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Software for Measuring Acoustic Parameters in Open-plan Offices

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Abstract— **This paper presents a development of software for the measurement and estimation of the quality of acoustic design in open-plan offices. The measurement method is defined by a standard and measurements should be performed in a large number of different spatial positions in order to measure and determine the acoustic quality of the surroundings of as many workstations as possible. The presented software significantly simplifies the process of analyzing open-plan offices by processing the signal at the time of measurement. Furthermore, it allows for the storing of measured signals and parameter values, which can be of importance for further analysis or repeated calculations. The presented procedure leads to the reduction of the necessary hardware elements for completing the analysis. All that is needed for measurement are a sound source, a microphone, a sound card and a computer. As a result, the user obtains a detailed report of the characteristics of the analyzed office space and a descriptive mark which depends on the values of the calculated parameters.**

Keywords-noise; privacy index; measurement; open-plan office; software; STI;

I. INTRODUCTION

„*Open-plan*", "*open space*", or „*open*" offices are rooms in which a large number of people work and physically share a volume of space. The advantage of this type of office is the quality of workspace, which facilitates the interaction between co-workers and allows for the smooth flow of information throughout the office. From the architectural point of view, it allows for the flexibility of workspace organization and an adaptation of workstations for temporary or long-term job tasks. During corporate building project planning, when terms and conditions of future users are still unknown, this represents one of the most important advantages of this type of workspace.

Open-plan offices have their downsides, especially in terms of comfort level, since workers are constantly exposed to the activities of co-workers in their immediate surroundings. The most influential distraction of all is the sound generated by other people and the lack of speech privacy within the workstation. Distractions caused by the surrounding sounds affect the workers' attention span, resulting in a reduced level of concentration and work efficiency. Speech in open-plan offices can be a distraction to those that are not part of the conversation. Those involved in conversation can experience a lack of speech privacy and the spreading of possibly confidential information. Therefore, the interior design of open-plan offices, apart from providing functionality and aesthetics, has to provide a high enough comfort level and ensure a quality way of handling the sound throughout the entire room. These types of spaces demand a design which will minimize distractions and allow for better concentration on work [1].

The acoustic quality of open-plan offices is quantified using several numerical parameters, which are defined by standard 3382-3 [2]. During measurements, parameters defined by other standards are also being calculated [3-6]. Two parameters of significant importance are defined: distraction distance [7] and privacy distance [1]. Both of them are based on the evaluation of the speech intelligibility of a distant speaker. Privacy distance is defined as the distance from a speaker at which speech becomes completely incomprehensible (value of *Speech Transmission Index* – STI is less than 0.2). The distraction distance is defined in a similar way. It defines the distance at which speech is comprehensible and poses a distraction to others (value of STI parameter is less than 0.5). The aforementioned standard defines the following parameters:

- Spatial sound distribution of the A-weighted sound pressure level of speech [9]. This curve represents how the A-weighted sound pressure level decreases as a function of the distance from the sound source emitting noise with the sound power spectrum of normal speech.
- Spatial decay rate of speech $D_{2,S}$, represents the rate of spatial decay of A-weighted sound pressure level of speech, when doubling the distance.
- A-weighted sound pressure level of speech at a distance of 4 meters, $L_{p,A,S,4m}$.

This paper is a revised and expanded version of the paper presented at the 18th International Symposium INFOTEH-JAHORINA 2019 [22]

- Speech transmission index, *STI* [10], physical quantity representing the transmission quality of speech with respect to intelligibility.
- Spatial sound distribution of the speech transmission index, represents the curve which shows how the speech transmission index decreases from a reference sound source when distance increases.
- Distraction distance r^D - the distance from the speaker at which the speech transmission index falls under 0.5, and is expressed in meters [7], [11].
- Privacy distance r_P - distance from the speaker at which the speech transmission index falls below 0.2, and is expressed in meters [7], [11].
- Background noise level $L_{p,B}$, represents the sound pressure level in octave bands present at the workstation during working hours with people absent.

All parameters which are not expressed as single number quantities shall be calculated and measured in octave bands from 125 Hz to 8000 Hz [12]. During the analysis of the offices, it is necessary to determine values of the parameters in a higher number of points distributed over the office floor in order to cover a bigger surface, which results in a more precise spatial resolution of the needed parameters. To complete the measurement process defined by the standard, the following equipment is necessary: a generator of the pink noise, a loudspeaker, a background noise level meter, a microphone and a computer with STI measurement software. This procedure is very time-consuming and susceptible to errors, because of the usage of data obtained from a large number of measurements and different tools. The evaluation of the acoustic quality of the office design can be used for improvement and adaptation of the existing situation in order to achieve the required quality [1]. By keeping the measured signals and calculated parameters after the analysis of the office, the degree of improvement can be determined after the appropriate acoustic intervention.

The goal of this paper is the development of a software solution with the task of incorporating all the necessary steps required for the measurement and analysis of a single office, in order to facilitate and speed up the process, as well as to reduce calculation errors and the quantity of necessary hardware equipment. The presented system (software) for measuring the acoustic parameters of offices does not exist on the market. The advantages of this software are a reduced amount of necessary equipment, significant time saving due to the automated process of calculation and reduced error rate while processing the measured data. This software is developed in C++ programming language. Beside the standard library, an open source JUCE framework was used, which contains a large number of functions for digital sound processing, as well as functions for creating graphical user interfaces. In this paper, the software's functionalities and workflow are presented. Finally, the results from the analyses of two offices with different acoustic specifications will be shown. These examples will be used to discuss how different methods of acoustic treatments can impact the overall acoustic quality of an office space.

The paper is organized as follows. In the second chapter, all the parameters defined by the standard are listed and explained in detail, as well as the method of measuring and calculating each one of them. The third chapter presents the software which is used to ease the process of evaluation of acoustic quality of an office space, its functionalities and user interface. In the fourth chapter, experimental results are shown, acquired from one office space, followed by a discussion of the results. In the end, a conclusion is given on the advantages of using this software for the analysis of open-plan offices.

II. METHOD OF CALCULATION OF THE ACOUSTIC PARAMETERS

During the analysis of acoustic quality in an open-plan office, measurements must be made with present furniture, but without people present in the office [13]. All persons, except the ones responsible for measurement and analysis, must vacate the office because analyses performed in such conditions are the most credible. The presence of other people during the analysis leads to an improvement of the results, because they have an impact on the spreading of sound as every person introduces a corresponding level of sound absorption [8]. All devices which create background noise, such as the HVAC systems, must be powered on during the analysis process. By turning those devices off, an error in evaluation will occur as the measured level of background noise will be lower than the usual level during working hours.

Measurements shall be performed on many different positions located on lines that go through workstations. Those lines can be straight or curved. It is recommended to position the measurement points on a straight line whenever it is possible. Larger office spaces often consist of two or more zones where there are different types of ceiling materials or the furniture design differs significantly. In such cases, the measurements should preferably be made in each zone and the measurement lines should not cross two or more different zones. The preferred number of successive measurement positions in a line is 6 to 10, while the minimum number of needed positions is 4[1]. The first measurement position shall be located at the nearest workstation on the line. The positions of the loudspeaker and the microphone shall be at least 0.5 m from tables and at least 2.0 m from walls and other reflecting surfaces. The loudspeaker and the microphone shall be placed at a height of 1.2 m above the floor. At every measurement point, four measurements are made:

- Sound pressure level in octave bands of pink noise, $L_{p,Ls}$;
- *STI*;
- Background noise level in octave bands, $L_{p,B}$;
- Distance to the sound source, *r*.

The integration time should be at least 10 seconds. While calculating the single number quantities, a table with values of sound pressure level of normal speech in octave bands is used. The table contains levels on a distance of 1 m from the sound source in the free field, $L_{p,S,1m}$. The sound power level of a sound source should be sufficiently high in each octave band so that the sound pressure level exceeds the background noise level by 6 dB at the most distant measurement point. The signal reproduced by the loudspeaker is pink noise. The sound pressure level at a distance of 1 m from the acoustic center of the loudspeaker in a free field, $L_{p,Ls,1m}$ in decibels, is:

$$L_{p,Ls,1m,i} = L_{w,Ls,i} + 10\log_{10}\frac{1}{4\pi * 1.0^2} \approx L_{w,Ls,i} - 11\,\text{dB}. \tag{1}$$

After that, it is necessary to measure the sound pressure level of pink noise which is being reproduced from the sound source. Measurements have to be taken at every measurement position on the corresponding measurement line. The value of $L_{p,Ls,n,i}$, where n represents an index of the measurement position, is obtained. The following formula is used to determine the attenuation in decibels in every octave band of interest:

$$D_{n,i} = L_{p,Ls,1m,i} - L_{p,Ls,n,i}. \tag{2}$$

The calculated attenuation is valid for any sound power level of the loudspeaker, and therefore it is applied to the sound power level of speech. The value of the sound pressure level of speech is taken from the aforementioned table [1], [5], and it is being attenuated by the obtained attenuation:

$$L_{p,S,n,i} = L_{p,Ls,1m,i} - D_{n,i}. \tag{3}$$

Finally, the A-weighted speech level in position n is obtained by adding the values for A-weighting at each octave band and summing on energy basis:

$$L_{p,A,S,n,i} = 10\log_{10}\left(\sum_{i=1}^{7} 10^{\frac{L_{p,S,n,i}+A_i}{10}}\right). \tag{4}$$

In order to calculate the parameter that describes the acoustic quality of the space - the spatial decay rate of speech $D_{2,S}$, it is necessary to measure background noise levels in every octave band of interest at each measurement position as well as the *STI* parameter. The determination of $D_{2,S}$ is made from the results at measurement positions at distances within the range from 2 m to 16 m from the sound source. The spatial decay rate of speech is then calculated using the least squares method:

$$D_{2,S} = -\log_{10}\frac{\left(N\sum_{n=1}^{N}\left[L_{p,A,S,n}\log_{10}\left(\frac{r_n}{r_0}\right)\right] - \sum_{n=1}^{N}\left[L_{p,A,S,n}\sum_{n=1}^{N}\log_{10}\left(\frac{r_n}{r_0}\right)\right]\right)}{N\sum_{n=1}^{N}\left[\log_{10}\left(\frac{r_n}{r_0}\right)\right]^2 - \left[\sum_{n=1}^{N}\log_{10}\left(\frac{r_n}{r_0}\right)\right]^2}, \tag{5}$$

where n is the index number of the current measurement position, N is the total number of measurement positions on the corresponding line, r_n is the distance to the measurement position n, and r_0 is the reference distance which is 1 m. The process of calculating and measuring is the same for all of the defined measurement lines. In the following table, the values of some of the parameters are shown for the offices with good and poor acoustic quality.

TABLE I. VALUES OF PARAMETERS FOR OFFICES WITH GOOD AND POOR ACOUSTIC QUALITY

Parameters	$D_{2,S}$	$L_{p,A,S,4m}$	r_D
Poor acoustics	< 5 dB	> 50 dB	>10 m
Good acoustics	< 7 dB	< 48 dB	< 5 m

III. SOFTWARE REALIZATION FOR THE CALCULATION OF ACOUSTIC PARAMETERS

The analysis and evaluation of the acoustic qualities of an office requires the connecting of the microphone and the sound source to a computer and the completion of the whole process using the software which is presented in this chapter. The calculation of the parameters is possible right after measuring all the necessary signals at every measurement position. Before starting the recording process, it is necessary to set up measurement preferences. Each one of them is shown in the follow-up.

On Figure 1, a window for creating a new project is shown. It consists of all necessary parameters that are required to be defined before further analysis. In addition to naming the project and saving the folder destination, an audio input and output must be chosen. It is important that the audio input and output have the same sampling frequency, otherwise it will not be possible to create a new project. The recording length defines the length of each recorded signal in a project, in seconds, and it can be selected only once for each project. Audio input and output can be changed at any time by clicking the *I/O Options* button.

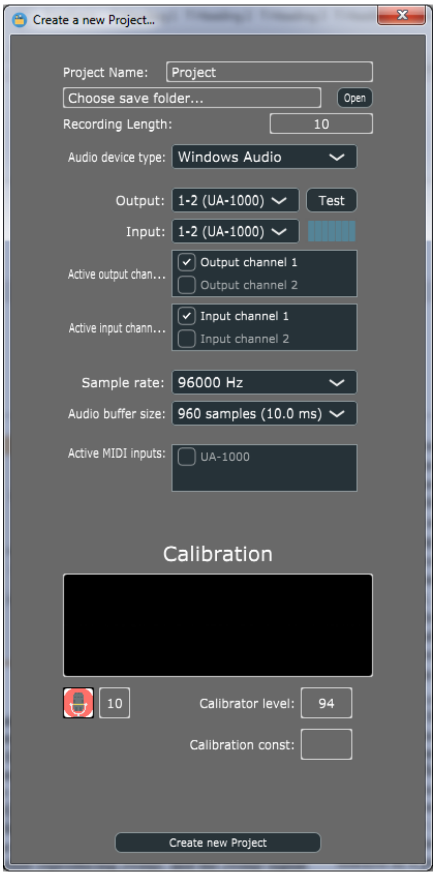

Figure 1. Create new project with necessary parameters

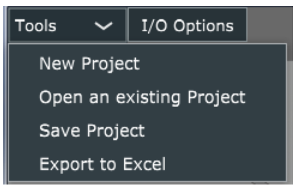

Figure 2. Menu with project options

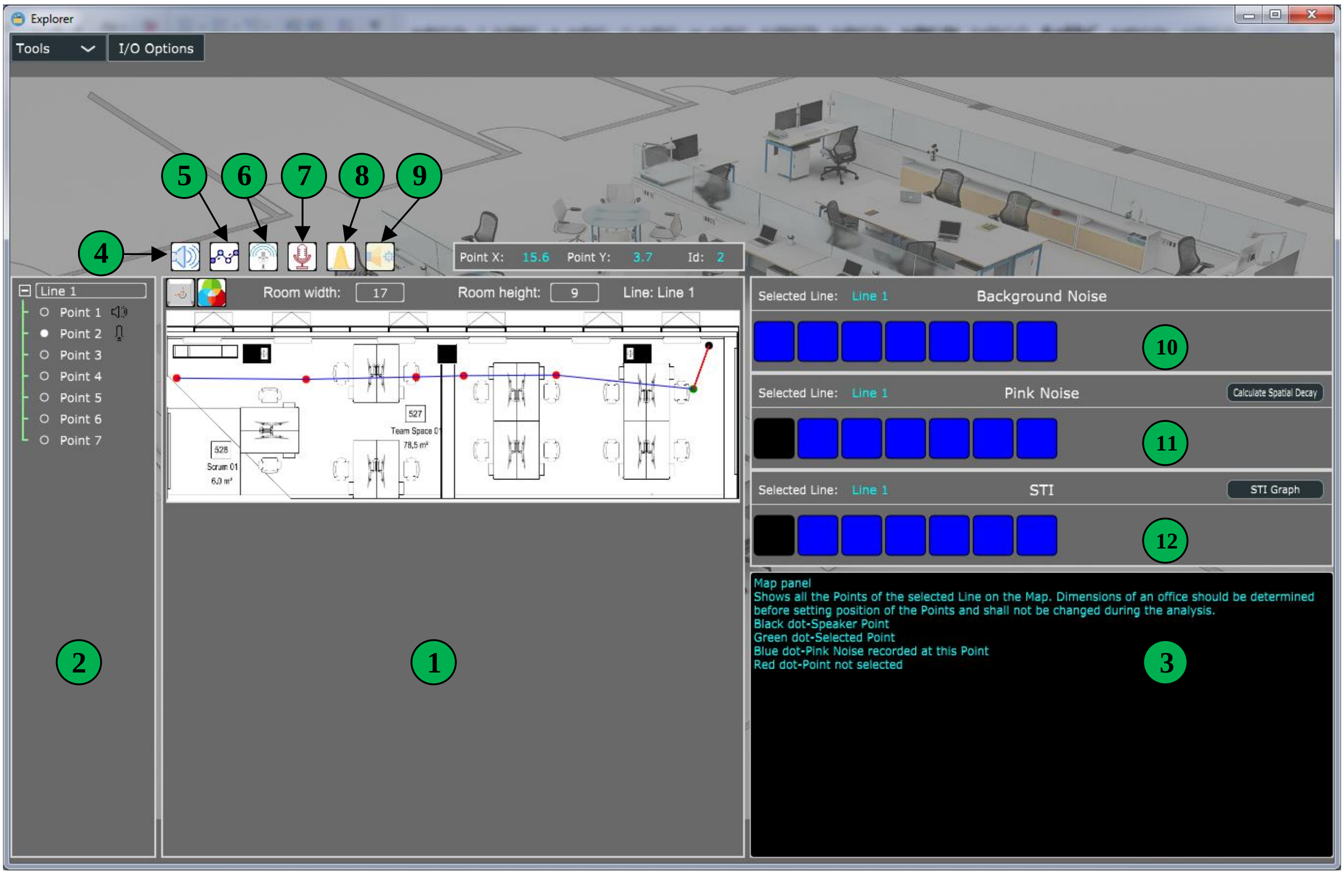

Figure 3. The main window with enumerated panels

The system must then be calibrated. This process is done before the recording or any calculations, since the results would otherwise be unreliable. Calibration ensures that the recorded input is the absolute level of the measured sound pressure registered by the microphone. The calibrator is placed on the microphone [5] and the output power of the calibrator in decibels is entered as an input in the preference window. Calibration must be performed at the begining of the process. Afterwards, the values of hardware knobs must stay unaltered. In Figure 2, a menu with options to save a project or load an existing one is shown. The files that are generated while saving a project have a specific .etf extension. Export to Excel offers the possibility of exporting wanted measurement lines to a predefined Excel table which can be used for further analysis.

Figure 3 shows the main software window. Each panel is numbered 1 to 12 and each will be individually explained. Before the start of the recording process, and after the successful creation of a project, it is necessary to load a map of the room being analyzed. Panel 1 shows the map with all of the relevant information. Room dimensions have to be defined, in meters, and after that the map can be loaded as an image with commonly used image extensions like .png or .jpg.

Panel 2 consists of measurement lines and their measurement points. It is possible, by right clicking on a panel, to add new lines. Each line has its name, which can be changed anytime. It is also possible to copy and paste points from one line to another. Right clicking on a line displays a menu for adding new points to that line. For each measurement point it is required to define its coordinates on a map. This can be done manually by typing the values or with a mouse cursor by clicking on a map. These options are displayed in Figure 4.

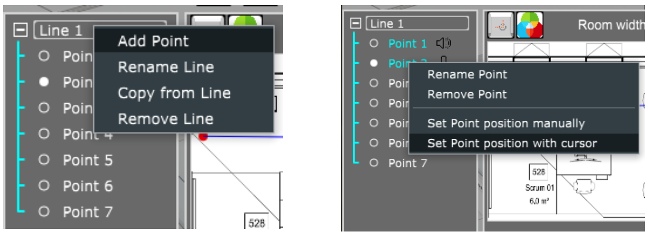

Figure 4. Options for adding new points to a line

The software uses a set of different colours to define and mark different types of measurement positions during the process of measurement. A green point is displayed on the map when it is selected. All other points, in that case, are displayed in red colour. When a point is being selected, or marked as a point in which the speaker is located, it becomes black. Speaker position is defined by clicking on button number 4, shown in Figure 3, after selecting the wanted point. After that selection, the rest of the points can be displayed either in blue or red colour. If, at a point, pink noise had already been recorded, which was reproduced from a sound source located in a selected point, the corresponding black point, the point will be coloured blue, otherwise it will be red and it implies that pink noise has yet to be recorded in that position.

When a certain number of points are added to a line, options for further calculations are displayed. As already mentioned, in every measurement position the background noise level has to be recorded and it has to be done using the button *Record Background Noise*, marked with number 6 in Figure 3. After finishing the recording it is possible to repeat the recording if the obtained signal is not suitable for further processing. By clicking on the button *Select Point as Speaker*, the selected point, of which the information is shown in the *Point Info* panel, is being marked as a position where the sound source is located. The loudspeaker should be placed at the forementioned height and at the location of the marked point, after which the microphone should be positioned at the location of the next position on the line. The microphone has to be located in all other measurement positions and recordings of the background noise level, pink noise and impulse response have to be made, which leads to the calculation of STI value in those positions. The window for recording pink noise is shown by clicking on button number 7, Figure 3. The Speech Transmission Index, at each measurement point, can be calculated by clicking on button number 8, Figure 3. The impulse response is used for the calculation of the speech transmission index, *STI* parameter. The process of the impulse response recording can be performed using an MLS signal [14, 15] or *sweep* signal [16]. The length of the recording, the number of averaging, and the number of recordings can be changed. Impulse response is calculated as a deconvolution of the recorded signal while reproducing *sweep*, and the *sweep* signal itself [16].

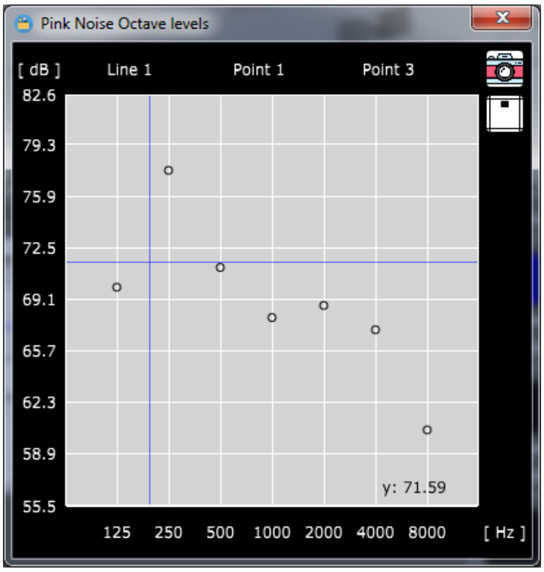

Figure 5. Graph showing recorded pink noise levels

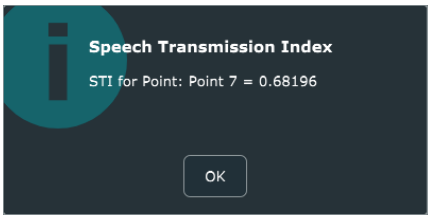

Figure 6. A display of the STI value for the selected point

Panels 10, 11 and 12 hold the information of the state of each measurement point on the selected line. Each of the boxes designates one of the points. The box is blue if a corresponding signal is recorded for that position. In the order from top to bottom, the panels correspond to background noise, pink noise and STI values. The black box represents the point at which the speaker is located. By clicking on a box, a graph with signal information is shown as seen in Figure 5. Clicking on any of the boxes on panel 12, displays the value of the STI parameter calculated for that position, as seen in Figure 6. When all of the necessary signals are recorded and all of the boxes on panels 10 and 11 are coloured, it is possible to calculate the spatial decay by clicking on the Calculate Spatial Decay button. Similarly, after calculating the STI parameter for each of the points on the selected line, clicking on the STI Graph button displays the graph with STI curve, privacy and distraction distance.

Once measurements are completed in every measurement position, the spatial decay rate of speech has to be calculated for the selected point, $D_{2,S}$. Points in which the recording has not been performed are easily noticed as they are not displayed in blue color. When the spatial decay rate of speech is calculated for one point, the next one is selected as the sound source position and the whole process repeats. It is not necessary to perform a recording at the position in which the sound source was already located as it is assumed that the attenuation between two points does not depend on the positions of the speaker and the microphone. All recorded signals are saved in a folder predefined by the user and can be used for repeated calculations or some other type of research. Furthermore, all relevant data is saved as well: levels of background noise, spatial decay rates and STIs.

IV. USING THE SOFTWARE FOR THE ANALYSIS OF ACOUSTIC PARAMETERS

The workflow of the software had to verified by finishing the measurement process in a real situation. The measurements were performed in two rooms. A map of the first office, with defined measurement points is shown in Figure 3, while the second office is displayed in Figure 7. The measurement procedure was performed in order to check the workflow of the program from the technical viewpoint. For the signal recording, microphone *MiniSPL Measurement Microphone* [17] was used, and an external sound card *Steinberg UR22* [18]. An omnidirectional loudspeaker was used as a sound source [19].

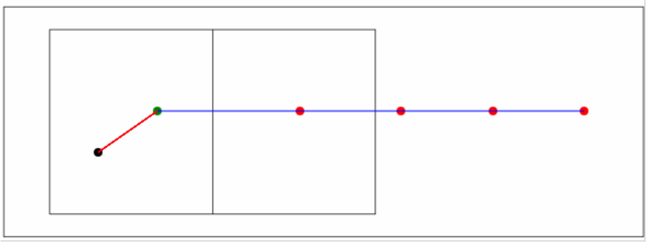

Figure 7. Point positions on the map of the second room

The measurement in the first office was performed on one straight line, which consisted of 6 measurement positions. Distances between the points can vary, while the distance between the first and the last point in the line is around 16 meters. Distance between the points are from 1 m to 3 m. The line crosses five workstations. Displayed results are related to a case where the sound speaker was located at the first measurement position as it is shown in Figure 8. In the rest of the points, the described measurements were performed. In Figure 8, a spatial decay rate of the speech is shown, acquired by measuring pink noise at every position.

The recorded pink noise was scaled with A-weights and averaged for every point. Decay of speech level can be easily noticed as the distance from the sound source increases, which is expected. The average level of the background noise at every position is also shown on the same figure and it does not vary much throughout the whole office. The level of the background noise is in the range of the average everyday level of the noise. Sound power level fulfills the requirement of being 6 dB above the level of the background noise in the furthest point from the speaker. It is noticeable that the gradient of the displayed line which shows how sound pressure level is decaying as the distance increases, is not steep enough to fulfill the requirements of good acoustics in an open plan office. This is caused by the poor acoustical treatment of the room interior, as well as the distance between workstations being too low and sound pressure level not decaying enough during propagation.

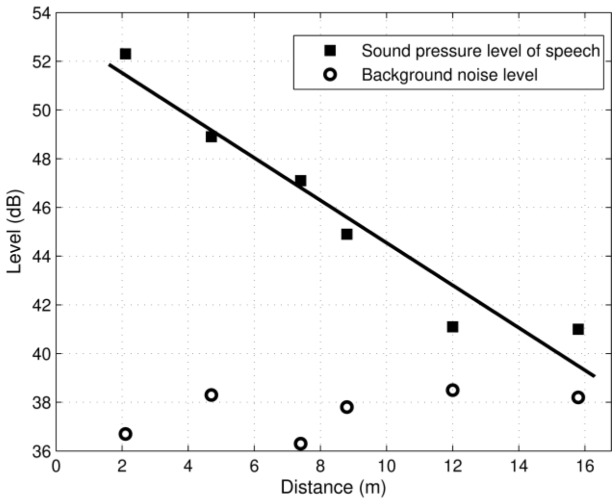

Figure 8. Figure with spatial decay rate of speech with levels of background noise

All of those values are of importance for calculation of $D_{2,S}$. The value of this parameter, in decibels, is also shown in the same figure. Parameter L_{4m} is 49.7 dB. Analyzing the value for $D_{2,S}$=4.1 dB, based on the Table 1, this room has bad acoustic quality.

Figure 9 shows the value of parameter STI at every position. Values of this parameter vary from 0 to 1, and in this room this parameter has quite a high value. Speech transmission index is important for calculating the two very important parameters for the evaluation of the open plan offices: distraction and privacy distance.

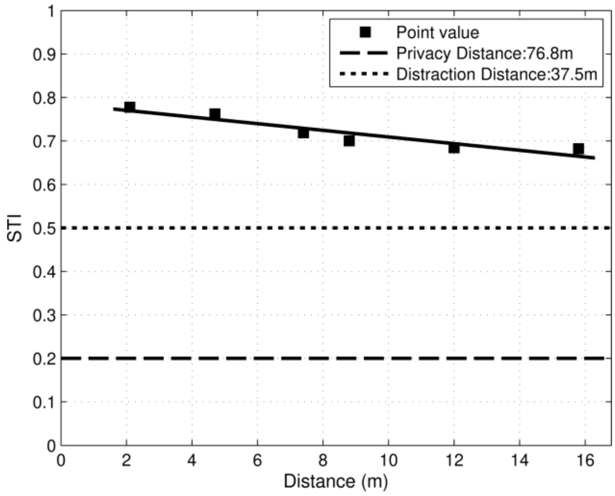

Figure 9. Value of the speech transmission index at every position

In the analyzed room, the speech transmission index never falls below 0.2 or 0.5. Because of that, privacy and distraction distances cannot be determined. Observing the linear decay of the STI, the distraction distance is around 37.5 meters while the privacy distance is at 76.8 meters. By analyzing those results, it was determined that the acoustic design of this room does not fulfil the required acoustic conditions for an open plan office. To improve the results, an extra attenuation on speech transmission shall be introduced by acoustic interventions.

The results are shown for the second measurement point on the line, which is displayed in black on the map. As in the previous measurement process, the process was performed in the same way. In Figure 10, sound pressure levels of the recorded pink noise on the positions of the rest of the points, are presented, while the loudspeaker was located at the marked point.

The second analysis performed showed better results in terms of distraction and privacy distances. Figure 10 shows spatial decay of speech sound levels, where the value of parameter $D_{2,S}$ is 5.6 dB, while the sound pressure level at a distance of 4 meters from the source is 50 dB. Analyzing those results leads to the conclusion that this room has a good acoustic quality.

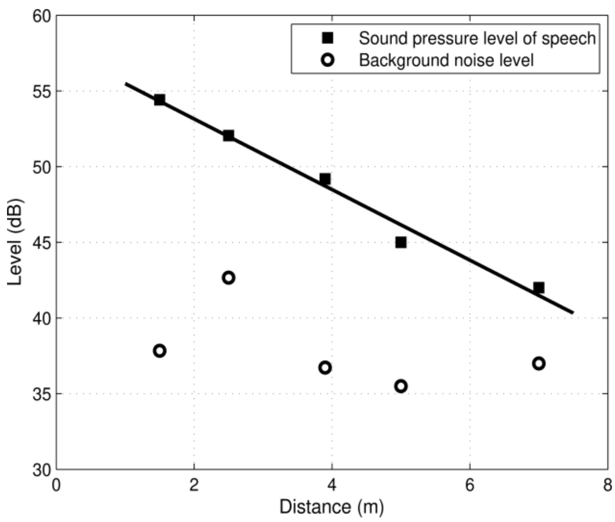

Figure 10. Figure with spatial decay rate of speech with levels of background noise

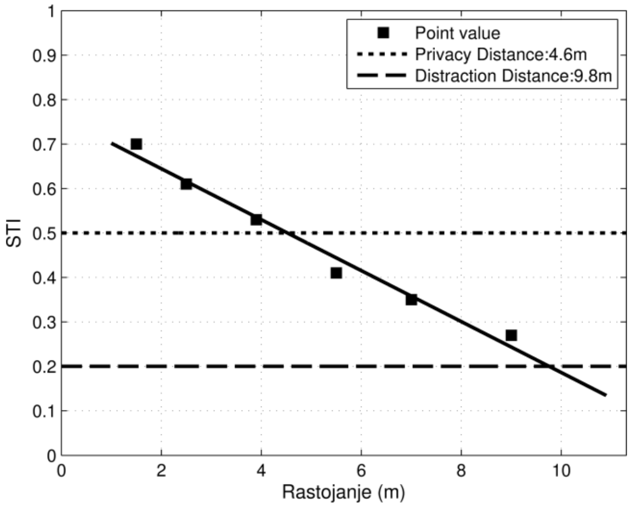

Figure 11. Value of the speech transmission index at every position

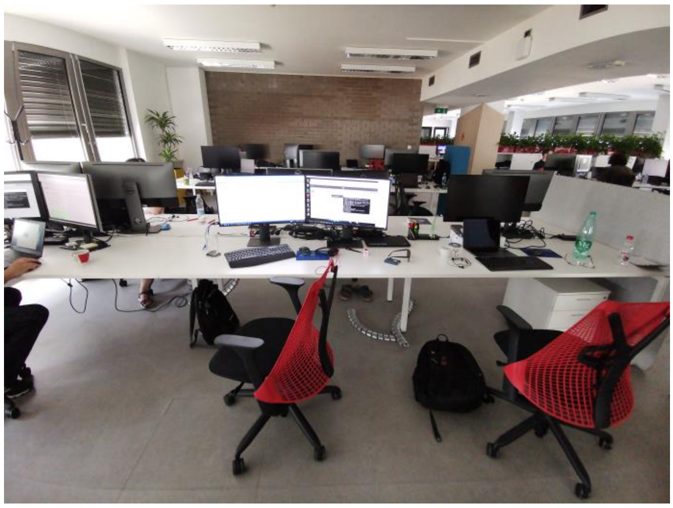

Figure 12. Open plan office without acoustic barriers

On the other side, results in Figure 11, show good declining of speech transmission index with an increase of the distance from the sound source. Already after 4.6 meters, index has a value less than 0.5 which, compared to the Table 1, describes the office as a room with good acoustic quality. Privacy distance is 9.8 meters from the sound source. The comparison of the results acquired by the analysis of two different open plan offices shows that different acoustic treatment can give noticeable deviations of values of relevant acoustic parameters.

The value of the speech transmission index of 0.5 means that around 90% of sentences reproduced by sound source is intelligible, while a value of 0.2 reduces that percentage down to 10% [21]. The spatial decay of the sound level and the decay of speech transmission index impact the level of distraction for the employees in different ways. That is the reason why both parameters have to reach the predefined optimal values from Table 1. Sound level spatial decay is important in terms of the reduction of the total background noise generated by all of the sound sources. On the other hand, high values of speech transmission index impact the clarity of speech which represents an increased distraction. To achieve good values for both parameters it is necessary to carry out different acoustic treatments. The first office that was analyzed only had absorption paneling on the ceiling and because of that, the speech transmission index values did not decline fast enough with an increase in the distance from the sound source, as seen in Figure 8.

By increasing the absorption in the room, levels of reflected sound are reduced, which is good because it impacts the spatial decay of sound in a good way. On the other hand, it also reduces the reverberation time, which means that speech clarity will increase and with that also distraction and privacy distances. The reduction of the distraction and privacy distances can be achieved by placing barriers between the workstations. The barriers have to be high enough and built from hard materials. An additional increase in acoustic quality can be achieved by adding absorption to the barriers [21]. Figure 12 shows the first office analyzed. Acoustic treatment exists only on the ceiling, and because of that, spatial decay of sound is relatively good while declining of STI is poor.

In Figure 13, a workstation in the second office is shown. Here the acoustic treatment was done on the ceiling and workstations are separated with massive barriers around 1.2 meters high, which are located on three out of four sides of the table. Additionally, the barriers have an absorption material on them. This type of acoustic treatment enabled a steep enough decline of the speech transmission index. When workstations are not far from each other, it is difficult to accomplish good attenuation of the direct sound without the barriers [21]. The situation can be improved by increasing the background noise, as HVAC and computers. In some situations, loudspeakers are used to generate the background noise and impact the values of the speech transmission index.

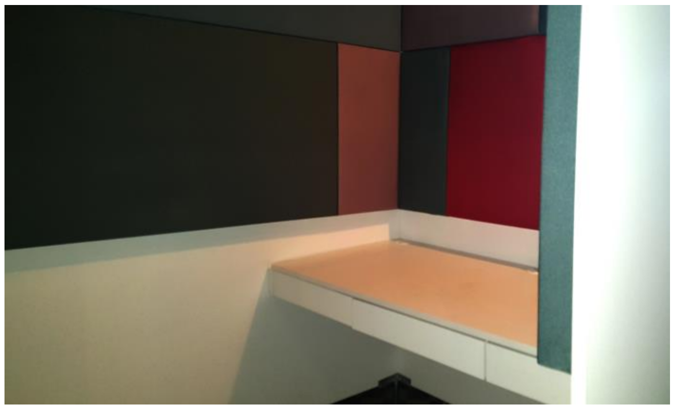

Figure 13. An example of a workstation in an open plan office with acoustic barriers

V. Conclusion

This paper presented the development of software used for measurement and analysis of the acoustic parameters of open plan offices. The measurement method and evaluation are defined by a standard, and it involves the usage of various hardware tools. This software significantly simplifies the measurement process and parameter calculations by reducing amount of necessary hardware. The automated process of parameter calculation enabled by this software reduces the amount of time required to finish the process. Measurements require only a microphone, sound source, sound card and computer. As a result, the user obtains a detailed report with relevant parameters used for the evaluation of acoustic quality of the analyzed office. Furthermore, the software saves all the necessary data, recorded signals and calculated parameters, which can later be used for repeated calculations or extended research. In this paper, the software realization was explained in detail, and an example of measurement was shown. The analysis of the results, acquired with the presented software, for two different open plan office types, was used to define guidelines for acoustic treatment for achieving better acoustic quality. For future research related to acoustic quality of work spaces, this software will be used as a tool for the analysis of a higher number of open plan offices.

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