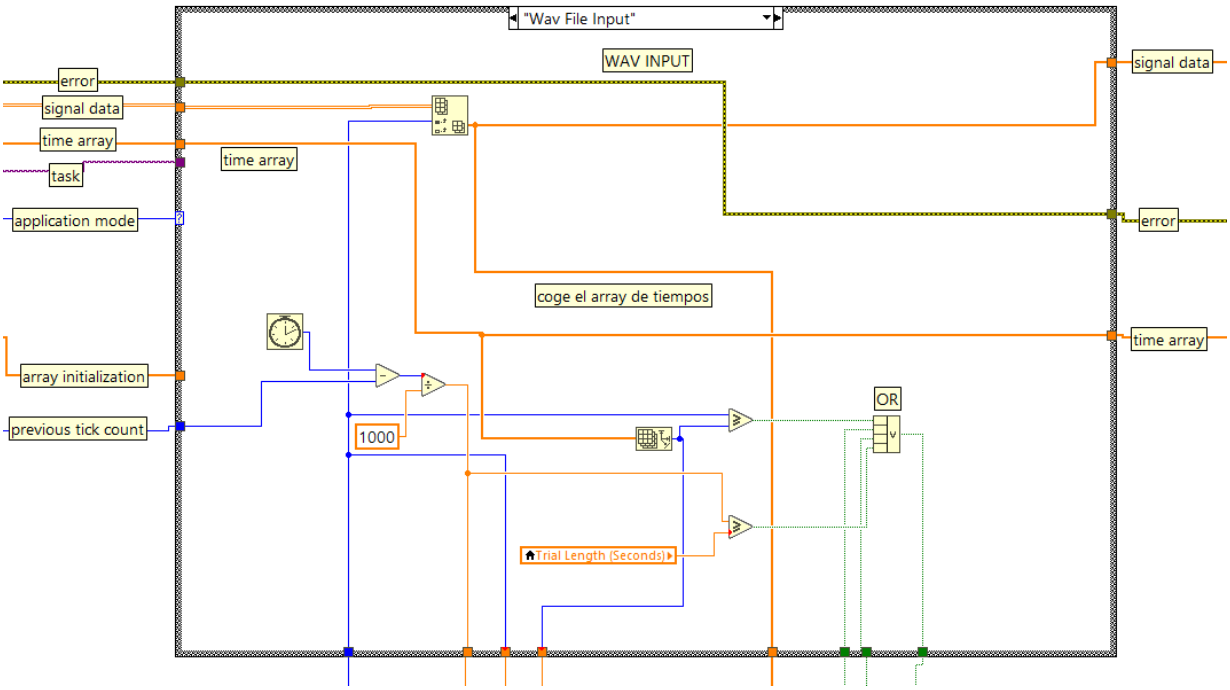


Figure 11.5: Subdiagram 4: Acquisition, Block Diagram of the Case Structure when Wav File Input Mode Selected. Source: own elaboration

In this instance, the variables that will be incorporated into the structure as inputs are as follows:

- Error signal from subdiagram 2 selection of the application mode.
- Signal data from subdiagram 2 selection of the application mode.
- Ticks from the subdiagram 3 timestamp.
- Time one dimension array.

The program will proceed through the following steps and subVIs.

1. The error will be passed through the case structure and will be outputted, but will not be utilized.
2. The number of iterations of the while loop is used to access the appropriate row of signal data. In this case, unlike the previous case, every element of the row is used. In this case, the time array has been created separately. An array is created only with the corresponding row. This one-dimensional array with the four values of the microphone data at this time is one of the outputs of the case structure.
3. Similarly to the cases before, an analogous tick function will retrieve the milliseconds, which minus the value of the ticks from subdiagram 3 will lead into the trial elapsed time. This elapsed time will be compared to the trial length. Depending on this, the while loop will be terminated.
4. However, in this case, the time array will be the output array from subdiagram 2 selection of the application mode.

This case structure will then yield the following outputs:

1. Error signal
2. Time one dimension array (same to the one created in the subdiagram 2 selection of the application mode).
3. One dimension array with one sample of signal from each microphone. It should be noted that the while loop has an auto-indexed array with this output, which means that in each iteration, each of the samples will be appended to the final two-dimensional array with the signals of the four microphones.

11.2 Real Time Graphs

This section outlines the procedure for displaying the real-time graph of each microphone as it is being acquired, refer to figure 11.6 for a representation of the block diagram of this procedure.

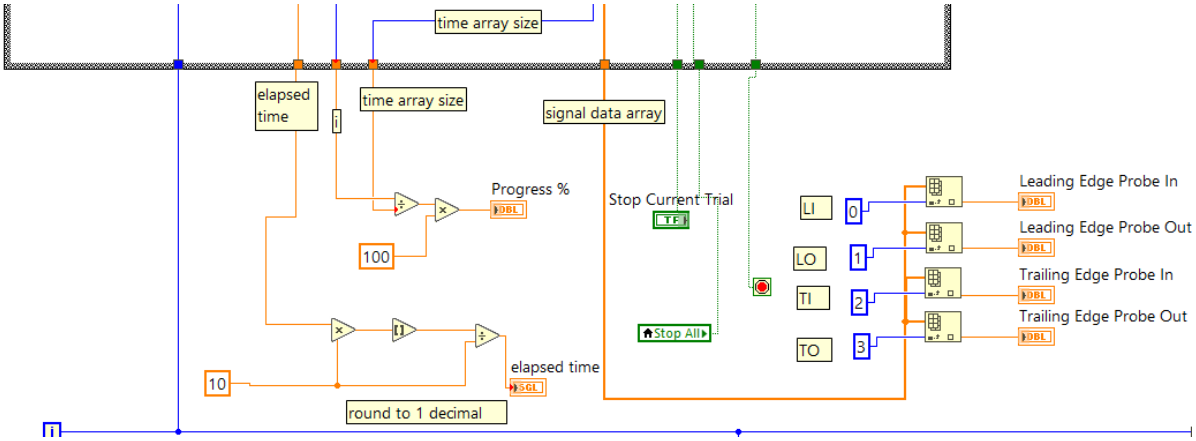


Figure 11.6: Subdiagram 4: Acquisition, Block Diagram of the Real Time Graphs Procedure. Source: own elaboration

The program will proceed through the following steps and subVIs.

1. The elapsed time computed in the case structure will be utilized. It will be rounded to one decimal place and presented visually in the user interface.

2. Furthermore, the time array from the case structure will be utilized. Its size will be calculated. The number of iterations the while loop is in divided by this size will yield the progress percentage, which will be displayed in the user interface.

3. Finally, the signal data array will be divided into the four microphones. Consequently, each of the four values will be obtained and graphed into the four graphs displayed in the user interface. This process will be repeated for each iteration, resulting in the display of a chart for each signal value (as presented in figure 7.6).

Chapter 12

Subdiagram 5: Signal Division and Errors

This chapter will present the block diagram of the fifth subdiagram, Signal Division and Errors. As previously stated, this subdiagram represents the state in which the program will split the whole signal array into the four different microphones array, handle the error messages (if any), and obtain the temperature through an API query. This chapter will be divided into these three parts.

In this subdiagram, the variables that will be incorporated into the diagram as inputs are as follows:

- Error signal from subdiagram 4 Acquisition.
- Signal data two dimension array from 4 Acquisition.
- Timestamp from subdiagram 3 Timestamp.

12.1 Signal Division

The output of the preceding subdiagram is a two-dimensional array, in which each of the microphone is represented by a column and each value is represented by a row. Consequently, a division of the columns is necessary in order to obtain a column array comprising only one microphone's values. Figure 12.1 depicts the block diagram of the signal division procedure from the subdiagram 5.

The program will proceed through the following steps and subVIs.

1. The signal data array will be divided into four one-dimensional sub-arrays. Each column will be added to a new array, resulting in the data from each microphone being stored in a separate array.

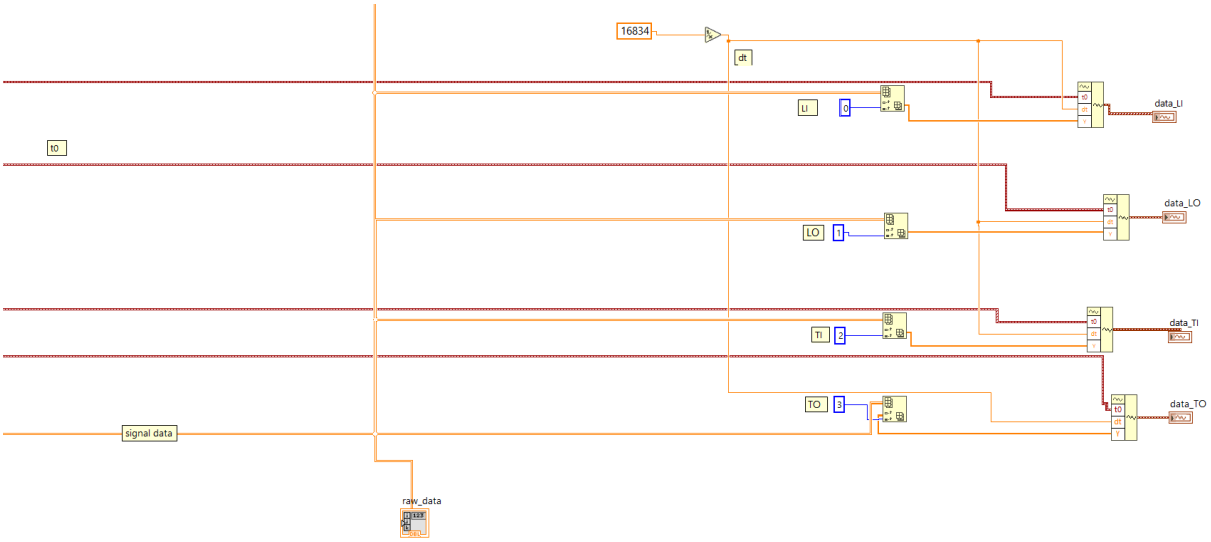


Figure 12.1: Subdiagram 5: Signal Division, Error Handling, and Temperature API, Block Diagram of the Signal Division. Source: own elaboration

2. A waveform for each microphone will be created. The input timestamp will be used as t_0 of the waveform. The elements or values will be represented by the constructed column array. And the dt will be 1/16834, as this is the rate at which the program will operate (16834 Hz).

3. Four local variables will be created from the new waveforms:

   - data_LI
   - data_LO
   - data_TI
   - data_TO

12.2 Error Handling

In this section is where every error is handled. This is of paramount importance for any software developed program, because if this program detects an error, it will continue to run until this section is reached, and the error message will be displayed to the user. It was decided to establish the error detection at this point of the program, because it is the first section after the acquisition of the signal, which is the most delicate part of the system, so if an error occurs, the user will know it as soon as possible. Figure 12.2 depicts the block diagram of the signal division procedure from the subdiagram 5.

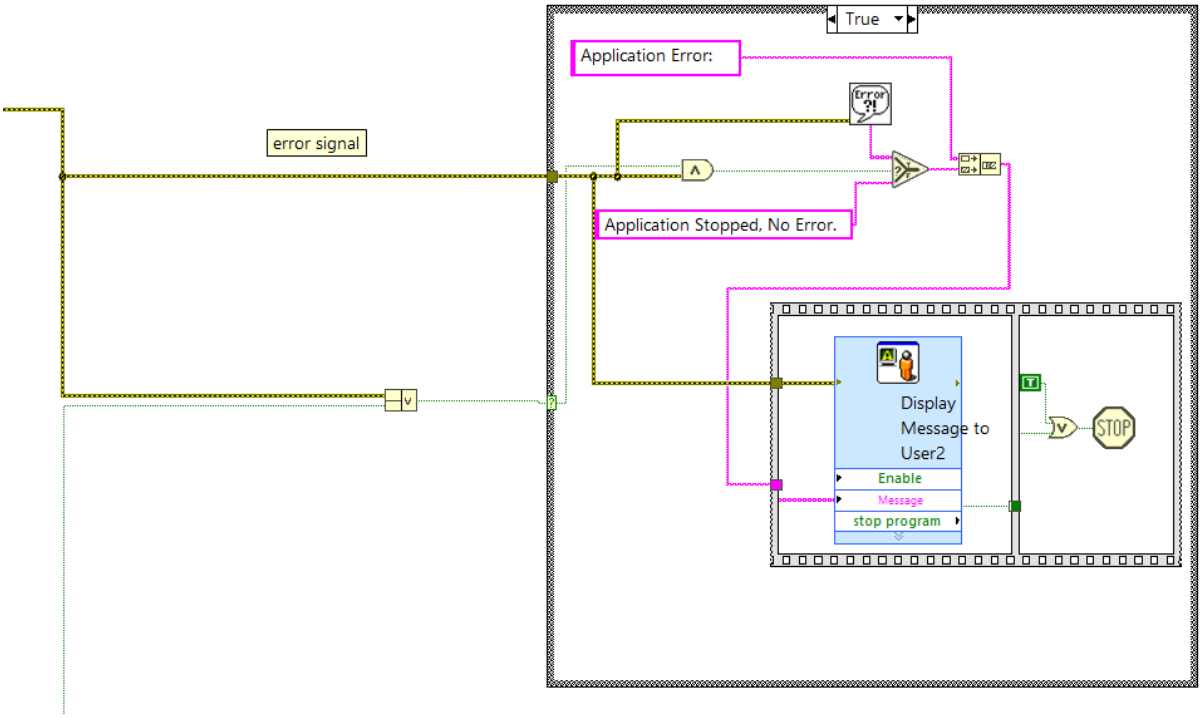


Figure 12.2: Subdiagram 5: Signal Division, Error Handling, and Temperature API, Block Diagram of the Error Handling. Source: own elaboration

The program will proceed through the following steps and subVIs.

1. The error will go through a case structure. If there is an error, the signal will be "TRUE".
2. The error handler will then check the nature of the error and display an error message to the user indicating what caused the program to stop.
3. Immediately after this, the program will stop as if the user had pressed the "EXIT" button.

12.3 Temperature API

In this part, a query request is sent to a meteorological API in order to get the current temperature. So the sound pressure could be normalized, as explained in section 2.5.3. Figure 12.3 depicts the block diagram of the temperature API procedure from the subdiagram 5.

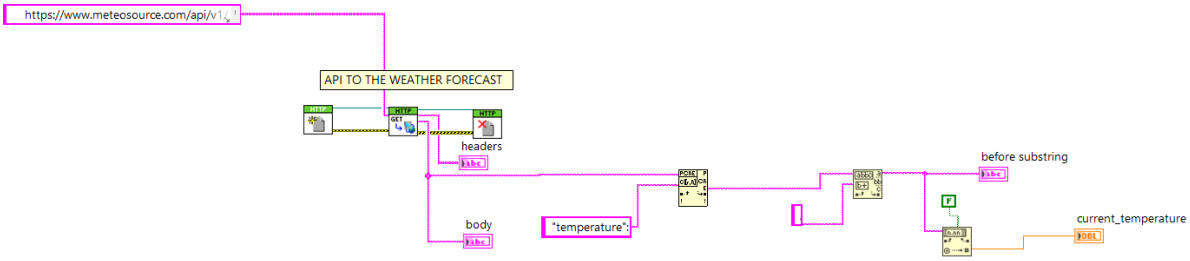


Figure 12.3: Subdiagram 5: Signal Division, Error Handling, and Temperature API; Block Diagram of the Temperature API. Source: own elaboration

Along this procedure, the LabVIEW "HTTP Client" functions palette will be used. Note that the API used is from the site `meteosource.com` reference [83]. Figure 12.4 depicts an example of the response from this API program.

{"lat":"40.4165N","lon":"3.70256W","elevation":665,"timezone":"Europe/Madrid","units":"metric","current":
{"icon":"partly_sunny","icon_num":4,"summary":"Partly sunny","temperature":28.2,"wind":
{"speed":3.2,"angle":223,"dir":"SW"},"precipitation":
{"total":0.0,"type":"none"},"cloud_cover":23},"hourly":null,"daily":null}

Figure 12.4: Response example from the Meteosource Temperature API. Source: [83]

In order to obtain the temperature in LabVIEW, the program will proceed through the following steps and subVIs.

1. The program will open a client handler using the "OpenHandle.vi".
2. Subsequently, the client out signal is connected to the function "GET HTTP", where the API URL is incorporated. This subVi returns the headers and the body of the API response.
3. Finally, the client handler will be closed using the "Close Handle.vi".
4. The response from the API will be formatted in such a way that only the temperature in Celsius is obtained. This value will then be added as a local variable so that it can be utilized by the subsequent subdiagramsa.

Chapter 13

Subdiagram 6: Signal Processing

This chapter presents the block diagram of the sixth subdiagram, Signal Processing. As previously stated, this subdiagram represents the state in which the program will compute the necessary signal processing in order to obtain the required values of the AASHTO T360 standard, as well as the additional computations performed by OBSIDAS, which were described in section5.2. This diagram block is replicated four times, one per each microphone. Consequently, this section will present only one of the analogous microphone's computations. All the performed analyses are displayed in the result tab of the user interface. This chapter is divided into the different computations.

13.1 Raw Signal

In this subdiagram the local variable data_micr created in the previous subdiagram is used as the signal source for the computations. Figure 13.1 represents the block diagram for displaying the raw signal graph in the user interface. The element used is a Waveform Graph from LabVIEW.

13.2 Frequency Domain Computations

Figure 13.2 depicts the block diagram of the frequency domain computations.

The computations performed by this block diagram, along with the specific steps and subVIs required to perform them, are explained in the following sections.



Figure 13.1: Subdiagram 6: Signal Processing, Block Diagram of the Raw Signal Graphical Display. Source: own elaboration.

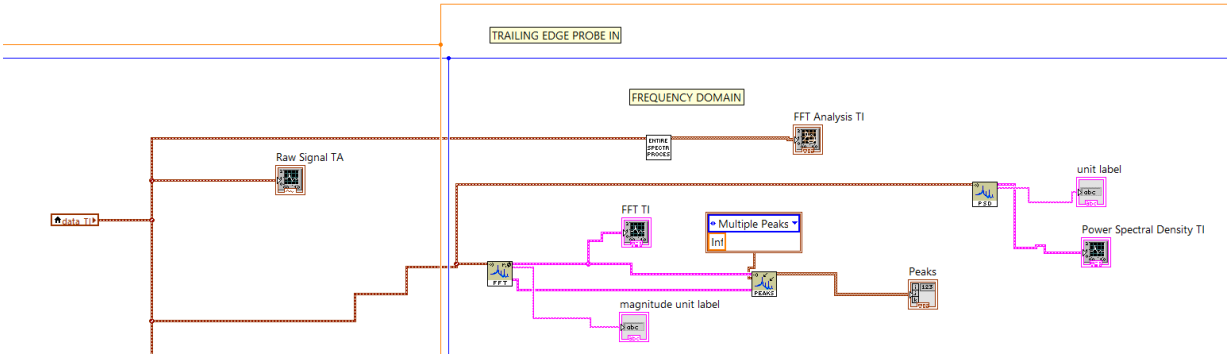


Figure 13.2: Subdiagram 6: Signal Processing, Block Diagram of the Frequency Domain Computations. Source: own elaboration

13.2.1 FFT

OBSIDAS obtains the signal FFT through two different methods: LabVIEW native computations, and using the Sound and Vibration Toolkit VIs.

FFT Using LabVIEW Native

In order to obtain the signal FFT a subVI was created. Figure 13.3 represent this process. The subVIs name is: "Entire Spectrum Process". The input is the signal and the output the FFT of such signal.

FFT Using Sound and Vibration Toolkit VIs

On the other hand, other way of performing the FFT is using the Sound and Vibration Toolkit VI: "SVFA FFT Spectrum (Mag-Phase).vi", as displayed in figure 11.5.

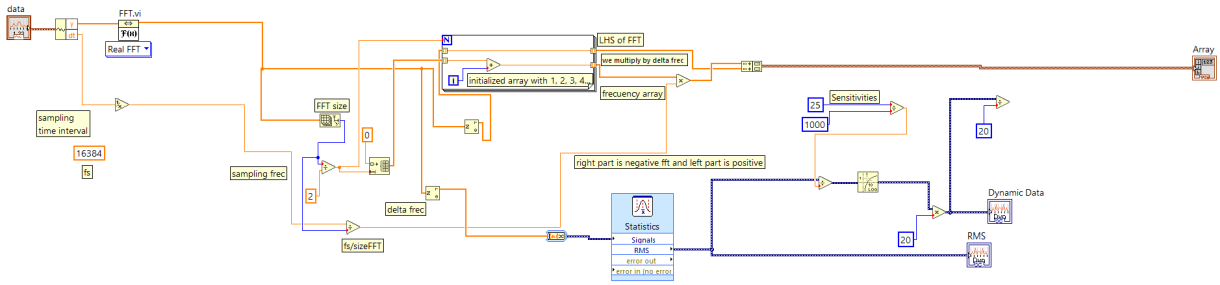


Figure 13.3: Subdiagram 6: Signal Processing, Block Diagram of the LabVIEW Native FFT Computation. Source: own elaboration

13.2.2 Power spectral density

In order to obtain the power spectral density, the program proceeded through the Sound and Vibration Toolkit VI: "SVFA Power Spectral Density.vi". The output of this VI is the waveform graph where the value is shown and the units label.

13.2.3 FFT peaks

In order to obtain the FFT peaks, the program proceeded through the Sound and Vibration Toolkit VI: "SVFA Spectrum Peak Search.vi". These peaks will be computed from the Sound and Vibration FFT.

13.3 Single Microphone Sound Pressure Level Computations

Figure 13.4 depicts the block diagram of the single microphone sound pressure level computations.

The computations performed by this block diagram, along with the specific steps and subVIs required to perform them, are explained in the following sections.

13.3.1 Microphone Calibration

In order to calibrate the microphones, the program proceeded through the Sound and Vibration Toolkit VI: "SVL Scale Voltage to EU VI .vi". The input of this subVI is the microphone data signal. The output of this VI is the waveform calibrated signal. This subVI uses the information from the calibration settings of the front panel of the user interface.

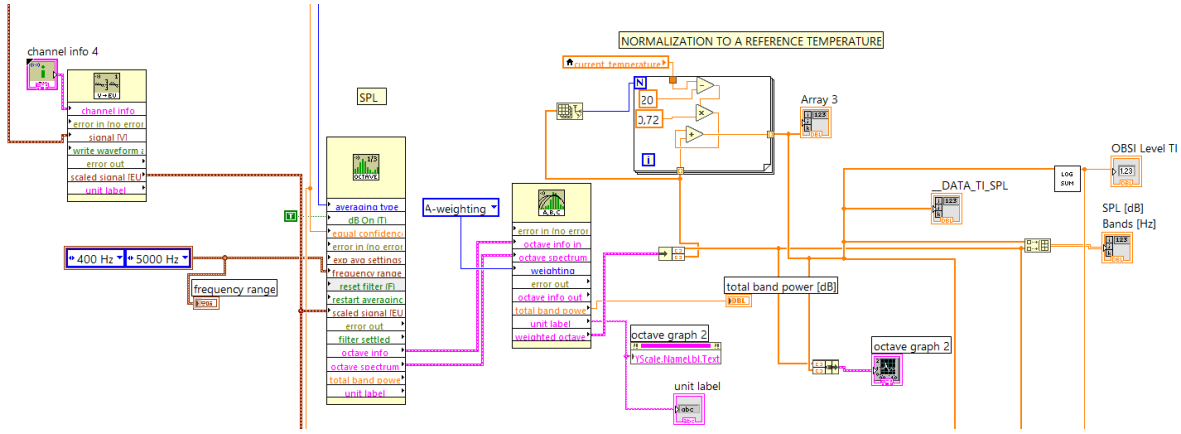


Figure 13.4: Subdiagram 6: Signal Processing, Block Diagram of the Single Microphone Sound Pressure Level Computations. Source: own elaboration

13.3.2 Normalized To a Reference Temperature A-weighted One-Third Octave Band Sound Pressure Level

In order to obtain this value, the program proceeded through the following Sound and Vibration Toolkit VIs: "SVT Third-octave Analysis VI.vi", and "SVT Weighting Filter (octave) VI.vi". The input of these subVIs is the microphone data signal. The output of the last VI is the A-weighted One-Third Octave Band Sound Pressure Level. This value will then be normalized through the equation provided in the standard.

13.3.3 A-Weighted One-Third Octave Band Sound Pressure Level Total Band Power

In order to obtain this value, the program proceeded through the following Sound and Vibration Toolkit VIs: "SVT Third-octave Analysis VI.vi", and "SVT Weighting Filter (octave) VI.vi". The input of these subVIs is the microphone data signal. The output of the last VI is the A-Weighted One-Third Octave Band Sound Pressure Level Total Band Power.

13.3.4 Individual OBSI Level For Each Microphone

In order to obtain the Individual OBSI Level For Each Microphone FFT a subVI was created. Figure 13.5 represents this process. The subVIs name is: "Logarithm Sum". The input is the Normalized To a Reference Temperature A-weighted One-Third Octave Band Sound Pressure Level and the output Individual OBSI Level For Each Microphone. This subVI logarithmically summarize the SPLs.

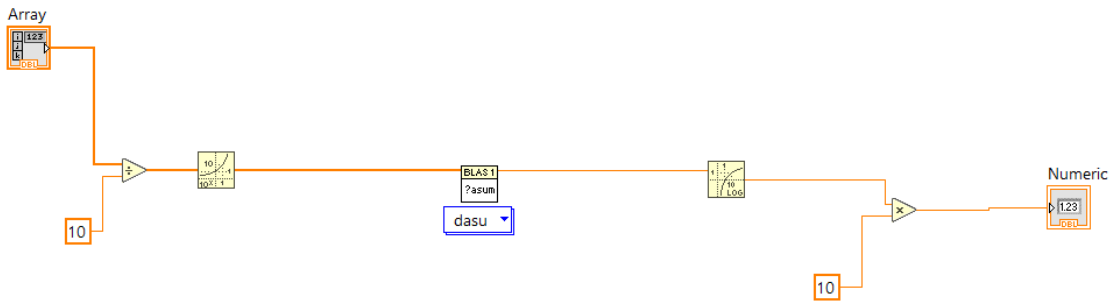


Figure 13.5: Subdiagram 6: Signal Processing, Block Diagram of the subVI "Logarithm Sum". Source: own elaboration

13.4 Across-probe OBSI Level Computation

Figure 13.6 depicts the block diagram of the Across-probe OBSI Level Computation.

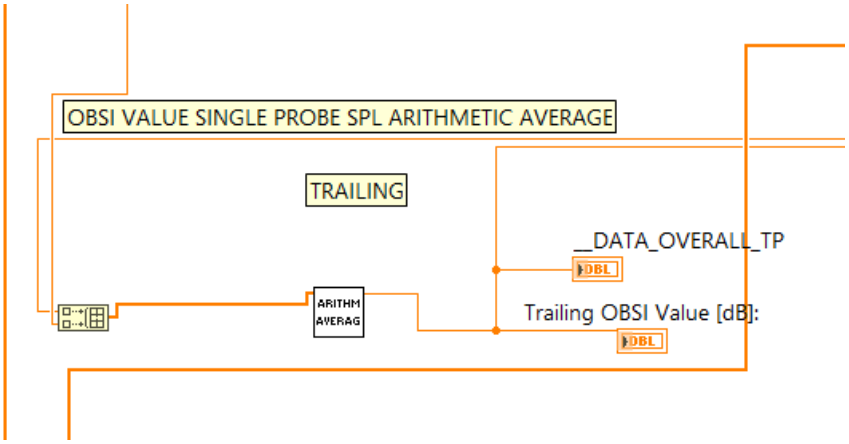


Figure 13.6: Subdiagram 6: Signal Processing, Block Diagram of the Across-probe OBSI Level Computation. Source: own elaboration

In order to arithmetically average the OBSI levels, a subVI has been created named "Arithmetic Average". Figure 13.7 represent this process. The input are both probes OBSI levels and the output the across probe OBSI level.

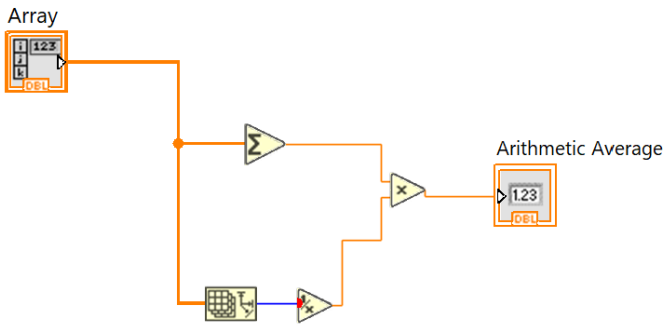


Figure 13.7: Subdiagram 6: Signal Processing, Block Diagram of the Arithmetically Average Computation. Source: own elaboration

13.5 Single Microphone Pressure Intensity Index Computation

Figure 13.8 represents the process used in order to obtain the Single Microphone Pressure Intensity Index Computation. The input are the SPL averaged, and the SPL bands. While the output is the single microphone pressure intensity index for each band.

13.6 Energy Averaged Overall Run Test OBSI Level Computation

Figure 13.9 depicts the process used to obtain the energy-averaged overall run test OBSI level computation. The input parameters are the across-leading probe OBSI level and the across-trailing probe OBSI level. The output is the overall run test OBSI level. It should be noted that in order to perform the energy-averaged subVI, a subVI named "Energy Average" has been created. Figure 13.10 illustrates the methodology employed to compute this average. It adheres to the procedures delineated in the AASHTO standard.

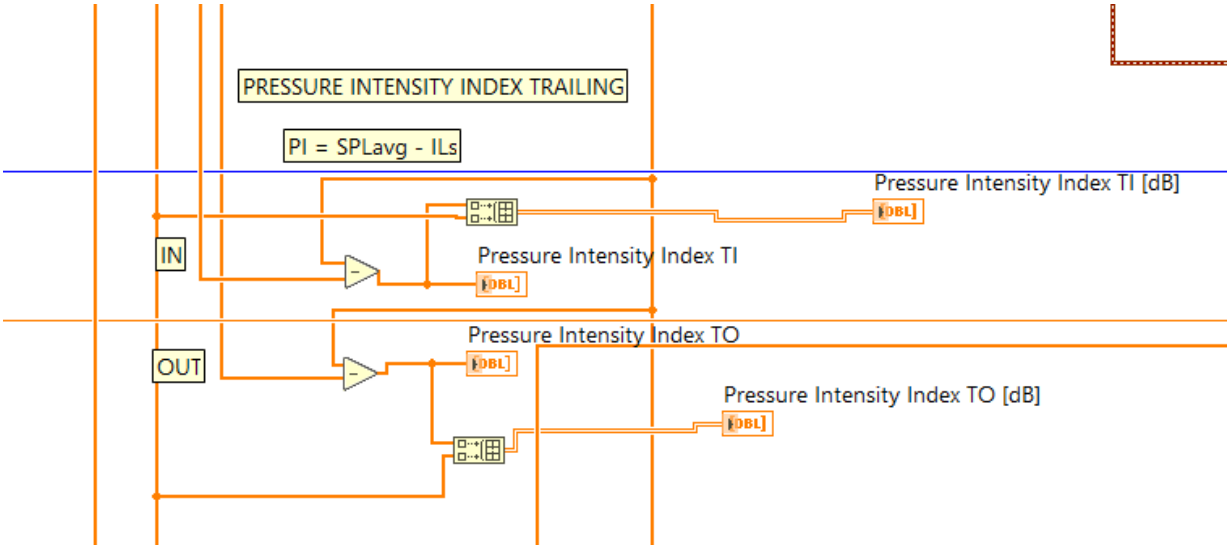


Figure 13.8: Subdiagram 6: Signal Processing, Block Diagram of the Single Microphone Pressure Intensity Index Computation. Source: own elaboration

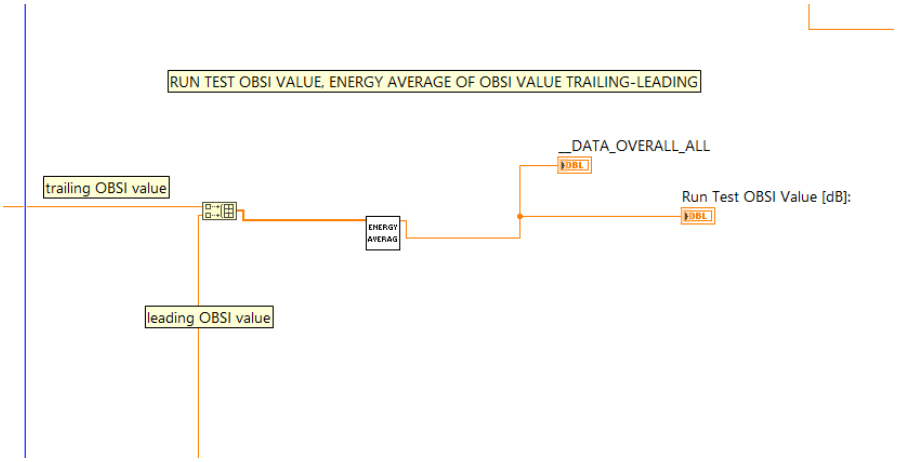


Figure 13.9: Subdiagram 6: Signal Processing, Block Diagram of the Energy Averaged Overall Run Test OBSI Level Computation. Source: own elaboration

13.7 Energy Averaged Overall SPL Computation

Figure 13.11 depicts the process used to obtain the energy-averaged overall SPL computation. The input parameters are the across-leading probe SPL bands and the across-trailing probe SPL bands. The output is the overall SPL bands av-

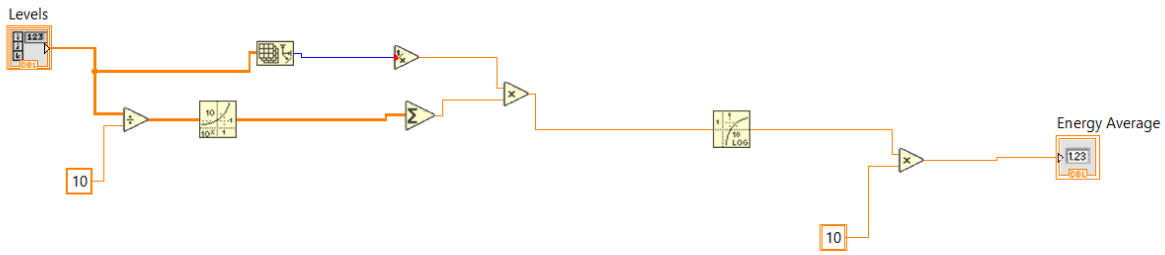


Figure 13.10: Subdiagram 6: Signal Processing, Block Diagram of the Energy Average subVI. Source: own elaboration

eraged. It should be noted that in order to perform the average, the "Energy Average" subVI from previous section has been utilized. Figure 13.10 illustrates the methodology employed to compute this average. In order to perform the average, the program created a for loop which iterates across the bands of each input, performs the average and appends it to a new output array. Finally, the averaged SPL array is concatenated to the frequencies array.

13.8 Single Microphone Sound Pressure Computations

Figure 13.12 depicts the block diagram of the single microphone sound pressure computations.

The computations performed by this block diagram, along with the specific steps and subVIs required to perform them, are explained in the following sections.

13.8.1 A-Weighted One-Third Octave Band Sound Pressure

In order to obtain this value, the program proceeded through the following Sound and Vibration Toolkit VIs: "SVT Third-octave Analysis VI.vi", and "SVT Weighting Filter (octave) VI.vi". The input of these subVIs is the microphone data signal. The output of the last VI is the A-Weighted One-Third Octave Band Sound Pressure. It is notable that in order to obtain the sound pressure or the sound pressure level, the subVI "SVT Third-octave Analysis VI.vi" "db On" input must be set

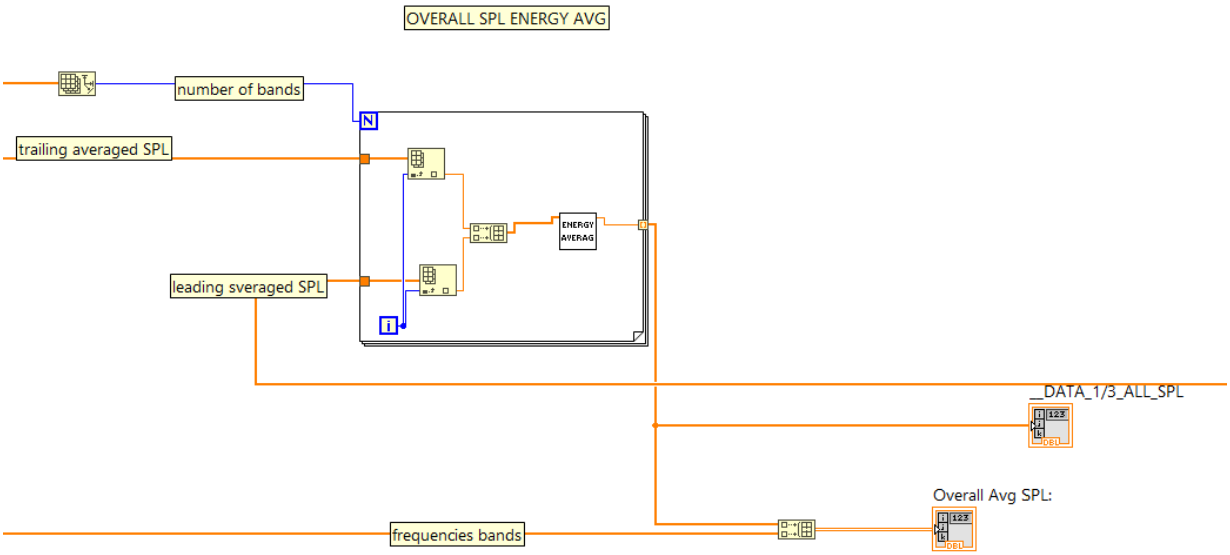


Figure 13.11: Subdiagram 6: Signal Processing, Block Diagram of the Energy Averaged Overall SPL Computation. Source: own elaboration

to TRUE in order to obtain the sound pressure level, and to FALSE if the sound pressure is required.

13.8.2 A-Weighted One-Third Octave Band Sound Pressure Total Band Power

In order to obtain this value, the program proceeded through the following Sound and Vibration Toolkit VIs: "SVT Third-octave Analysis VI.vi", and "SVT Weighting Filter (octave) VI.vi". The input of these subVIs is the microphone data signal. The output of the last VI is the A-Weighted One-Third Octave Band Sound Pressure Total Band Power.

13.9 Equivalent Continuous Sound Level L_{eq} Computations

Figure 13.13 depicts the block diagram of the equivalent continuous sound level L_{eq} computations.

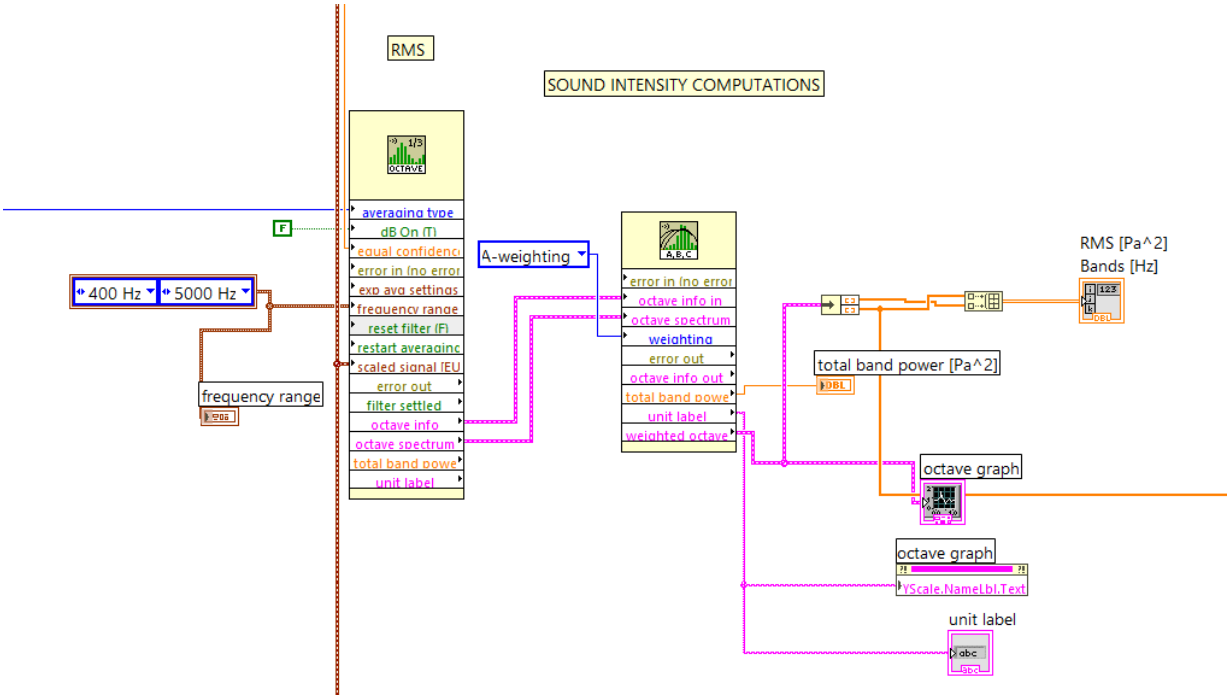


Figure 13.12: Subdiagram 6: Signal Processing, Block Diagram of the Single Microphone Sound Pressure Computations. Source: own elaboration

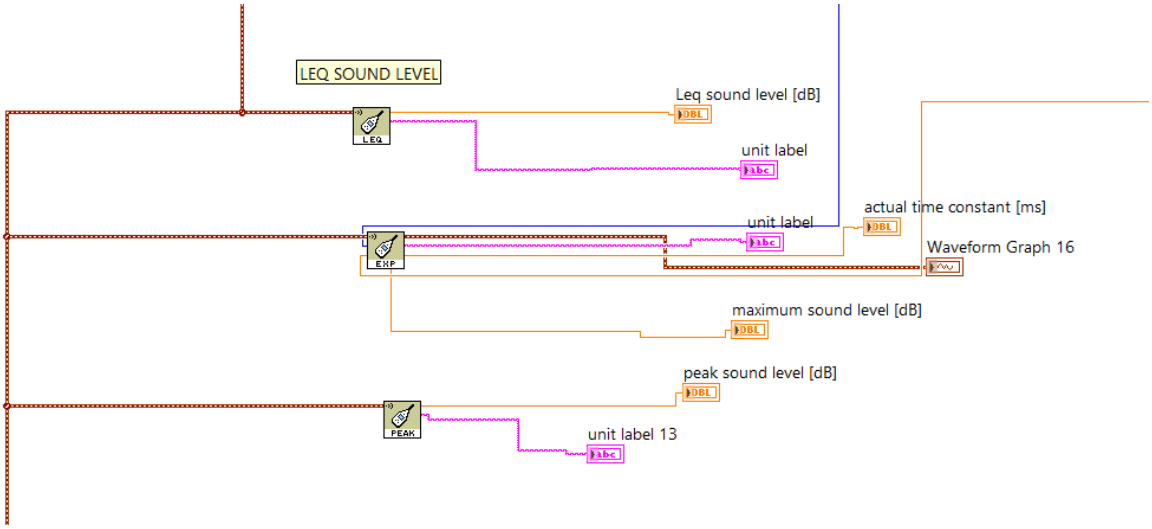


Figure 13.13: Subdiagram 6: Signal Processing, Block Diagram of the Equivalent Continuous Sound Level L_{eq} Computations. Source: own elaboration

The computations performed by this block diagram, along with the specific steps and subVIs required to perform them, are explained in the following sections.

13.9.1 Linear Averaged Equivalent Continuous Sound Level L_{eq}

In order to obtain this value, the program proceeded through the "SVL Leq Sound Level VI.vi". The input of this subVI is the microphone-calibrated data signal, while the output is the linear Averaged Equivalent Continuous Sound Level L_{eq}. This value is displayed as a numerical value, and its units are also exposed.

13.9.2 Exponential Averaged Equivalent Continuous Sound Level L_{eq}

In order to obtain this value, the program proceeded through the "SVL Exp Avg Sound Level VI.vi". The input of this subVI is the microphone-calibrated data signal, while the output is the exponential Averaged Equivalent Continuous Sound Level L_{eq}. This value is displayed as a graphical value, and its units are also exposed. Furthermore, as the user has the option of changing the exponential mode, the time constant in milliseconds used for the computations is displayed.

13.9.3 Peak Sound Level L_{eq}

In order to obtain this value, the program proceeded through the "SVL Peak Sound Level VI.vi". The input of this subVI is the microphone-calibrated data signal, while the output is the Peak Sound Level L_{eq}. This value is displayed as a numerical value, and its units are also exposed.

13.10 Sound Quality Computations

Figure 13.14 depicts the block diagram of the sound quality computations.

The computations performed by this block diagram, along with the specific steps and subVIs required to perform them, are explained in the following sections.

13.10.1 Stationary Loudness

In order to obtain this value, the program proceeded through the "SV Stationary Loudness VI.vi". The input of this subVI is the microphone-calibrated data signal, while the output is the stationary loudness. This value is displayed in two ways: a graphical waveform graph and a numerical array where the value and the critical band rate are represented. Furthermore, the perceived loudness in units of sones and the perceived loudness level in units of phons are also numerically displayed.

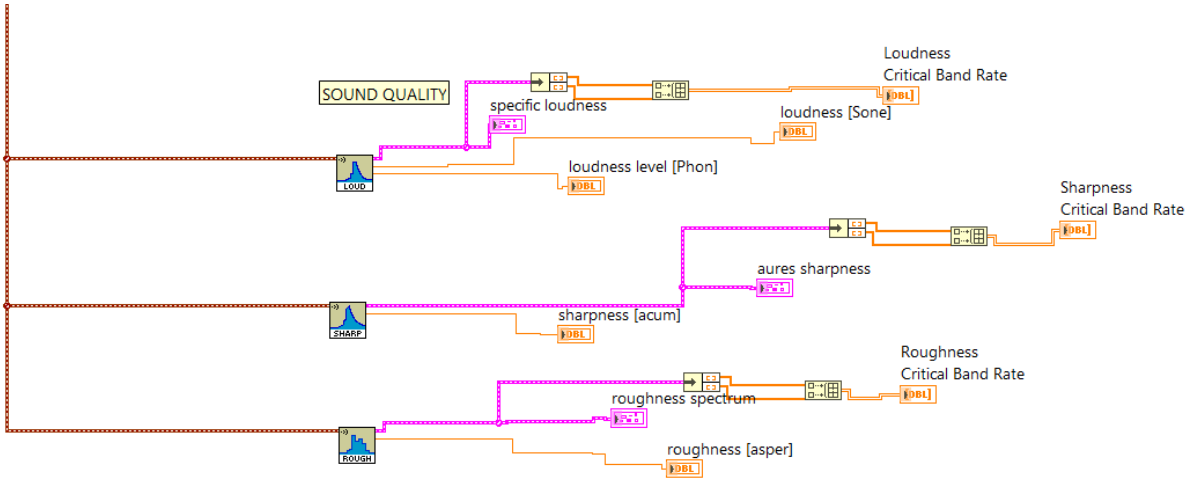


Figure 13.14: Subdiagram 6: Signal Processing, Block Diagram of the Sound Quality Computations. Source: own elaboration

13.10.2 Aures Sharpness

In order to obtain this value, the program proceeded through the "SV Aures Sharpness VI.vi". The input of this subVI is the microphone-calibrated data signal, while the output is the sharpness. This value is displayed in two ways: a graphical waveform graph and a numerical array where the value and the critical band rate are represented. Furthermore, the sharpness measurement in units of acum is also numerically displayed.

13.10.3 Aures Roughness

In order to obtain this value, the program proceeded through the "SV Aures Roughness VI.vi". The input of this subVI is the microphone-calibrated data signal, while the output is the roughness. This value is displayed in two ways: a graphical waveform graph and a numerical array where the value and the critical band rate are represented. Furthermore, the roughness measurement in units of asper is also numerically displayed.

Chapter 14

Subdiagram 7: Save to File

This chapter presents the block diagram of the seventh subdiagram, Save to File. As previously stated, this subdiagram represents the state in which the program will perform the computation in order to ask the user if the trial should be saved, and if so, create two CSV files. The first file will contain the raw acquired data, while the second file will contain the performed computations.

This chapter is divided into three parts:

1. Prompt Message to the User, which will determine whether the user will choose to save the trial into a CSV file.

2. Save Raw Data, Case structure with the required computations to create and save the CSV file.

3. Save Processed Data, analogous to the previous part, however, the saved data will be the processed data obtained through the signal processing computation from subdiagram 6.

14.1 Prompt Message to the User

Figure 14.1 depicts the process used to display the message. Once the signal computations have been completed, the program will proceed to the final and seventh subdiagram. The initial block to be executed is the "Two Button Dialog Box.vi", which will present the user with the question, "Save to File?" The user will then select either "Yes" or "No". In the event that the user selects "No", the computations associated with this subdiagram will not be executed, and they will be omitted. In the event that the user responds in the affirmative, the program will then request that the user input a name for the file. Once the user has

provided a name, the program will initiate the two-case structure, which contains the computations required to create and save the file.

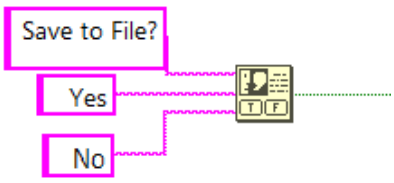


Figure 14.1: Subdiagram 7: Save to File, Block Diagram of the Prompt Message to the User Procedure. Source: own elaboration

14.2 Save Raw Data

Figure 14.2 depicts the process used to save the raw acquired data from the microphone into a CSV file. The computations performed by this block diagram, along with the specific steps and subVIs required to perform them, are explained in the following sections.

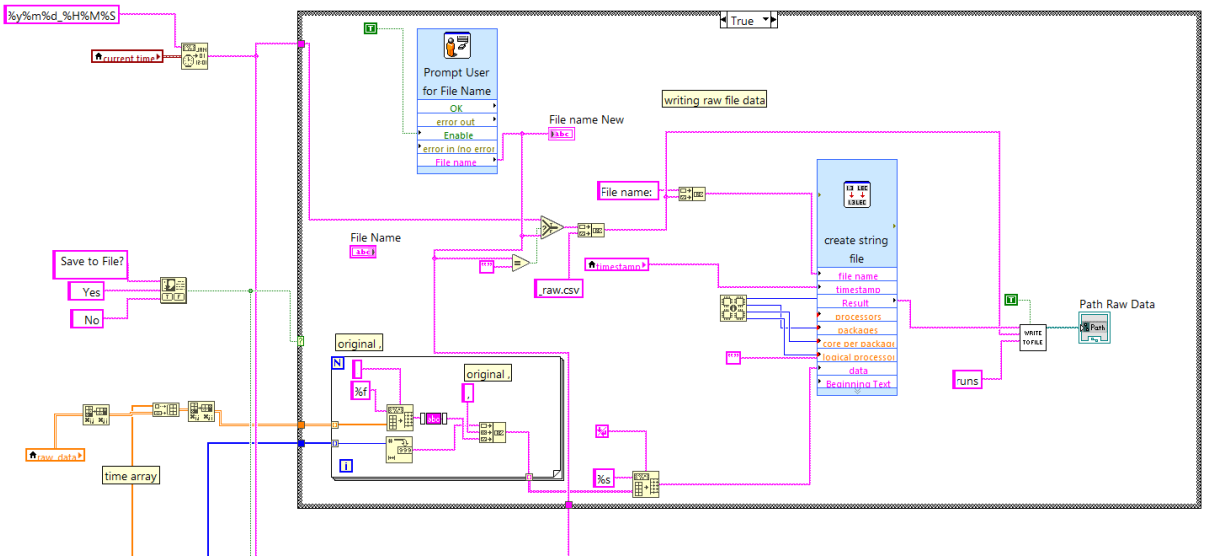


Figure 14.2: Subdiagram 7: Save to File, Block Diagram of the Save Raw Data Procedure. Source: own elaboration

14.2.1 Array to Spreadsheet

The first step is to convert the data, which is in an array variable type, into a spreadsheet string. In order to do so, the followings steps are required:

1. The raw data array will be concatenated with the time array.
2. A for loop will be employed within the case structure to traverse each row, where the time array and the four data values (one for each microphone) are situated. During each iteration, these values will be transformed into strings, and an index will be appended.
3. All of the aforementioned information will be incorporated into a single row, comprising the following elements: index, time(s), dataMic1, dataMic2, dataMic3, dataMic4.

14.2.2 Concatenating Data and Information

In this section, the spreadsheet string created will be combined with the information data from the trial. To achieve this, the following steps must be completed:

1. The information data from the trial will be:
   - File name
   - Timestamp
   - Number of processors (user's CPU)
   - Number of packages (user's CPU)
   - Number of cores per packages (user's CPU)
   - Number of logical processors per core (user's CPU)
   -
2. Subsequently, all of this information will be assembled into a single string.

14.2.3 Write to File

The last step is the creation and the save into the file of the string above created. In order to do so, a subVI has been created named "Write to File". Figure 14.3 represent this process. The input is the concatenated string, and the folder where it will be stored, while the output the path of the file.

Figure 14.4 represent an example of the CSV created where the raw data of the trial is stored.

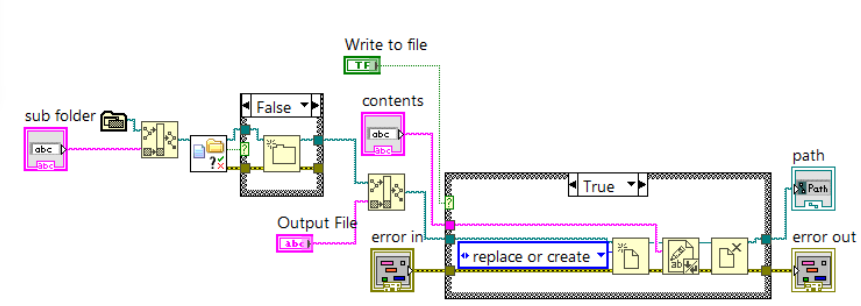


Figure 14.3: Subdiagram 7: Save to File, Block Diagram of the Write to File subVI. Source: own elaboration

I25

	A	B	C	D
1	File name: File name: DEMO TONE 1100_raw.csv			
2	Thursday, May 9, 2024 12:53:53 PM			
3	# Processors: 8			
4	# Packages: 1			
5	# Cores per Packages: 4			
6	# Logical Processors per Core: 2			
7	index, time (s), ai0 (V), ai1 (V), ai2 (V), ai3 (V)			
8	0,0.000000,-0.005686,0.057538,0.001270,-0.002542			
9	1,0.043000,-0.005467,0.010757,0.000561,-0.003609			
10	2,0.044000,-0.005598,-0.030995,-0.000017,-0.003640			
11	3,0.045000,-0.005928,-0.058441,-0.000674,-0.003657			
12	4,0.046000,-0.004728,-0.074082,-0.001825,-0.003202			
13	5,0.047000,-0.002472,-0.083926,-0.002934,-0.001769			
14	6,0.047000,-0.001088,-0.096754,-0.003633,0.000245			
15	7,0.050000,-0.000651,-0.090404,-0.003278,0.002375			
16	8,0.052000,-0.001135,-0.044838,-0.002794,0.004647			
17	9,0.053000,-0.002360,0.009049,-0.002800,0.005311			
18	10,0.053000,-0.002878,0.051704,-0.002794,0.004240			
19	11,0.055000,-0.003063,0.076411,-0.002175,0.003704			
20	12,0.055000,-0.004802,0.080540,-0.001607,0.002974			

Figure 14.4: Example of the CSV Created Where the Raw Data of the Trial is Stored. Source: own elaboration

14.3 Save Processed Data

Figure 14.5 depicts the process used to save the results of the computations into a CSV file. The computations performed by this block diagram, along with the specific steps and subVIs required to perform them, are explained in the following sections.

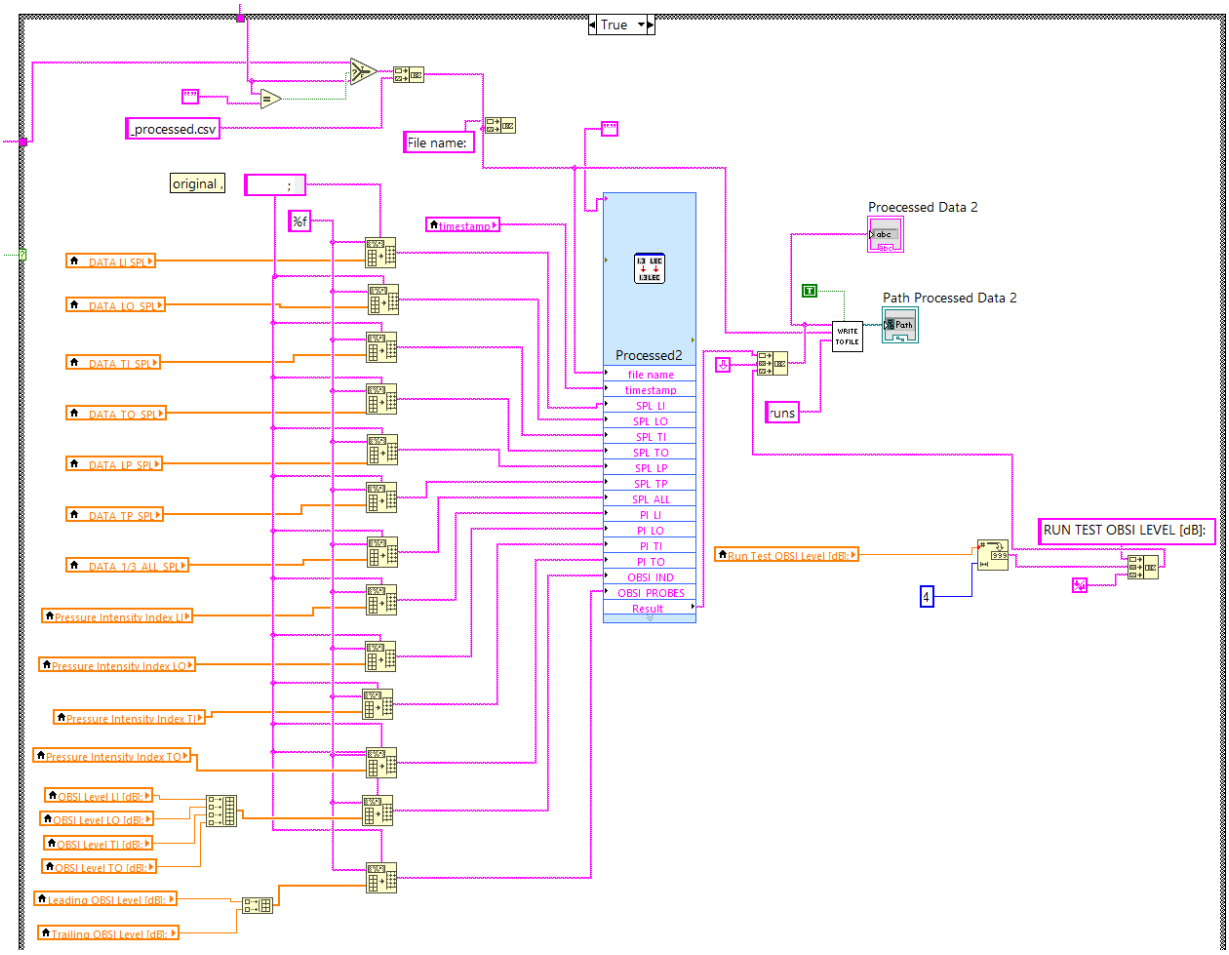


Figure 14.5: Subdiagram 7: Save to File, Block Diagram of the Save Processed Data Procedure. Source: own elaboration

14.3.1 Computed Data Stored in the CSV File

The final CSV file will store the following information about the trial and the performed computations. In other words, row by row, the CSV with the processed data will contain:

- File name
- Timestamp
- Leading Edge In Normalized A-weighted One-Third Octave Band SPL
- Leading Edge Out Normalized A-weighted One-Third Octave Band SPL

- Trailing Edge In Normalized A-weighted One-Third Octave Band SPL
- Trailing Edge Out Normalized A-weighted One-Third Octave Band SPL
- Across Leading Probe Arithmetic Averaged SPL (one value for each band)
- Energy Averaged Overall SPL (in each bands)
- Pressure Intensity Index Leading Edge Probe In (one value for each band)
- Pressure Intensity Index Leading Edge Probe Out (one value for each band)
- Pressure Intensity Index Trailing Edge Probe In (one value for each band)
- Pressure Intensity Index Trailing Edge Probe Out (one value for each band)
- OBSI Level LI, LO, TI, TO (4 values in total)
- Across Leading-Trailing Probe Arithmetic Averaged OBSI Level (two values in total)
- RUN TEST OBSI LEVEL (one value in total)

14.3.2 Results Array to String

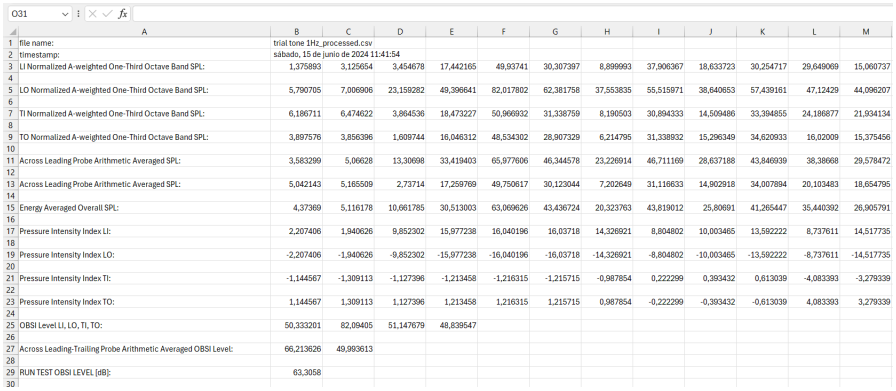
The subVI "Build Text.vi" was utilized in order to convert the arrays in the previous section into strings. In order to do so, the build text subVI needs to know which variables need to be converted. Therefore, a string with the variables was passed, this string is as follows:

file name: %file name%
timestamp: %timestamp%
LI Normalized A-weighted One-Third Octave Band SPL: %SPL_LI%
LO Normalized A-weighted One-Third Octave Band SPL: %SPL_LO%
TI Normalized A-weighted One-Third Octave Band SPL: %SPL_TI%
TO Normalized A-weighted One-Third Octave Band SPL: %SPL_TO%
Across Leading Probe Arithmetic Averaged SPL: %SPL_LP%
Across Leading Probe Arithmetic Averaged SPL: %SPL_TP%
Energy Averaged Overall SPL: %SPL_ALL%
Pressure Intensity Index LI: %PI_LI%
Pressure Intensity Index LO: %PI_LO%
Pressure Intensity Index TI: %PI_TI%
Pressure Intensity Index TO: %PI_TO%
OBSI Level LI, LO, TI, TO: %OBSI_IND%
Across Leading-Trailing Probe Arithmetic Averaged OBSI Level: %OBSI_PROBES%

14.3.3 Write to File

The last step is the creation and the save into the file of the string above created. In order to do so, a subVI has been created named "Write to File". Figure 14.3 represent this process. The input is the concatenated string, and the folder where it will be stored, while the output the path of the

Figure 14.6 represent an example of the CSV created where the processed data of the trial is stored. The signal used was a tone of 1kHz.



	A	B	C	D	E	F	G	H	I	J	K	L	M
1	file name:	trial tone 1Hz_processed.csv											
2	timestamp:	sábado, 15 de junio de 2024 11:41:54											
3	LI Normalized A-weighted One-Third Octave Band SPL:	1,375893	3,125654	3,454678	17,442165	49,93741	30,307397	8,899993	37,906367	18,633723	30,254717	29,649069	15,060737
4													
5	LO Normalized A-weighted One-Third Octave Band SPL:	5,790705	7,006906	23,159282	49,396641	82,017802	62,381758	37,553835	55,515971	38,640653	57,439161	47,12429	44,096207
6													
7	TI Normalized A-weighted One-Third Octave Band SPL:	6,186711	6,474622	3,864536	18,473227	50,966932	31,338759	8,190503	30,894333	14,509486	33,394855	24,186877	21,934134
8													
9	TO Normalized A-weighted One-Third Octave Band SPL:	3,897576	3,856396	1,609744	16,046312	48,534302	28,907329	6,214795	31,338932	15,296349	34,620933	16,02009	15,375456
10													
11	Across Leading Probe Arithmetic Averaged SPL:	3,583299	5,06628	13,30698	33,419403	65,977606	46,344578	23,226914	46,711169	28,637188	43,846939	38,38668	29,578472
12													
13	Across Leading Probe Arithmetic Averaged SPL:	5,042143	5,165509	2,73714	17,259769	49,750617	30,123044	7,202649	31,116633	14,902918	34,007894	20,103483	18,654795
14													
15	Energy Averaged Overall SPL:	4,37369	5,116178	10,661785	30,513003	63,069626	43,436724	20,323763	43,819012	25,80691	41,265447	35,440392	26,905791
16													
17	Pressure Intensity Index LI:	2,207406	1,940626	9,852302	15,977238	16,040196	16,03718	14,326921	8,804802	10,003465	13,592222	8,737611	14,517735
18													
19	Pressure Intensity Index LO:	-2,207406	-1,940626	-9,852302	-15,977238	-16,040196	-16,03718	-14,326921	-8,804802	-10,003465	-13,592222	-8,737611	-14,517735
20													
21	Pressure Intensity Index TI:	-1,144567	-1,309113	-1,127396	-1,213458	-1,216315	-1,215715	-0,987854	0,222299	0,393432	0,613039	-4,083393	-3,279339
22													
23	Pressure Intensity Index TO:	1,144567	1,309113	1,127396	1,213458	1,216315	1,215715	0,987854	-0,222299	-0,393432	-0,613039	4,083393	3,279339
24													
25	OBSI Level LI, LO, TI, TO:	50,333201	82,09405	51,147679	48,839547								
26													
27	Across Leading-Trailing Probe Arithmetic Averaged OBSI Level:	66,213626	49,993613										
28													
29	RUN TEST OBSI LEVEL [dB]:	63,3058											
30													

Figure 14.6: Example of the CSV Created Where the Processed Data of the Trial is Stored. Source: own elaboration

Chapter 15

Result Analysis

Throughout this chapter, the results obtained from the OBSIDAS program are going to be displayed, explained, and interpreted. It is noteworthy that this project has been focused on the development of a program in order to adhere to the computational requirements of the standard AASHTO T360. Therefore, the primary objective of this project is the software, and the results will now be detailed.

The analysis will be divided into two primary sections, each focusing on a distinct signal that has been acquired through the system. The two signals that will be analyzed in this section are as follows:

1. Calibration: Calibration Signal, the second part will be the analysis of the calibration signal. In this part, the system will be fully calibrated in order to obtain the correct values in the subsequent trials and analysis.

2. OBSI Trial, in this last part, a real OBSI signal acquired through the OBSI method and in accordance with the AAHSTO T360 standard will be analyzed.

15.1 Calibration Analysis

In this section the calibration procedure will be employed, as stated in section 2.4. As expressed before, the calibrator signal is designated as the reference signal and is known to have a specific frequency and sound pressure level. These values are used to compare the signal captured by the microphone. The reference value for the sound pressure level will be 94 dBSPL, while the signal frequency will be 1 kHz.

In the first attempt, the signal will not adhere to the 94 dB SPL in the 1kHz. The user will then need to adjust in the calibration settings the sensor sensitiv-

ity. Figure 15.1 represents the A-weighted SPL results from the OBSIDAS user interface.

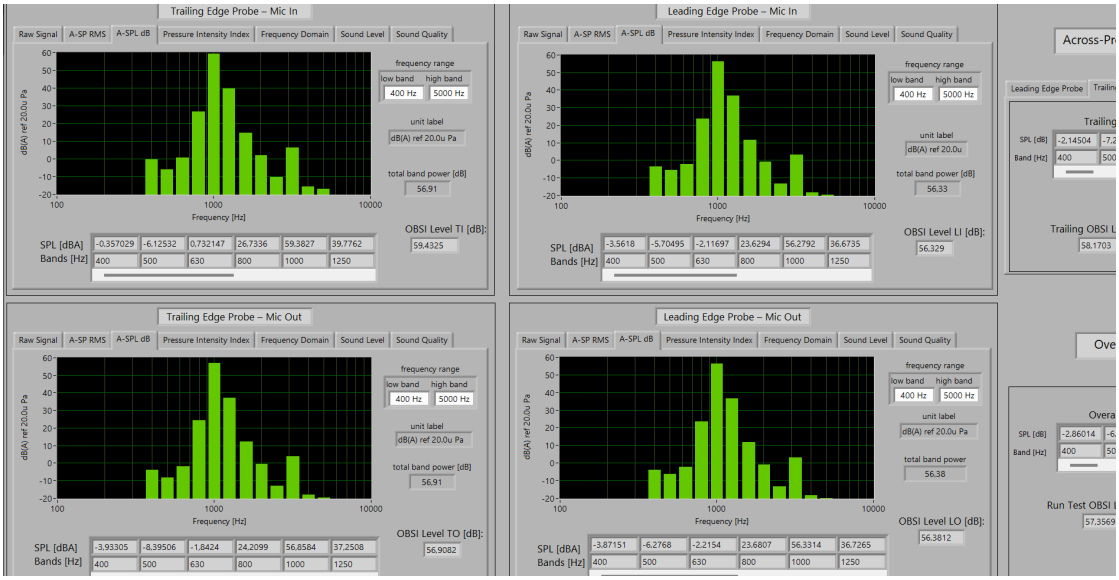


Figure 15.1: OBSIDAS Results of the Calibration Signal With 1 V/Pa as Sensitivity. Source: own elaboration

Therefore, the following dB in the 1kHz band (since it was a 1KHz tone) have been acquired from the calibration signal:

- LI: 56,2792 dBA
- LO: 56,3314 dBA
- TI: 59,3827 dBA
- TO: 56,8584 dBA

Consequently, in order to calibrate the microphones, the results in dBA must be 94. The sensitivity will adjusted accordingly.

Using a sensitivity of 100 mV/Pa in every microphone:

- LI: 76,2941 dBA
- LO: 76,3226 dBA
- TI: 80,6209 dBA

- TO: 76,8714 dBA

Using a sensitivity of 80 mV/Pa in every microphone:

- LI: 78,2221 dBA
- LO: 78,2667 dBA
- TI: 82,549 dBA
- TO: 78,8006 dBA

Using a sensitivity of 60 mV/Pa in every microphone:

- LI: 78,2221 dBA
- LO: 78,2667 dBA
- TI: 82,549 dBA
- TO: 78,8006 dBA

Using the following sensitivities:

- LI: 22 mV/Pa
- LO: 15 mV/Pa
- TI: 12 mV/Pa
- TO: 12 mV/Pa

Figure 15.2 shows the results obtained. Which are the following:

- LI: 94,6971 dBA
- LO: 94,7434 dBA
- TI: 93,759 dBA
- TO: 93,3334 dBA

Consequently, the SPL values from the 1 kHz octave band are found to be approximately 94 dBA when injected with the calibration signal into the program. Therefore, it can be proved that the microphones are set to be calibrated, and that the subsequent trials will be correct, with the results being adjusted to reflect reality.

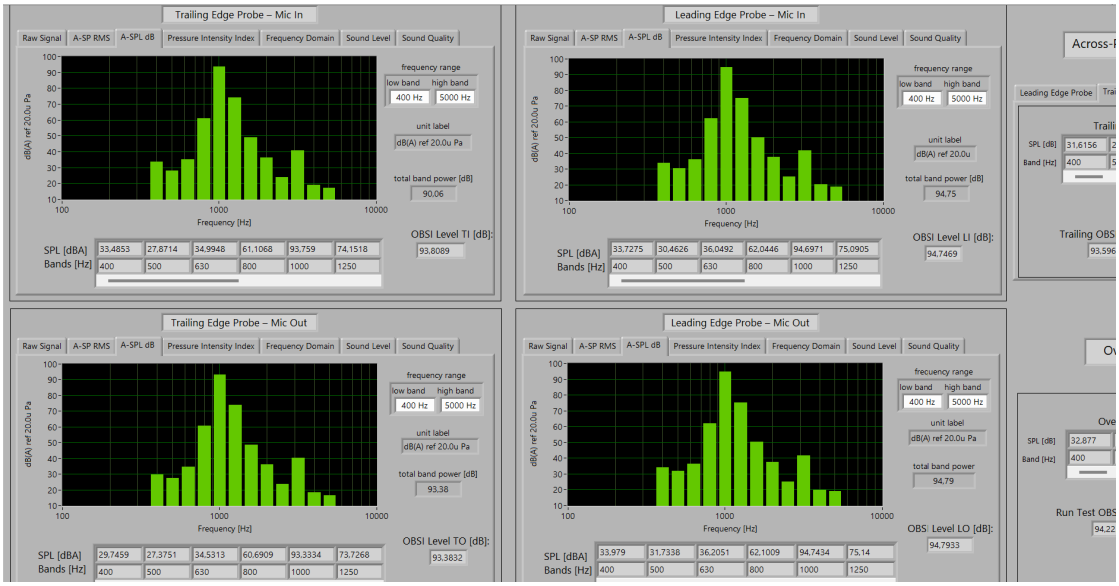


Figure 15.2: OBSIDAS Results of the Calibration Signal With Adjusted Sensitivities. Source: own elaboration

15.2 OBSI Trial

This section will analyze a real-world OBSI trial that adheres to the AASHTO standard. This project has been developed in order to be able to perform this type of trials and to conduct the required measurements. In this analysis, every feature of the OBSIDAS program will be utilized.

It is noteworthy that this test has been performed by John A. Volpe National Transportation Systems Center. The test was performed in accordance with the AASHTO standard, which is explained in detail throughout this document. It is also worth noting that the vehicle was tested at a speed of 72 km/h (45 mph), which corresponds to a trial period of 6.7 ±0.2 seconds.

Unfortunately, the OBSI signals of the trial, which its results will be analyzed in this section, were not acquired in situ on the vehicle performing the test. The .wav file audio has been inserted into the OBSIDAS program in order to obtain the results and information hereby. Additionally, the previous section, where the microphones have been calibrated, used .wav file as calibration signals. The same equipment acquired the calibration signals and the OBSI noise. Therefore, the calibration has been developed in a correct procedure.

The results of this trial will be divided into two sections. The first section, named "AASHTO Required Results," consists of the requisite AASHTO standard measurements, the results which are added into the CSV file. These measurements

will be explained in detail. The second section, named "Full OBSIDAS Results," encompasses the entirety of the results and measurements generated by OBSIDAS. The results will be displayed as images captured directly from the user interface results tab.

15.2.1 AASHTO Required Results

Consequently, the CSV file with the results of the trial is shown in figure 15.3. The results encompass the requisite data as outlined in the standard.

	A	B	C	D	E	F	G	H	I	J	K	L	M
1	file name:	VOLPE OBSI CALIBRATED_processed.csv											
2	timestamp:	sábado, 15 de junio de 2024 13:46:20											
3	LI Normalized A-weighted One-Third Octave Band SPL:	91,301976	92,604261	96,245959	97,362593	92,909724	89,316975	85,854033	83,79025	80,040521	76,004942	70,908507	67,387878
4													
5	LO Normalized A-weighted One-Third Octave Band SPL:	90,251464	91,601761	95,242009	96,477338	91,830852	88,25002	84,707115	82,664948	78,974372	75,298579	70,147296	66,567656
6													
7	TI Normalized A-weighted One-Third Octave Band SPL:	87,928463	89,777194	91,531521	94,949731	92,629287	89,275473	84,632828	80,703865	78,365418	75,170118	70,500393	66,336229
8													
9	TO Normalized A-weighted One-Third Octave Band SPL:	86,79204	88,687474	90,506821	93,967743	91,450661	88,109374	83,409274	79,472757	77,180293	74,19863	69,402092	64,848639
10													
11	Across Leading Probe Arithmetic Averaged SPL:	90,77672	92,103011	95,743984	96,919966	92,370288	88,783497	85,280574	83,227599	79,507446	75,651761	70,527901	66,977767
12													
13	Across Leading Probe Arithmetic Averaged SPL:	87,360251	89,232334	91,019171	94,458737	92,039974	88,692424	84,021051	80,088311	77,772855	74,684374	69,951242	65,592434
14													
15	Energy Averaged Overall SPL:	89,396119	90,900665	93,994718	95,86142	92,20827	88,738199	84,696314	81,93564	78,726182	75,194947	70,249136	66,340105
16													
17	Pressure Intensity Index LI:	-0,525256	-0,50125	-0,501975	-0,442627	-0,539436	-0,533478	-0,573459	-0,562651	-0,533075	-0,353182	-0,380605	-0,410111
18													
19	Pressure Intensity Index LO:	0,525256	0,50125	0,501975	0,442627	0,539436	0,533478	0,573459	0,562651	0,533075	0,353182	0,380605	0,410111
20													
21	Pressure Intensity Index TI:	-0,568211	-0,54486	-0,51235	-0,490994	-0,589313	-0,58305	-0,611777	-0,615554	-0,592562	-0,485744	-0,549151	-0,743795
22													
23	Pressure Intensity Index TO:	0,568211	0,54486	0,51235	0,490994	0,589313	0,58305	0,611777	0,615554	0,592562	0,485744	0,549151	0,743795
24													
25	OBSI Level LI, LO, TI, TO:	102,15478	101,170544	99,695529	98,618782								
26													
27	Across Leading-Trailing Probe Arithmetic Averaged OBSI Level:	101,662662	99,157155										
28													
29	RUN TEST OBSI LEVEL [dB]:	100,5881											

Figure 15.3: OBSIDAS CSV Results of the OBSI Trial. Source: own elaboration

Normalized A-weighted One-Third Octave Band SPL

According to the results, the normalized A-weighted one-third octave band SPLs are consistently similar in each of the bands. There are no bands that are significantly better or worse than others in terms of noise. Notice that the SPL becomes more and more attenuated along the frequency spectrum. This means that the low frequencies are slightly louder than the high frequencies.

This natural order of frequencies can be seen in each of the four microphones. In addition, the trailing edge probe microphone out is the least noisy, while the leading edge probe microphone in is the loudest.

Furthermore, the leading microphones appear to be louder than the trailing microphones, which will be confirmed in the across-probe analysis in the subsequent sections.

Across-Probe Arithmetic Averaged SPL

As mentioned in the previous section, the leading edge is noisier than the trailing edge. This means that this part of the tire produces more loudness than the final part. In each of the bands, the leading part is higher in SP levels. However, the two probes become more and more similar along the frequencies. Therefore, in the high frequencies the difference is not noticeable, while in the low frequencies it could be said that the leading part produces more noise than the trailing part.

Energy Averaged Overall SPL

The values obtained in the energy averaged overall SPL indicate that the attenuation of levels along frequencies is less violent. However, it is still visible.

Pressure Intensity Index

In order for the pressure intensity index to be considered valid, it must meet the maximum and minimum levels shown in the figure 2.4. And there are no values that exceed or fall below the requirements. Therefore, it could be said that this trial is valid. Since the pressure intensity index is along the requirements.

Additionally, there are no values that are even close to the maximum or minimum levels, so this is a symptom that the trial was conducted in the most meticulous way and procedure.

OBSI Levels

The primary and most important measurement to define and state whether a road or pavement is noisy or not is the OBSI level. If only one measurement from the trial is needed to draw conclusions, it would be the OBSI level.

It is noteworthy that the OBSI level of each microphone is distinct. The microphone with the highest decibel level is the leading edge probe microphone in, followed by the leading edge microphone out, along with the trailing edge probe microphone in. The trailing edge microphone out represents the least noisy levels. A change of approximately 1 dB can be observed in both microphones of a single probe.

As identified in the previous section, the across-probe OBSI level indicates that the leading edge probe is noisier than the trailing. A difference of more than 1.5 decibels can be observed. This demonstrates that, although the difference is not significant, OBSIDAS can detect minimum changes between microphones.

Finally, the overall run test OBSI level is 100,5881 dB. Which, compared to the table in figure 2.3 indicates that the road is highly noisy. Therefore, it is recommended that improvements in the quality of the pavement should be implemented

in order to make it quieter. Note that this decibel level could be seen as high, however, the measurement is realized in a really close distance, so the pedestrians would not experience this level but rather a lower one, depending on the distance they are from the road.

15.2.2 Full OBSIDAS Results

In the following figures, the entirety of the OBSIDAS results will be shown.

Raw Graphs

Figure 15.4 represents the raw signal graphs of each of the individual microphones.

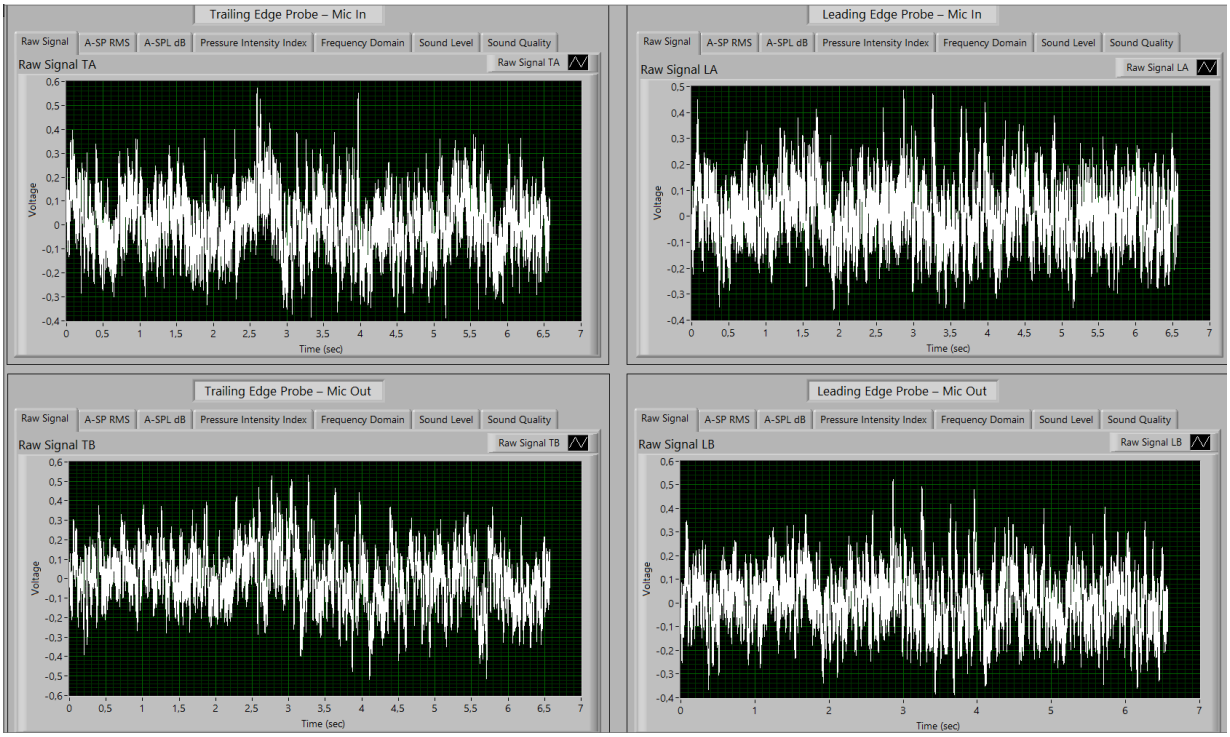


Figure 15.4: Raw Signal Graphs from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

A-Weighted One-Third Octave Band Sound Pressure RMS Analysis

Figure 15.5 represents the A-weighted one-third octave band sound pressure RMS graphs of each of the individual microphones.

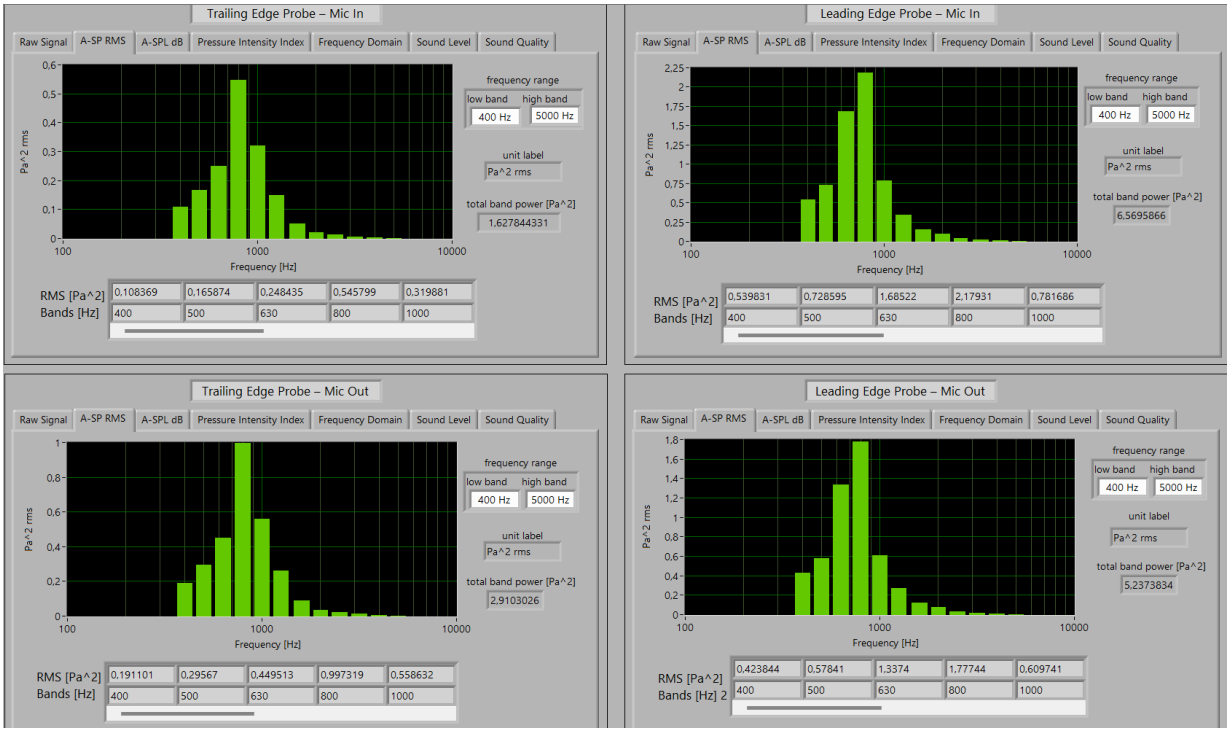


Figure 15.5: A-Weighted One-Third Octave Band Sound Pressure RMS Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

A-Weighted One-Third Octave Band Sound Pressure Level Analysis

Figure 15.6 represents the A-weighted one-third octave band sound pressure level graphs of each of the individual microphones.

Pressure Intensity Values

Figure 15.7 represents the pressure intensity values of each of the individual microphones.

LabVIEW Native FFT Analysis

Figure 15.8 represents the LabVIEW native FFT analysis of each of the individual microphones.

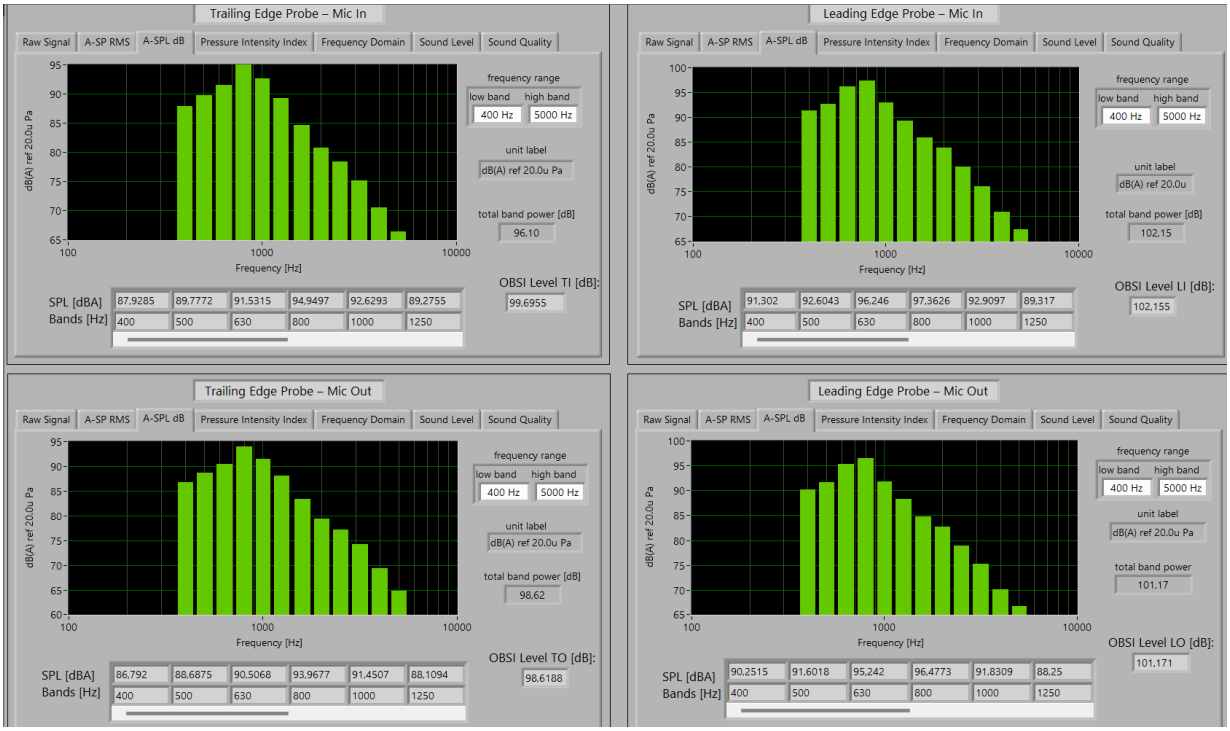


Figure 15.6: A-Weighted One-Third Octave Band Sound Pressure Level Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

Sound and Vibration Toolkit FFT Analysis

Figure 15.9 represents the Sound and Vibration FFT analysis of each of the individual microphones.

Power Spectral Analysis

Figure 15.10 represents the power spectral analysis of each of the individual microphones.

Sound Level Analysis

Figure 15.11 represents the linear and peak equivalent continuous sound level analysis of each of the individual microphones.

Figure 15.12 represents the exponential equivalent continuous sound level analysis of each of the individual microphones.

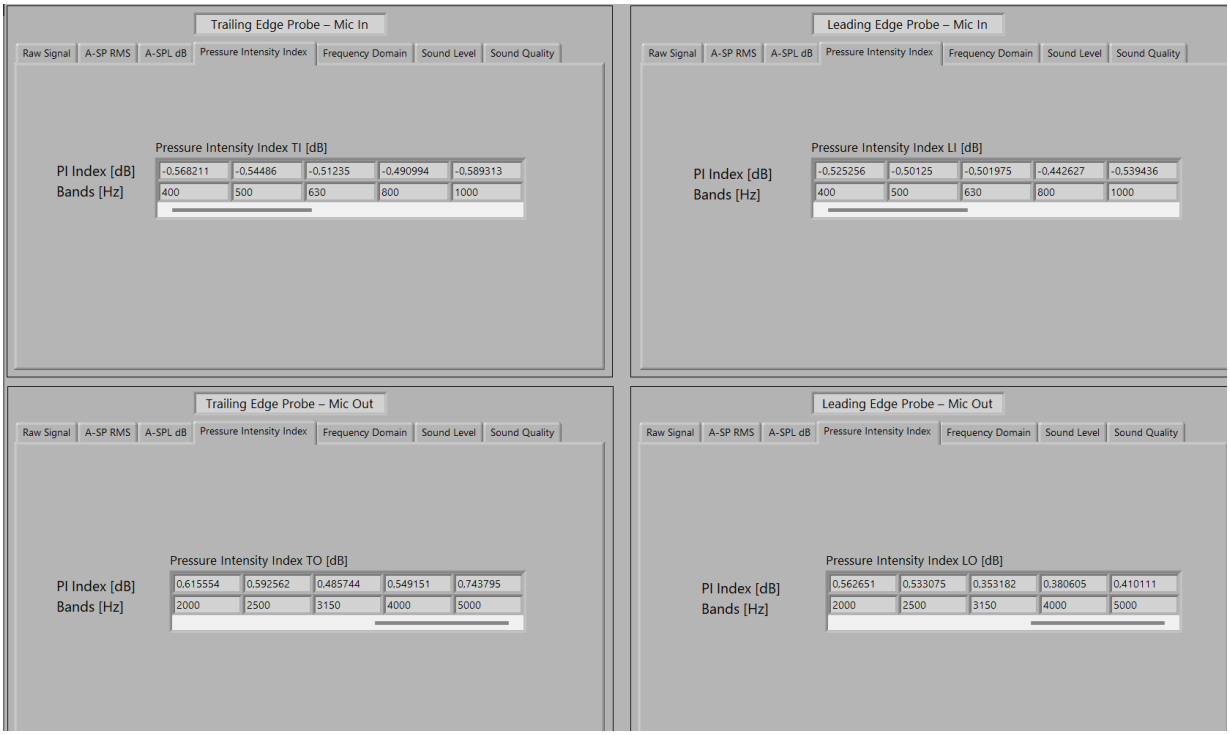


Figure 15.7: Pressure Intensity Values from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

Sound Quality Analysis

Figure 15.13 represents the stationary loudness analysis of each of the individual microphones.

Figure 15.14 represents the Aures sharpness analysis of each of the individual microphones.

Figure 15.15 represents the Aures roughness analysis of each of the individual microphones.

Across-Probe Analysis

Figure 15.16 represents the across leading probe analysis of each of the individual microphones.

Figure 15.17 represents the across trailing probe analysis of each of the individual microphones.

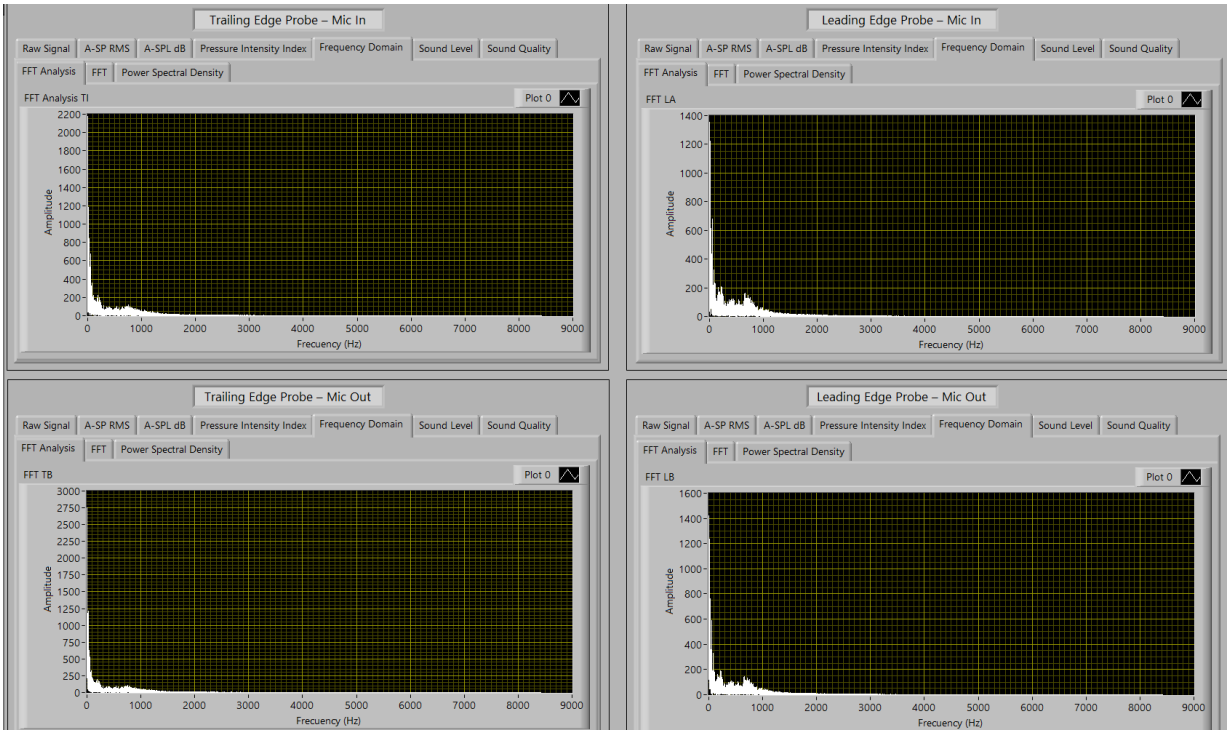


Figure 15.8: LabVIEW Native FFT Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

Overall Analysis

Figure 15.18 represents the overall analysis of each of the individual microphones.

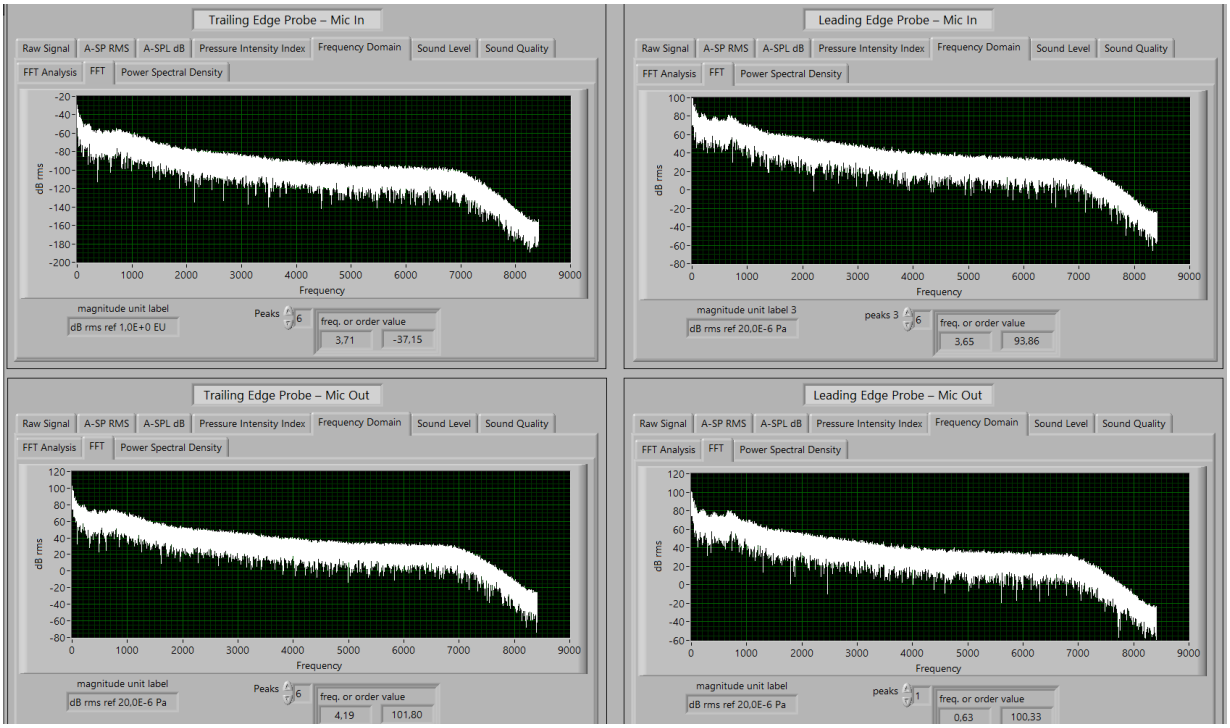


Figure 15.9: Sound and Vibration Toolkit FFT Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

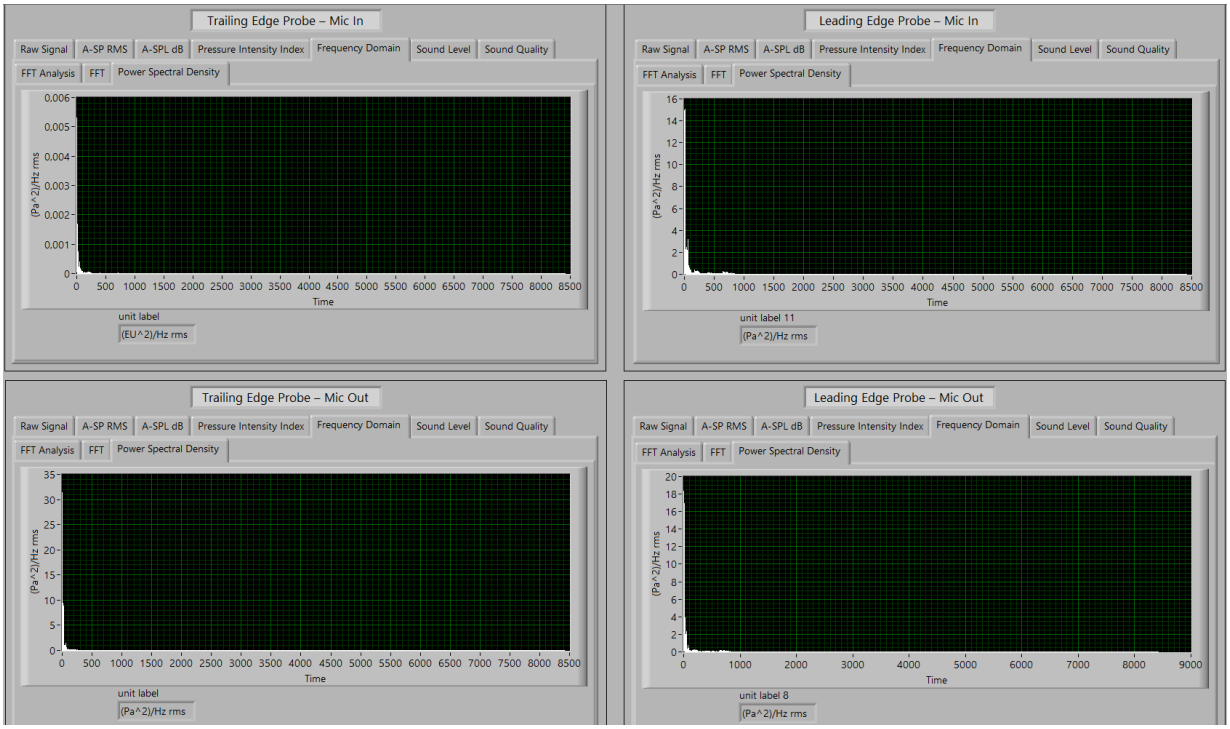


Figure 15.10: Power Spectral Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

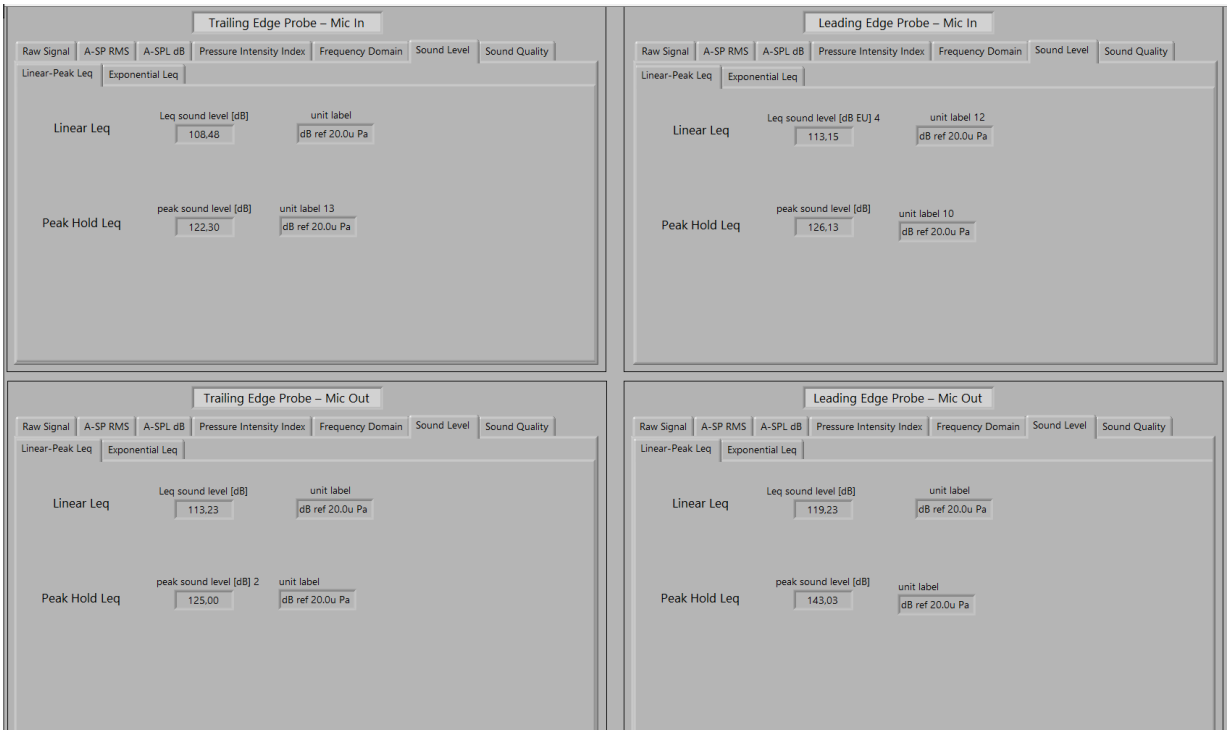


Figure 15.11: Linear and Equivalent Continuous Sound Level Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

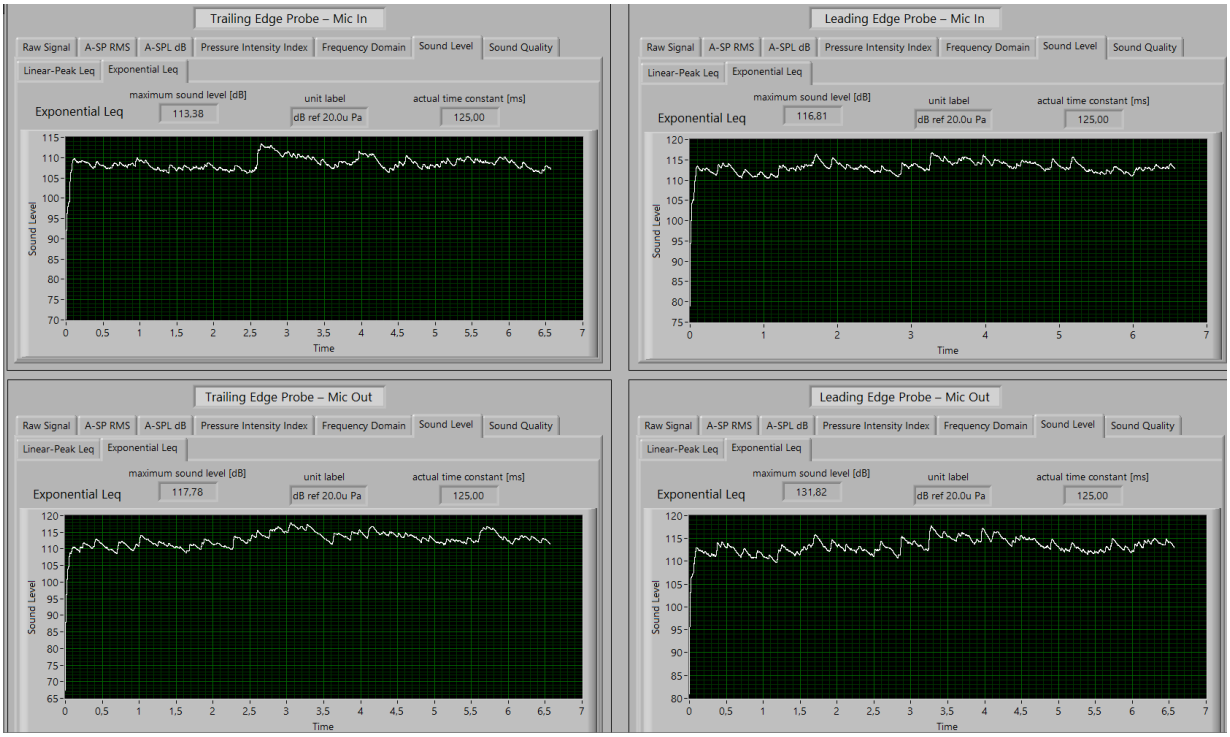


Figure 15.12: Exponential Continuous Sound Level Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

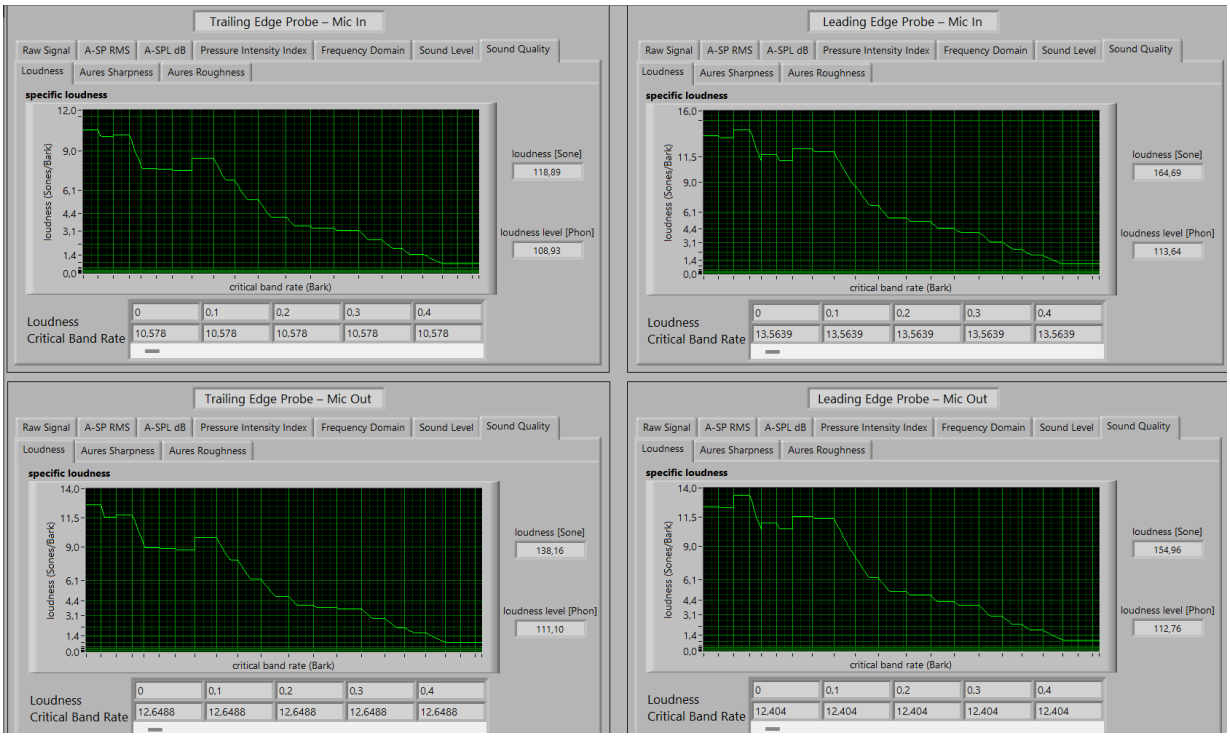


Figure 15.13: Stationary Loudness Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

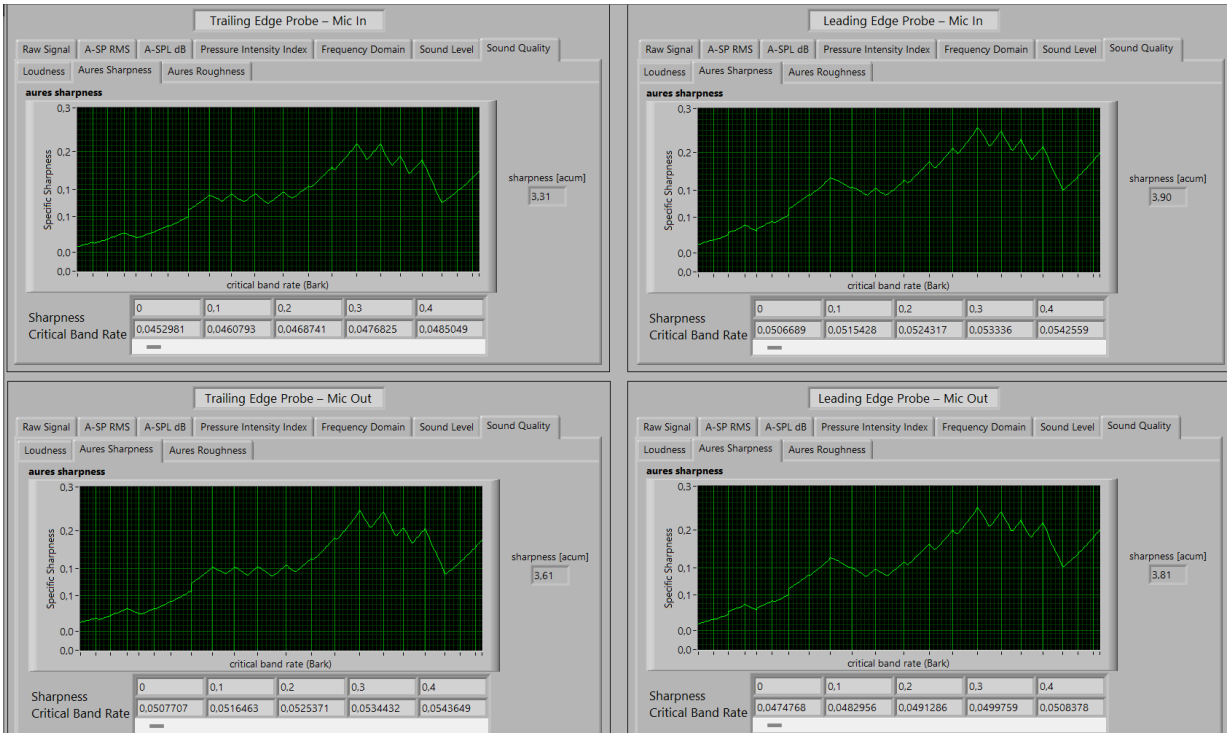


Figure 15.14: Aures Sharpness Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

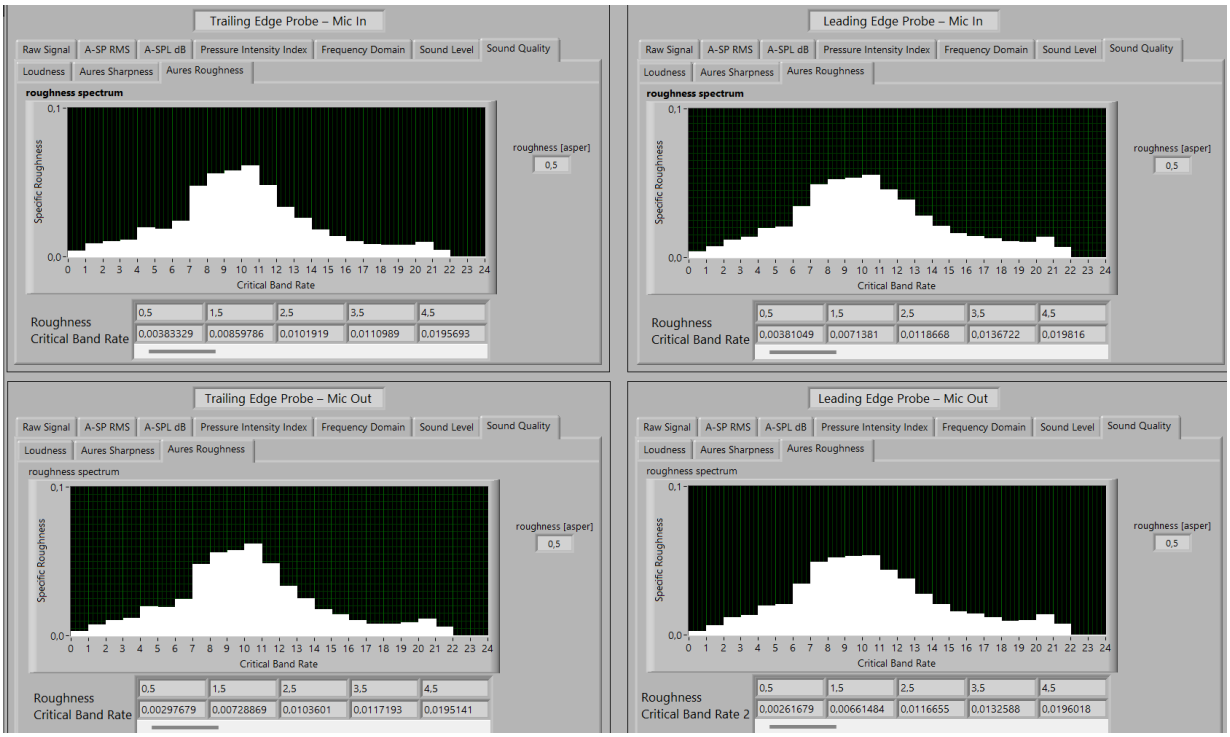


Figure 15.15: Aures Roughness Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

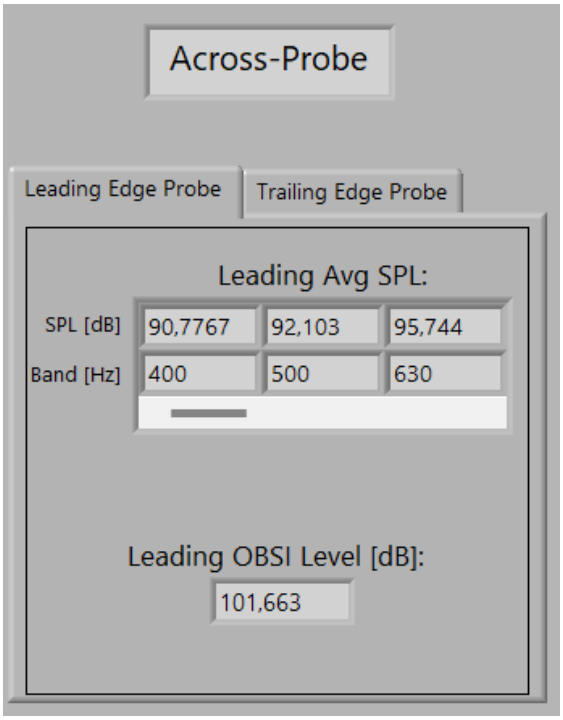


Figure 15.16: Across Leading Probe Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

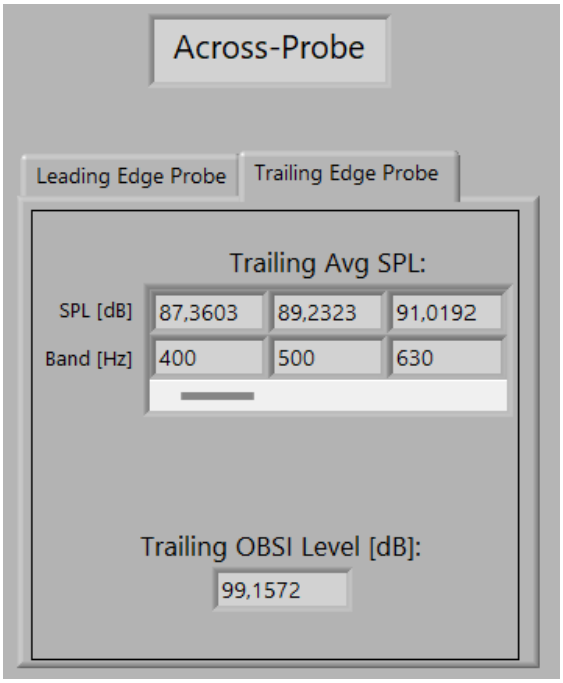


Figure 15.17: Across Trailing Probe Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

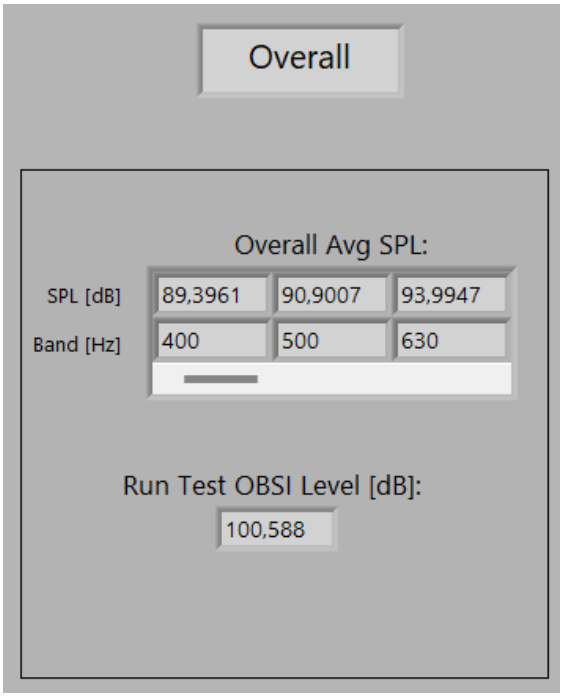


Figure 15.18: Overall Analysis from the OBSI Trial in the OBSIDAS User Interface. Source: own elaboration

Chapter 16

Conclusions and Future Work

To conclude this project, an overview of the objectives at the beginning of this document will be conducted.

In the objectives to be achieved in the section 5.2, it was stated that the main goal of this project was to implement a software program developed in LabVIEW, designed to acquire the noise signal, compute the necessary sound level measurements, and graphically display the results in order to perform the AASHTO T360 standard.

The OBSIDAS program not only adheres to all AASHTO T360 procedures, but also presents the results in graphs and schemes in a highly intuitive user interface. Furthermore, it performs additional sound quality and level measurements, as well as frequency analysis, thereby offering the user more information about the OBSI test. In conclusion, each part of the objectives has been achieved.

This project has contributed to the sector by creating this software. Which is accessible and downloadable on any current machine. It has achieved its main purpose: to quantify and better understand the problem of automotive noise on the roads. Additionally, it obtains the necessary data in order to discover new technologies to reduce the noise level on the roads.

In conclusion, this project contributes to the investigation of the reduction of tire pavement noise, which is of paramount importance given its adverse health and welfare consequences for humans and wildlife.

16.1 Future Work

The project could always be improved. A list with the possible improvements that the software could implement is the following: Every project has the option to be improved. The following is a list of possible improvements that the software could implement:

- Both, the normalized to a reference temperature A-weighted one-third octave band sound pressure level and the measured A-weighted one-third octave band sound pressure level, could be diplayed to the user. Currently, only the normalized values are showed.
- A new feature which informs the user when test is not valid due to the requirements emphasized by the standard (e.g., pressure index). In the current software, the user has to make sure about this manually, however, an automatic procedure could be implemented.
- The software could implement a greater variety of file formats in which the data would be saved.
- Additionally, the software could implement a way to read a wider variety of file types.

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Sustainable Development Goals Alignment

In 2015, the United Nations adopted the 2030 Agenda for Sustainable Development. It proposes a goal for nations, businesses, and individuals to improve the lives of all people without excluding anyone. To this end, the Agenda consists of 17 Sustainable Development Goals, which state that: *"ending poverty must go hand-in-hand with strategies that build economic growth and address a range of social needs including education, health, social protection, and job opportunities, while tackling climate change and environmental protection"* [84] These 17 Sustainable Development Goals are shown in the figure 1. These goals are divided into three different dimensions of action: economy (SDGs 8, 9, 10, and 12), society (SDGs 1, 2, 3, 4, 5, 7, 11, 16), biosphere (SDGs 6, 13, 14, 15), and partnership to achieve the goals (SDG 17). The Figure 2 below illustrates the dimension of each goal.

The current Bachelor's Final Project must align with these objectives, advance them, and contribute to their realization. Each objective is associated with a specific scope and purpose, and upon examination, it has been determined that this project has a greater impact on the objectives that will be highlighted below, in the table 1.

The project has an impact on the three dimensions of the United Nations goals. In terms of the society dimension, the project contributes to objective number 3: "Good Health and Well-Being" in a primary role. More specifically, it contributes to target 3.4, as described in table 1. As discussed in the introduction of hereby document, many non-communicable diseases, such as hypertension leading to myocardial infarction, renal failure, thrombosis, atherosclerosis or stroke, are caused by road noise. Consequently, the study and research that this project would facilitate may result in a reduction of these diseases and their prevention.

In terms of the economy dimension, the project contributes to objective number 9: "Industry, Innovation, and Infrastructure" in a secondary role. More specifically, it contributes to target 9.5, as described in table 1. This is because research on the noise produced by vehicles on the road is increasing, which is leading to



Figure 1: The Sustainable Development Goals (SDGs). Source: [85]

scientific research and improving the technological capacity of industrial sectors by promoting innovation.

Regarding the biosphere dimension, the project contributes to objective number 15: "Life on Land" in a primary role. More specifically, it addresses target 15.5, as described in table 1. As discussed in the introduction of this document, automobile road noise is a significant contributor to the displacement, reduction of activity areas, and low reproductive success of animal species. Consequently, the study and research that this project would facilitate may result in a reduction of the degradation of natural habitats and the stopping of the loss of biological diversity.

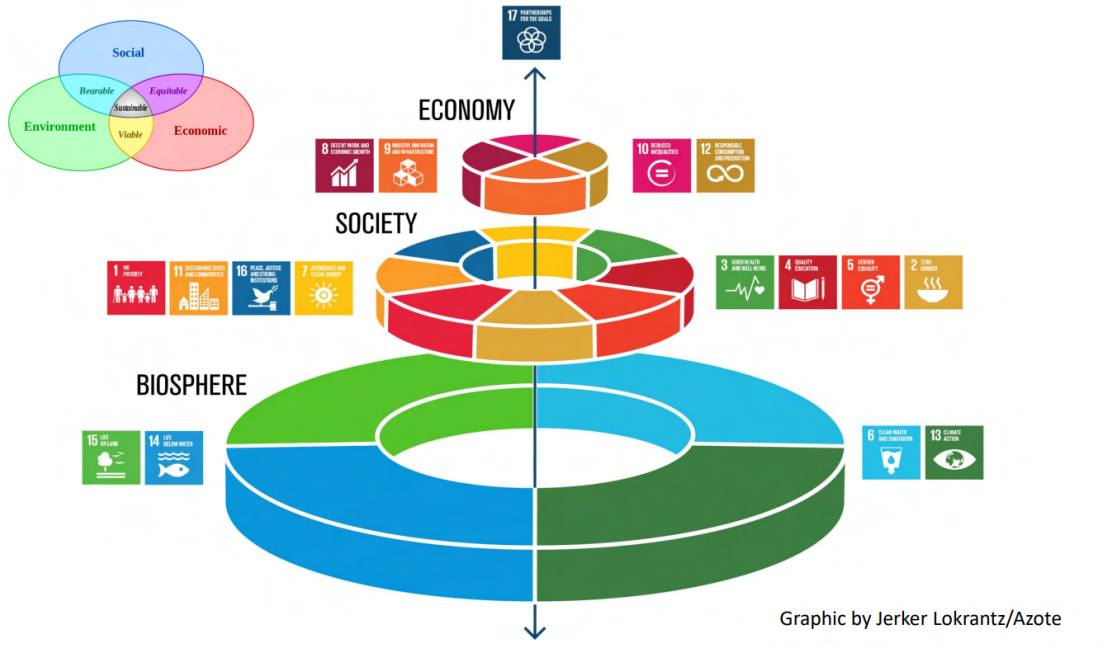


Figure 2: Dimensions of the 17 UN Sustainable Development Goals. Source: [86]

Dimension	SDG	Role	Target
Society	3. Good Health and Well-Being	Primary	**3.4 Reduce Mortality from Non-Communicable Diseases and Promote Mental Health;** by 2030, reduce by one third premature mortality from non-communicable diseases through prevention and treatment and promote mental health and well-being. [87].
Economy	9. Industry, Innovation and Infrastructure	Secondary	**9.5 Enhance Research and Upgrade Industrial Technologies;** Enhance scientific research, upgrade the technological capabilities of industrial sectors in all countries, in particular developing countries, including, by 2030, encouraging innovation and substantially increasing the number of research and development workers per 1 million people and public and private research and development spending [88].
Biosphere	15. Life on Land	Primary	**15.5 Protect Biodiversity and Natural Habitats;** Take urgent and significant action to reduce the degradation of natural habitats, halt the loss of biodiversity and, by 2020, protect and prevent the extinction of threatened species [89].

Table 1: Objectives on which the project has the greatest impact.

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