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Built2Spec

Built to Specifications – Tools for the 21st Century Construction Site

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D3.9 Mobile devices lab test results

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Executive summary

The Deliverable D3.9 entitled “Mobile Devices Lab Test Results” is a public document delivered in the context of WP3, Task 3.3: Acoustic performance

This work is part of the project on Tools for the 21st Century Construction Worksite (BUILT2SPEC) and is financed by the European Union under the Horizon 2020 Program.

This deliverable D3.9 reports the bibliographic and experimental work done to prepare the mobile device for sound insulation measurements.

This document is structured as follows:

- A general state of the art presents the acoustic descriptors needed to measure and the use of mobile phones for acoustic applications
- A characterization of the microphone used for the measurement is performed
- The structure of the application is then presented
- Finally, the calibration tests are reported

1 Table of Contents

Abbreviations

B2S = Built to Specifications

DOA = Description of Action;

CS = Communication Strategy;

SPL = Sound Pressure Level

FFT= Fast Fourier Transform

WP = Work Package.

1 Introduction

The deliverable 3.9 is related with subtask 3.3.1 *Evaluation of the use of mobile devices as an alternative to class 1 microphones for acoustic measurements*. This subtask evaluates the use of mobile devices for airborne sound insulation measurements.

The deliverable 3.9 summarizes the information obtained during the calibration process of the audio device.

In this process, an important work related to signal processing has been done. A software has been developed to test different input signals and an electronic device has been created to emulate the analog-digital converter of the mobile device. A set of experimental measurements have been performed to compare the results between a Class 1 microphone and the smartphone.

2 State of the art

2.1 Sound insulation measurements

Laboratory vs field measurements

Laboratory airborne sound insulation measurements are used to compare the sound insulation provided by different test elements. In this case it is assumed that all the sound is transmitted via the test element and that the structure of the facility plays no role in the acoustic measurement.

In this case, the sound reduction index is defined as

$$R = 10\log\left(\frac{1}{\tau}\right)$$

Where τ is the transmission coefficient, defined as the ratio of the sound power transmitted by the test element (W_2) to the sound power incident on the test element (W_1).

$$\tau = \frac{W_2}{W_1}$$

However, for in situ airborne sound insulation, sound is not only transmitted by the partition wall or floor, but also by flanking transmission. In this case, the transmission coefficient τ' includes the sound power transmitted by flanking elements (W_3)

$$\tau = \frac{W_2 + W_3}{W_1}$$

In this case, the apparent sound reduction index is defined as

$$R' = 10\log\left(\frac{W_1}{W_2 + W_3}\right) = L_{p1} - L_{p2} + 10\log\left(\frac{S}{A}\right)$$

Where L_{p1} and L_{p2} are the Sound Pressure Level in the source and receiver room respectively. The importance of the flanking transmission in the sound insulation results is why field measurements are necessary to guarantee a good acoustic performance.

Acoustic descriptors

For field measurements, airborne sound insulation can be described by the sound pressure level difference

$$D = L_{p1} - L_{p2}$$

This can add complexity to the regulatory process because the result will depend on the absorptive material from the receiving room. This forces to measure the reverberation time in order to normalize the level difference. The level difference, D, is *standardized* using a reference value for the reverberation time, and is *normalized* using a reference value for the absorption area. In that case, we can define the

- normalized level difference as $D_n = D - 10\log\left(\frac{A}{A_0}\right)$, where $A_0 = 10\ m^2$
- standardized level difference as $D_{nT} = D - 10\log\left(\frac{T}{T_0}\right)$, where $T_0 = 0.5\ s$

Regulatory requirements for buildings are often set using single-number quantities calculated according to the UNE EN ISO 717-1. The choice of the descriptor depends on how the regulation is implemented and D_{nT} and $R'$are the most used requirements.

To obtain the single-number quantity, a weighting is performed comparing the measured spectrum with a reference curve. The reference curve is moved in 1 dB steps while the unfavorable deviations is less than 32 dBs. The final value of the curve at 500 Hz is the weighted level (R_w, D_w, D_{nTw}).

To account for the relative loudness perceived by the human ear, some countries add a coefficient C to the weighted level. The resulting index is approximately equal to the A-weighted level.

In Figure 1 we can see some benchmark values from several European countries

Country	Descriptor	Multi-storey housing (dB)	Row housing (dB)
France	D_{nTw}+C	≥ 53	≥ 57
Germany	R'_w	≥ 53	≥ 57
Ireland	D_{nTw}	≥ 53	≥ 53
Italy	R'_w	≥ 50	≥ 50
Netherlands	R'_w +C	≥ 52	≥ 52
Spain	D_{nTw}+C	≥ 50	≥ 50
Switzerland	D_{nTw}+C	≥ 52	≥ 55
United Kingdom	$D_{nTw}+C_{tr}$	≥ 45	≥ 45

Figure 1 Acoustic descriptors for airborne sound insulation between dwellings

We can see that the descriptors and the values change substantially between different countries from the EU. In deliverable 3.14, a list of differences between the European sound insulation regulations will be presented, as part of the methodology.

Sound insulation field measurement

In situ airborne sound insulation measurements are performed according to UNE EN ISO 16283-1. This regulation describes the methodology and the requirements of the equipment used in the measurement. Class 1 microphones are required and different average times are specified depending on the type of position of the microphone (fixed or in movement). In our case we will use fixed positions and the average time required for these measurements is 6 seconds.

The regulation also specifies positions of the microphone, number of measurements required, and the minimum distance between different measurements. All this information has been collected and will be implemented in the final application to guide the user through all the measurement process.

2.2 Use of mobile devices for acoustic applications

The number of smartphones in use is growing day after day and it's expected than by the end of 2016 a quarter of the population will have an smartphone[1]. Smartphones have evolved into powerful computing machines and they include a set of sensors that give them exceptional features. Sensors such as accelerometers, microphones, gyroscopes and GPS make these devices suitable for a great number of measurements.

Nowadays, different mobile device applications make use of this microphone to perform acoustic measurements. For example, several organizations have realized noise pollution monitoring studies using mobile phones with their audio and GPS capabilities. There are also some studies that have evaluated the use of smartphone based sound level meter apps for the detection of workplace noise environment. (Robinson & Tingay, 2014)

Regarding the use of mobile devices for sound insulation applications, a representative sample of the popular smartphones and tablets have been tested in (Kardous & Shaw, 2014) . In this report, different smartphone apps were selected based on the ability to obtain weighted levels and calibrate the microphones. For this study, 10 iOS apps and 62 android apps were examined, using a randomization schedule and applying an analysis of variance to obtain the best suited results. Doing it that way allow the authors to determine the effect of each app, device and noise level when comparing the results with the ones measure with the reference microphone.

The previous study could only be done with iOS devices because there were a low number of android apps and a lack of conformity of features between android devices. The results show important difference between the results obtained with different apps. In this report, it is also highlighted that iPhone3 and 4 gave better performance than its newer generations due to the change of microphone and speech filters. This effect can be palliated with the use of external microphones as iW436.

[1] eMarketer Newsletter, "2 Billion Consumers Worldwide to Get Smart(phones) by 2016", 2014.

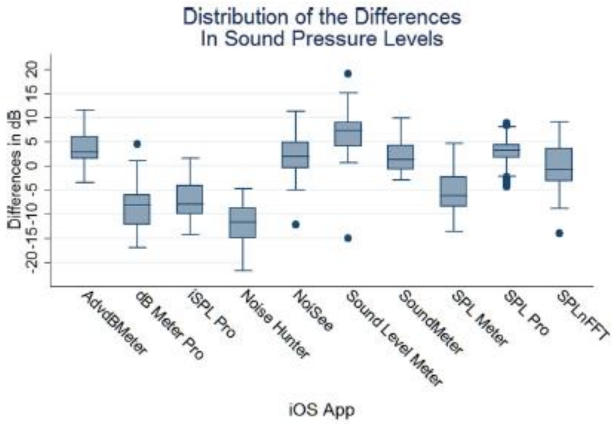


Figure 2 Differences in sound pressure levels obtained from several iOS applications (Robinson, D., & Tingay)

The use of iPhone for sound pressure level measurements has been studied and there has been proved that it is possible to fulfill the measurements according to the regulation (Torres Domínguez, Diego, 2013). In this study, the deviations from the flat frequency response have been studied. The main sources according to the author are: the external or integrated microphone used for measurement, the electrical response of the audio input to the analog - digital converter, and any processed digital signal filtering suffered before reaching the application. The application developed considers the Analogic-Digital Converter as a black-box. The conclusion obtained is that it is possible to implement a correction to the audio input to obtain good enough measurement.

From this study we can conclude that it is possible to use iOS devices for sound insulation measurements and that the design and construction of the applications will have a significant influence in the results. According to this, we have decided to develop our own application and to use an external microphone to ensure a good behavior of our device.

2.3 References

ISO 717-1: 1997 Acoustics-Rating of sound insulation in buildings and of building elements. Part 1. Airborne sound insulation

(ISO 16283-1:2014 Acoustics - Field measurement of sound insulation in buildings and of building elements - Part 1: Airborne sound insulation

Kardous, C. A., & Shaw, P. B. (2014). Evaluation of smarpthone sound measurement applications. *JASA Express Letters* .

Robinson, D., & Tingay, J. (2014). Comparative study of the performance of smartphone-based sound level meter apps, with and without the application of a ½" IEC-61094-4 working standard microphone, to IEC-61672 standard metering equipment in the detection of various problematic workplace n. *Internoise Australia* .

Torres Domínguez, Diego.(2013) *Sonophone: Desarrollo y evaluación de un sonómetro profesional para iOS*. Universidad Politécnica de Madrid

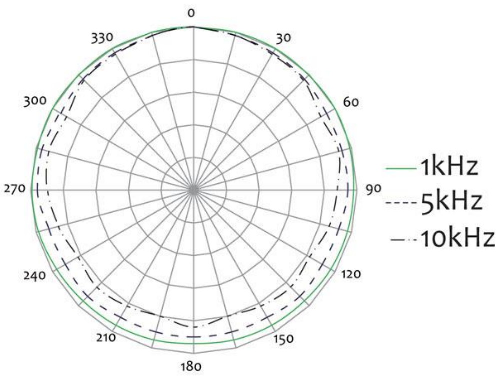
3 Microphone characterization

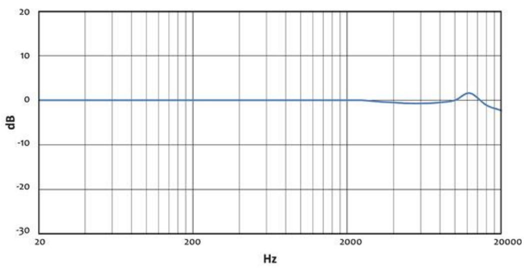
Nowadays there are many external microphones that can be plugged in the audio input of any iOS device. In this project, we were searching for an omnidirectional low-cost microphone with enough precision. According with this specifications, the i436 is the best candidate available, it is marked as compliant with IEC-61672 class 2 and has been suggested that its use improves the accuracy and precision of smartphones measurements[2]

3.1 i436 mic specifications

An external microphone has been used for acoustic measurements. The i436 is a calibrated measurement microphone that complies with the IEC 61672 Class 2 sound level meter standard.

The i436 has an omnidirectional directivity pattern and a flat frequency response. Those are fundamental requirements for the sound insulation measurements





[2] Kardous, Chucri A., Peter B. Shaw, and William J. Murphy. "Evaluation of smartphone sound measurement applications using external microphones—A follow-up study." *The Journal of the Acoustical Society of America* 139.4 (2016): 2036-2036.

Figure 3 Polar pattern and frequency response of i436

Although the microphone specifications seem appropriate for the application, one has to take into account all the electronic analog-digital converter present in the iOS device. The effect of the device in the measurement will affect the frequency response.

3.2 Microphone and device characterization

Although the frequency response of the microphone is flat, the influence of the internal microphone and the electronic part of the devices changes the overall frequency response. In order to be able to calibrate properly the microphone, we must know the electronic structure of the microphone and the iphone ADC.

In Figure 4 we can see the electronic structure of the i436 microphone and the iPhone 6. As we can see , the 4 -pin audio, both regular Left / Right in combination with GND , are outputs. Instead, for the iPhone to detect a microphone, is essential that the audio input of this device has a resistance between GND and MIC-IN of 1K , if this resistance is not included, the microphone is not detected. This is clearly an attenuator to be taken into account , since the output pin for iPhone MIC -IN has a phantom voltage with a negative value of -2.8V . The series resistors have a value of 4K7

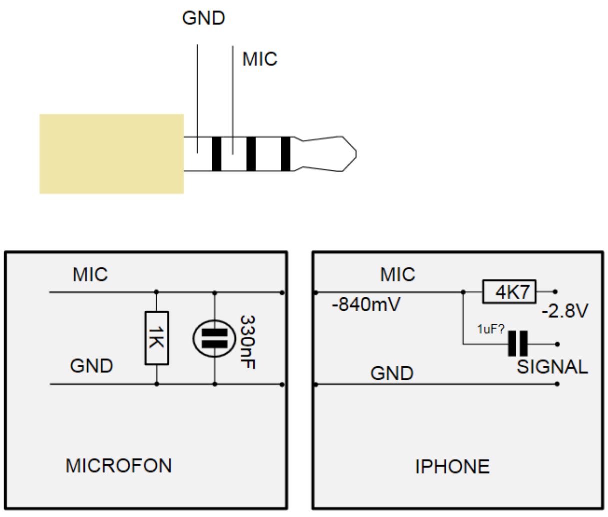


Figure 4 Electronic structure of the devices

The internal ADC iPhone is connected to a 1uF capacitor, which is common in audio systems to uncouple and centre the signal to GND, as shown in Figure 4.

In order to be able to develop an appropriate measurement system compatible with iPhone, a microphone adapter has been developed. The MIC-IN signal is the audio input system mass and it is used as a ground mass. This allows us to work directly with the PC and to record the microphone response appropriately. When working with the microphone connected to the PC we should note that the microphone cannot be touch because it causes an additional signal.

Figure 5 ADC iphone emulator

4 iOS application development

4.1 Signal processing libraries

Eurecat has developed several libraries to process the audio signal in order to obtain the sound pressure level at different frequencies.

Some of these libraries are used both in the PC application and the iPhone application. In Figure 6 we can see the libraries and its connection with the user interface, the external microphone and the calibration file.

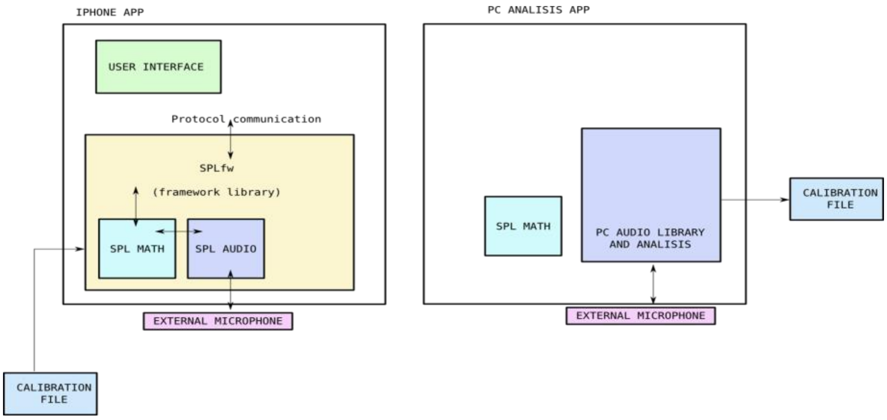


Figure 6: Application scheme

Library specifications:

SPL_MATH

The library implements the following functions

FftInit FFT * (int Points, long sampleRate);

> This routine prepares the FFT (Fast Fourier Transform)[3] used for the computations when the library SPL_AUDIO generates new data to fill the FFT window. In order to have enough accuracy in the low frequency, we need an FFT window of 16384 points.

FftTransform void (* fft fft);

> Once the window is filled with data captured by the microphone, this function is called to process the fast fourier transform.

double FftIntensity (fft fft *, int);

> The calculation will use the SPL to get every FFT point.

prepareSPL int ()

> Initialize the FFT tables and the necessary data to perform the calculation in real time.

SPL_CalcSPL void (int mode, int step)

> Returns the values calculated for the SPL for each appropriate mode and step

SPL_AUDIO

Responsible for the hardware of the iPhone microphone and to begin the process of audio recording. The samples obtained fill the FFT window using the library SPL_MATH

SPLfw

As we can see in Figure 6, the library dedicated to iPhone with a framework format uses a protocol with answers so that you do not need to create a thread part and allows you to obtain partial measures to visualize the process with a graphic display. This library packages all others, and that will help the user interface to make the app more compact and simple.

[3] The Fast Fourier Transform is an algorithm to compute the Discrete Fourier Transform. This analysis converts a signal from its original domain (often time or space) to a representation in the frequency domain and viceversa (*https://en.wikipedia.org/wiki/Fast_Fourier_transform*)

The steps of analysis are:

- *SPL_RESET: initializes the system, if not done yet*
- *SPL_BACKGROUND_NOISE calculates SPL for background noise*
- *SPL_RECEIVER_ROOM calculates SPL from the receiving room*
- *SPL_EMITER_ROOM calculates SPL IA from the room*
- *SPL_REVERB: Measurement of the reverberation time*
- *SPL_AILLAMENT_3OCT: calculation of the study, 3 octaves*
- *SPL_AILLAMENT_GLOB: calculation of the study*
- *SPL_RMS: dynamically allows visually level microphone*

The protocol implements answers for the errors for the partial results and for the final calculations.

Just invoke the function CalculSPL for each iteration step and applied:

+ (Void) CalculSPL: (Steps) step: (int) iteration: (id <Resp>) results;

This library has been implemented in Objective C, except for SPL_MAT, using C-Ansi.

4.2 UI interface

A customized iOS UI interface needs to be programmed to perform sound insulation measurements. This application consists on several modules that will guide the user through the process. In order to validate the results, a first module has been created. This module focus only in the audio signal acquisition.

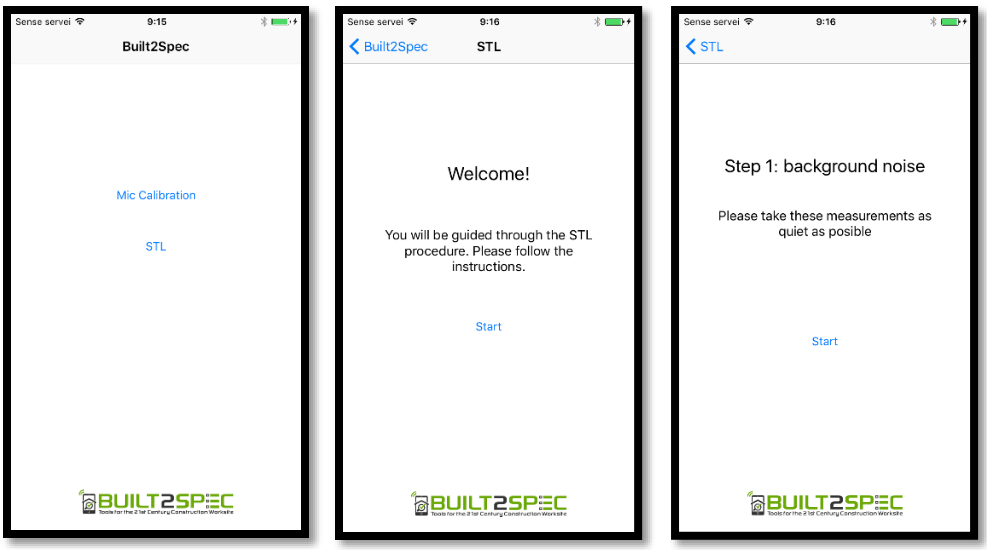


Figure 7 UI interface

The application is programmed in Swift language, the language that Apple created a couple of years ago to replace what had been used to write iOS applications until then, Objective C. However currently both languages are still valid and it is possible to make applications with one or the other.

The application obtains its main functionality from two libraries developed by Eurecat. On the one hand, an audio processing library, which is responsible for acquiring the sound of the microphone and make the signal processing, and the other, an indoor positioning library, which calculates the position that is mobile. Both libraries are completely autonomous and bundled in a framework format, allowing them to be added easily to other projects.

Apart from the libraries developed by Eurecat, also two third party libraries have been used:

• Charts library (https://github.com/danielgindi/Charts): library to create the bar graph to display the audio signal processing

• SQLite library (https://github.com/stephencelis/SQLite.swift): library used internally by the indoor positioning library to access and query a database SQLite format.

The workflow of the application is designed to guide the user through the measurement process. For the audio measurement calibration, only a simple step is needed. In this step the sound pressure level is recorded and averaged every 6 seconds.

Graphically, the application will show the sound pressure level of each third-octave band during the recording process in red colour. Once the measurement is done, the bar colour will change to green and the user will have the option to accept or reject the measurement.

Figure 8 Sound pressure levels displayed

5 Sound pressure level calibration tests

5.1 Recording magnitudes

The main international standard that regulates the sound level meters is the IEC 61672 : 2002 , which it has been harmonized by AENOR (UNE -EN 2005). This regulation has three parts and, part 1 is the one where the specifications are listed.

Two of the most important magnitudes are:

Sound Pressure Level with fast ponderation: Is the RMS value averaged every 125ms in dB. This response is suited to measure sound pressure level with low fluctuation and it is obtained from the acoustic pressure as:

$$L_F = 20\log\frac{1}{\tau}\int_{-\infty}^{T}\frac{\left(p^2(\xi)e^{\frac{(t-\xi)}{\tau}}d\xi\right)^{1/2}}{p_0}$$

Where

- τ is the weighting time constant in seconds
- ξ is a variable of integration from some point in the past to the observation time t
- p is the sound pressure and p0 is the reference sound pressure .

Equivalent Sound Pressure Level: Is the averaged squared pressure between two instants t_1 and t_2. This response is suited to measure sound pressure level with fluctuations and it is obtained from the acoustic pressure as:

$$L_F = 10\log\frac{1}{T}\int_{t_1}^{t_2}\frac{p^2(t)}{p_0^2}$$

Where

- T is the averaging time (in our case, we will use T=6 seconds).
- p(t) is the instantaneous sound pressure and p0 is the reference sound pressure .

For sound insulation measurements is mandatory to obtain the sound pressure lever for the frequency range between 100 Hz and 3150 Hz. The equipment that performs this operation is called a Real Time Analyser and most of the professional commercial sound level meters are also real time analysers.

Two common spectral bands are used in acoustic measurements, octave and third-octave bands. In our case, the RTA will only work with third-octave bands that give us more information.

The central frequency for each band is:

100 Hz	125 Hz	160 Hz	200 Hz	250 Hz	315 Hz	400 Hz	500 Hz
630 Hz	800 Hz	1000 Hz	1250 Hz	1600 Hz	2000 Hz	2500 Hz	3150 Hz

5.2 Analysis and characterization software

A PC software has been developed to analyse and characterize the audio input. The program has been developed to perform specific measures. The software allows to connect the audio output to an amplifier and generates several functions to characterize both the sender and the receiver.

In Figure 9 we can see a screen capture of the software developed.

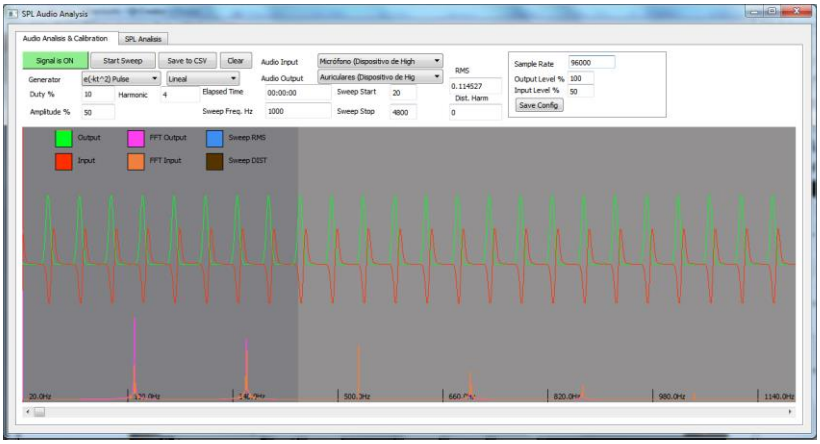


Figure 9 PC software screen capture (Tab 1)

The software has two tabs.

The first tab allows us to calibrate any microphone with a speaker system or amplified specified signals. The software displays also the behavior in frequency of the microphone

It has five types of source signals:

- sine wave
- square wave
- Gaussian pulse: exp (-kt ^ 2)
- Gaussian Wavelet: e (-kt ^ 2) sin (pi · f · t).
- White Noise
- Pink Noise

The software allows us to change the sampling frequency, the microphone levels and the type of source signal. It can also make frequency sweeps in order to do more precise measurements. Even you can add other parameters such as harmonic distortion analysis and others.

White and pink noise signals are used to have all frequencies tested in a single measurement. The problem is that speaker systems are usually not able to properly follow the white noise due to its own inertial system. Normal speakers do not have sufficient dynamic range to follow the signal. The same effect occurs with pink noise. For this measurement, a special speaker is needed (dodecahedric speaker).

The only way to obtain accurate results with any speaker is to make a sweep on all frequencies with a sinusoidal signal. Or to use an hybrid method (pulses that have a wide distribution in space and frequency and that allow to measure many frequencies simultaneously with precision). Examples of these pulses are the Gaussian pulses or wavelets.

The second tab of the application analysis shows the sound pressure level.

This information is the same than the SPL bars displayed in the iPhone application.

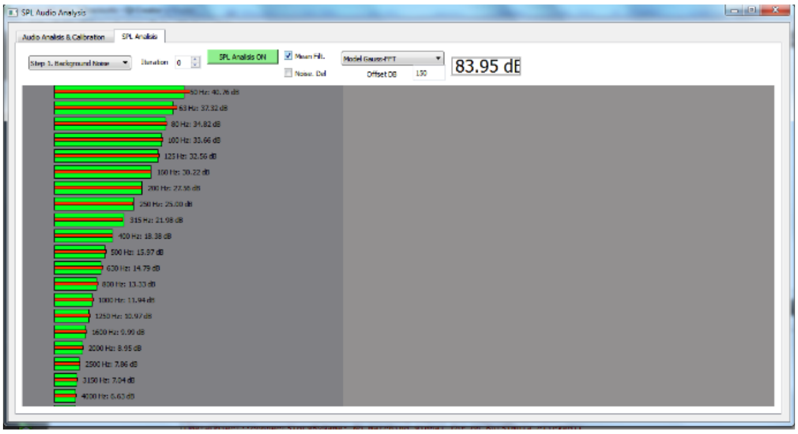


Figure 10 PC software screen capture (Tab 2)

In Figure 10 we see the SPL in third-octave bands.

In order to obtain the frequency response in third octave bands, we can use to different approaches:

- Use resonant filters for each frequency bandwidth of 70% resonance
- Use an FFT and then perform a gauss bell weighted filter. We have implemented this method in our program.

In Figure 11 we can see a description of the method used to compute the SPL for each third octabe band.

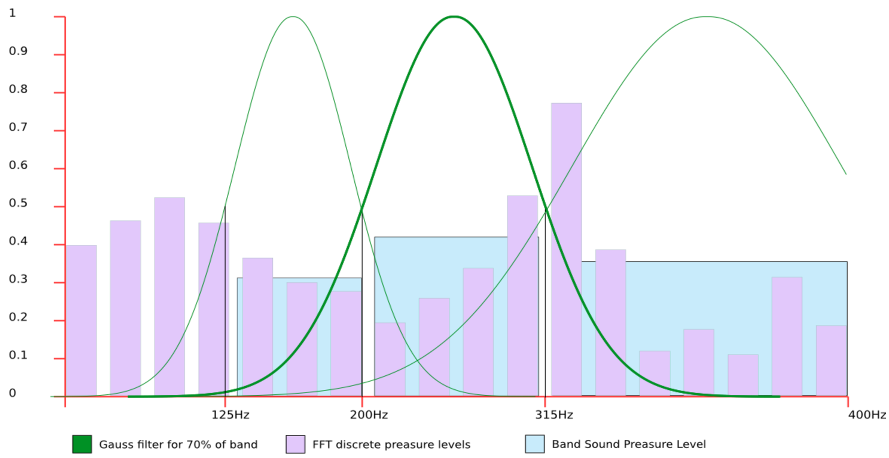


Figure 11 Computation model using an FFT

The problem with the FFT method is that the window should have enough points in the low frequency range. The fact that the FFT is linear but the bands to be weighted are logarithmic make that in the high frequency range we have a lot of frequency points measured. This could be computationally expensive.

In order to obtain the frequency range inside each third-octave frequency band, we could have all the values pre-calculated in tables or calculate the range with a filter using a discrete equation of the gauss bell.

In this case we have chosen to apply the filters.

5.3 Sound pressure level measurement

In order to calibrate the microphone, a set of experimental tests have been performed. The equipment used in the tests to evaluate the accuracy of the mobile device is compliant with ISO 16283.

Pink noise spectrum has been generated with a CESVA dodecahedrical loudspeaker (AP600). The SPL obtained has been recorded both with the microphone i436 and a Class 1 sound level meter (CESVA SC-310). The specifications of the sound level meter used are listed in ANNEX A

The recording facility selected is a reverberant room in Eurecat-Cerdanyola building. The high reverberation of the room assure us a diffuse field and reduces the possible errors due to the distance between the microphones. To minimize these errors, multiple measurements have been performed with the microphones located at different points in the room.

Figure 12 Experimental tests with pink noise

Five different levels of noise have been recorded (speaker levels of -10dB, -15dB,-20dB,-25dB and -30 dB) during intervals of 10 seconds. For each level, we have computed the Sound Pressure Level with fast weighting and the Equivalent Sound Pressure Level.

The results have been recorded with a customized PC interface for the i436 microphone (see 5.2) and with the commercial software Capture Sound Studio for the SC-310 sound level meter.

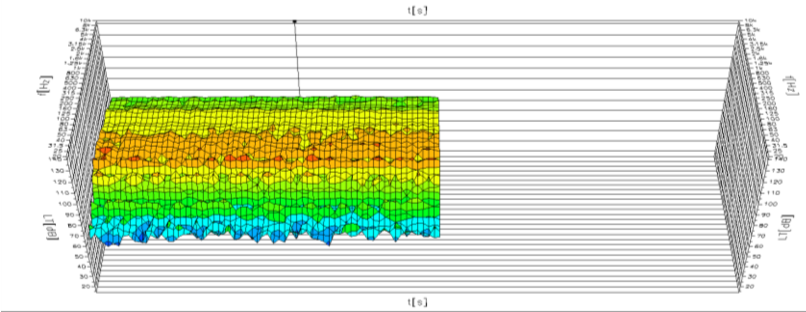


Figure 13Measurements obtained with CESVA Capture Sound Studio

5.4 Microphone calibration

From the study of the sound pressure level test done, we have seen that it is possible to make a calibration of the i436 microphone using a second degree polynomial. For this purpose, we will need to record 3 different points at different sound levels with both the mobile device and the sound level meter. These points will be saved in a calibration file that will be opened by the Built2Spec mobile interface when a calibration is needed.

In the picture below we can see the differences between the commercial sound level meter and the iOS device with the microphone without any calibration implemented:

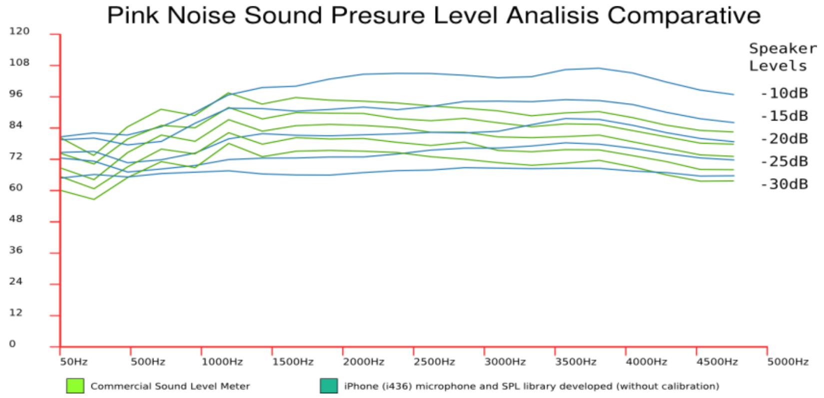


Figure 14 Sound Pressure Level comparative

As we can see in the graph, the pink noise emitted is captured perfectly with the commercial sound level meter because the lines are almost parallel. However in our device, without calibrating the system, we get a series of response curves that depend on the behaviour of the microphone, which is not ideal.

Near 50 Hz the calibration is much difficult to do because the microphone response is not good in this frequency range.

The following image shows the calibration should be done for a specific frequency of 315Hz:

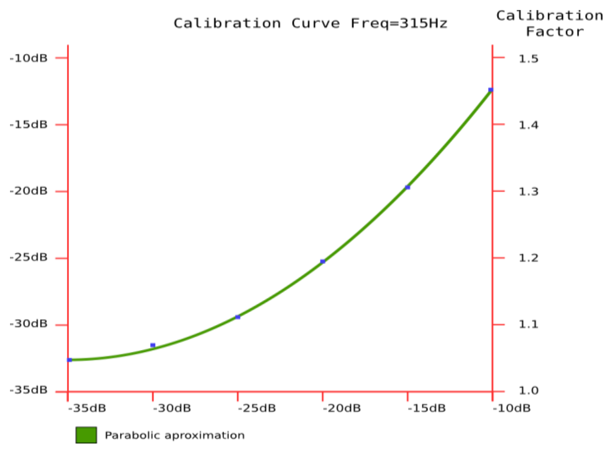


Figure 15 Calibration curve for 315 Hz

Except in the area of 50Hz , the calibration factor can be approximated with a parabola, which means that measuring three different sound levels, we could calibrate each microphone-iPhone system. This curve is completely different for each frequency. Therefore, a good method is calibrate it with pink noise, as we have done in the study.

6 Conclusions

Some of the conclusions obtained from the study are:

- It is possible to calibrate the mobile device to measure correctly the sound pressure level using a 3-point curve in each third octave frequency band.
- Both the UI interface and the signal processing functions have been implemented in the Built2Spec mobile application and are working properly.
- The electronic characterization of the mobile device audio input and the microphone allows us to test the equipment with multiple input functions in a PC environment.
- The external i436 microphone seems to be a good candidate to perform sound insulation measurements.

On the other hand, there is some work remaining in the next months

- It would be necessary to do a second set of test to measure the error obtained with the iOS device once the calibration is applied. The results from these tests will indicate us the limitations of the mobile device.

Annex A SC-310 Specifications

STANDARDS AND SPECIFICATIONS

Complies with the following standards:

- EN 60651:94 (A1:94) (A2 :01) type 1, EN 60804:00 type 1, EN 61260:95 (A1:01) type 1
- IEC 60651:01 type 1, IEC 60804:00 type1, IEC 61260:95 (A1:01) type1
- ANSI S1.4:83 (A1 :85) type 1, ANSI S1.43:97 type 1, ANSI S1.11:86
- CE mark. Complies with 73/23/CEE and CEM 89/336/CEE low-tension regulations, the latter amended by 93/68/CEE.

MEASUREMENT RANGE

L_F, L_S, L_I, L_T and L_t

Indicator limits: 0 – 157 dB

C-130 + PA-13

	A	C	Z
Primary range:			
Upper limit	120	120	120
Lower limit	30	32	38
Measurement range			
Upper limit	137	137	137
Crest factor 3:	130	130	130
Crest factor 5:	126	126	126
Crest factor 10:	120	120	120
Lower limit :	24	26	31

C-250 + PA-14

	A	C	Z
Primary range:			
Upper limit	120	120	120
Lower limit	28	29	34
Measurement range			
Upper limit	137	137	137
Crest factor 3:	130	130	130
Crest factor 5:	126	126	126
Crest factor 10:	120	120	120
Lower limit :	22	22	27

L_{peak}

Indicator limits: 0 – 160 dB

ELECTRICAL NOISE

C-130 + PA-13

Electrical noise:	A	C	Z
Maximum	14.4	16.8	21.9
Typical	13.4	15.8	20.0
Total noise (elec. + thermal of microphone)			
Maximum	19.6	21.1	25.9
Typical	17.6	19.0	22.0

C-250 + PA-14

Electrical noise:	A	C	Z
Maximum	9.4	10.5	18.5
Typical	8.6	8.8	16.3
Total Noise (elec. + thermal of microphone)			
Maximum	16.6	16.8	22.0
Typical	15.7	15.1	18.8

FREQUENCY WEIGHTING

Complies with the EN 60651 type 1 standard

Weightings A, C and Z

AC OUTPUT

Frequency weighting: linear

Sensitivity to 137 dB and 1 kHz (Gain = 0dB): 6.5 Vrms (max)

Upper limit: 8.1 Vrms (typical)

Output impedance: 100 Ω

Gain: 0 and 40 ± 0.2 dB

MEMORY

64 Mbytes

MICROPHONE

Model: CESVA C-130

Condenser microphone ½ "

Polarized: 200 V

Nominal capacity: 22.5 pF

Nominal sensitivity: 17.5 mV/Pa in reference conditions.

Preamplifier: PA-13

Model: CESVA C-250

Condenser microphone ½ "

Polarized: 0 V

Nominal capacity: 17.0 pF

Nominal sensitivity: 46.4 mV/Pa in reference conditions.

Preamplifier: PA-14

TIME WEIGHTING

L_F, L_S, L_I according to class 1 tolerances

PARAMETERS

See previous table| Resolution: 0.1dB

OCTAVE FILTERS

Type 1 according to IEC 61260:95/ A1:01

Nominal octave bands mid-freq.: 31.5, 63, 125, 250, 500, 1000, 2000, 4000, 8000, 16000 Hz

THIRD OCTAVE FILTERS

Type 1 according to IEC 61260:95/ A1:01

Nominal third octave bands mid-freq.: 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000 Hz

INFLUENCE OF HUMIDITY

Operation range: 30 to 90 %

Maximum error at 30%<R.H.<90%, 40 °C and 1 kHz: 0.5 dB

Storage without batteries: < 93 %

I EFFECTS OF MAGNETIC FIELDS

In an 80 A/m magnetic field (1 oersted) at 50 Hz, all gives a reading of less than 25 dB(A) is given.

INFLUENCE OF TEMPERATURE

Operation range: -10 to +50ºC

Maximum error (-10 to +50ºC): 0.5 dB

Storage without batteries: -20 to +60ºC

EFFECTS OF VIBRATIONS

For frequencies between 20 and 1000 Hz and 1 m/s^2: <75 dB(A)

BATTERY

Battery: 2 batteries 1.5 V size AA:

Battery life with continuous use:

Sound Level Meter mode : 15 h

Spectrum Analyzer mode 1/1: 13 h

Spectrum Analyzer mode 1/3: 11.5 h

Mains feeder: AM240 (EU) or AM241(USA)

DIMENSIONS and WEIGHT

Dimensions:

341 x 82 x 19 mm

Weight:	with battery	550 g
	without battery	500 g

CESVA *instruments, s.l.* reserves the right to change specifications and accessories without notice.