Noise control and forecast in offices

(Contribution at Workshop SAFE - Experts meeting 4. and 5. November at the BAuA in Dortmund)

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The workshop SAFE shall create a common understanding of noise effects in office environments even in an interdisciplinary way - therefore it is the intention of this contribution to inform about existing possibilities in prediction techniques. These techniques are successfully applied in production areas with extended equipment and machinery. Figure 1 shows the "virtual" model of a bottling plant in 3D-view created to predict the noise levels at work-places on the basis of emission values declared by the machine suppliers and the acoustical properties of the room.

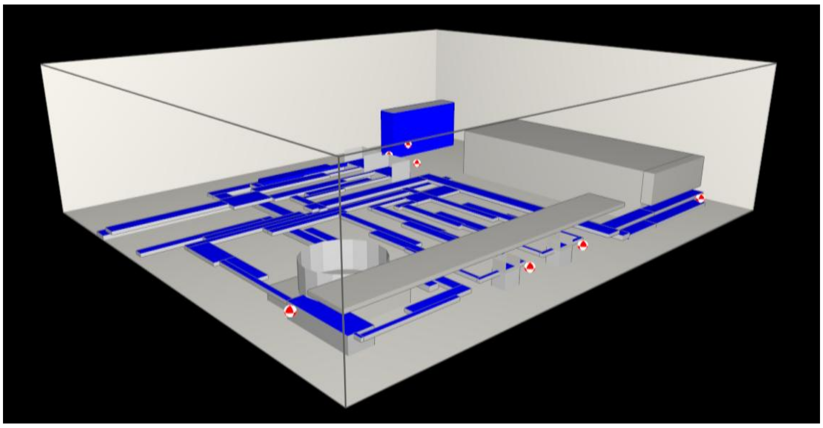


Figure 1: "Virtual" model of a bottling plant

Such a model is developed on an imported layout plan as shown in figure 2. The sound pressure levels calculated are compared with the target values agreed between supplier and buyer and in case those targets are not met noise abatement measures can be implemented in the model and the noise prediction procedure is repeated.

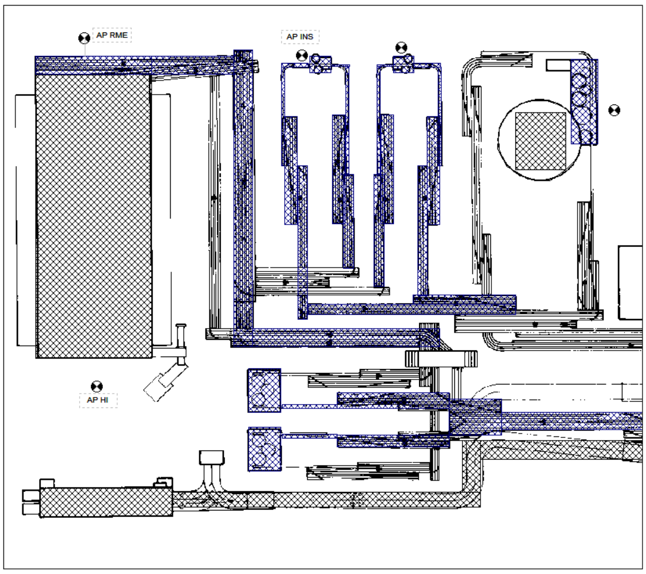


Figure 2: 2D layout plan with work places

Figure 3 shows the model with noise reduction measures inserted as e. g. the suspension of sound absorbing baffles, the noise protection of work places by transparent screens and even by primary measures like the reduction of transportation speed of the bottles. Table 1 shows the levels L_r for the plant according to the first configuration (figure 1) and the levels $L_{r,wM}$ after the mentioned improvement (figure 3).

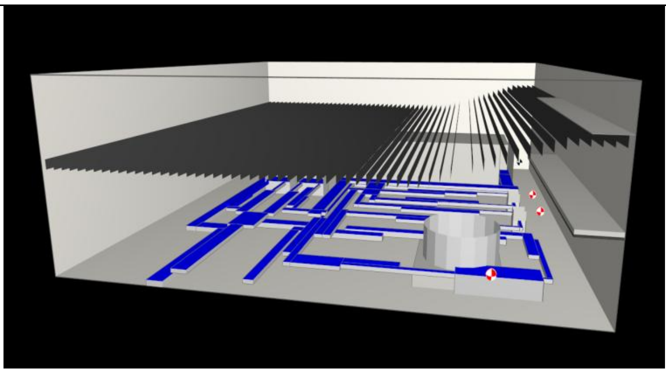


Figure 3: Bottling plant with inserted absorbing baffle system

Table 1: Levels predicted without and with measures

Workplace	L_r	$L_{r,wM}$
Depack	84.4	**84**
Washing	88.2	**84.3**
Inspection 1	87.5	**85.2**
Inspection 2	87.9	**85.4**
Filler	86.3	**84.3**
Labeller 1	86.2	**84.8**
Labeller 2	86.3	**84.9**
Packer	85.4	**85**
Palettiser	86.1	**85.3**

These same methods shall be applied to predict the - much lower - sound pressure levels in office environments to check the probable acoustic quality at work places. To understand the differences and the problems that have to be solved, the basic principles shall be presented in the following.

Mirror image method (specular reflection)

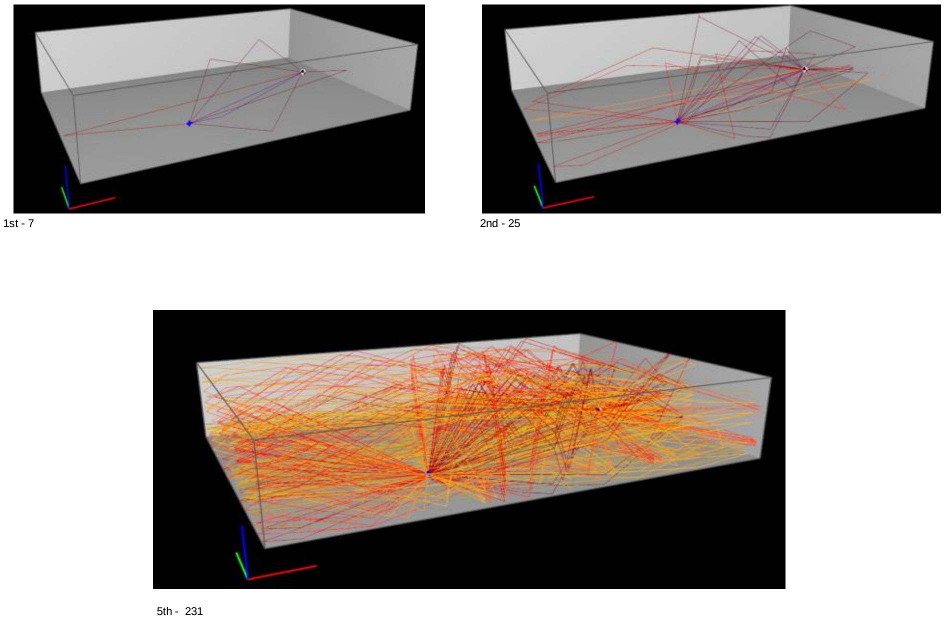


Figure 4: The calculated rays between a point source and a receiver with mirror image method

Using mirror image methods the ray paths between source and receiver are calculated deterministically - figure 4 shows a calculation with 7 rays up to 1st order, with 25 rays up to 2nd order and with 231 rays up to 5th order. Different orders are coded by different colors. The number of necessary calculations explodes if we take higher reflection orders and longer propagation times into account which is necessary if we check the temporal behavior expressed as impulse response of a room.

For such applications the SERT-method (**S**tochastic **E**nergy **R**ay **T**racing) is the better choice. Figure 5 shows the equally distributed emission of sound rays or particles in all directions by a point source in the first image. The following one shows particles after being reflected by the room's surfaces and finally after many reflections representing something similar to a diffuse sound field. Depending on accuracy-aspects about 1 million particles are radiated from each source.

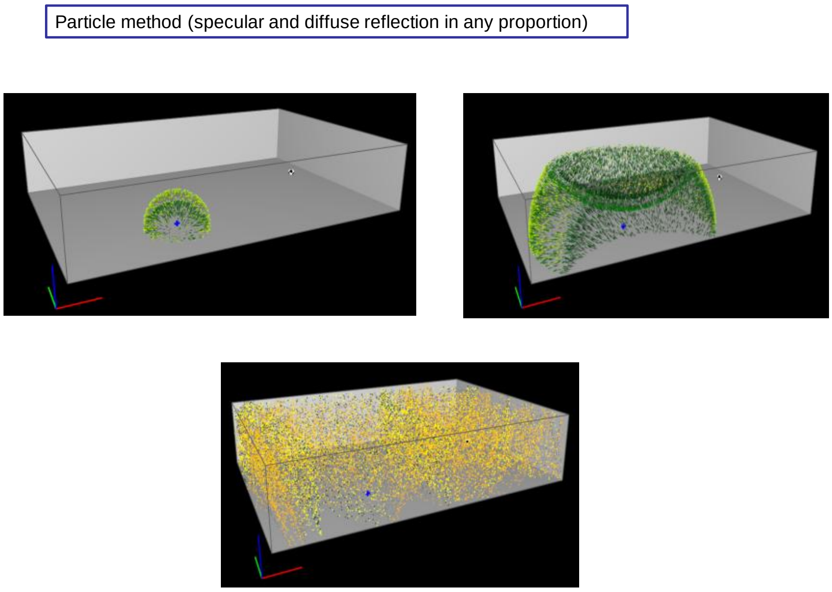


Figure 5: Sound "particles" being just radiated, then after being reflected and finally forming a diffuse sound field

The room of any shape is subdivided into smaller spherical or box shaped detector volumes called voxels and each particle crossing a voxel contributes to the final summed up energy density related to its center-point.

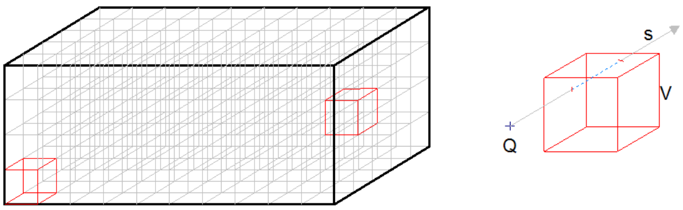


Figure 6: Subdivision of a room into detector volumes

From various studies we know that the human voice is the most relevant sound source within office environments and the intelligibility of speech is the most relevant criterion to decide about annoyance, decrease of concentration or other psycho-acoustic effects.

Therefore we replace the point source and the receiver in figure 5 by a speaking person and another - even involuntarily - listening person as it is shown in figure 7 with the girl making a telephone call and her college at a nearby office workplace. This is in a certain way the simplest possible situation and therefore an "atom" of any complex layout in open plan offices.

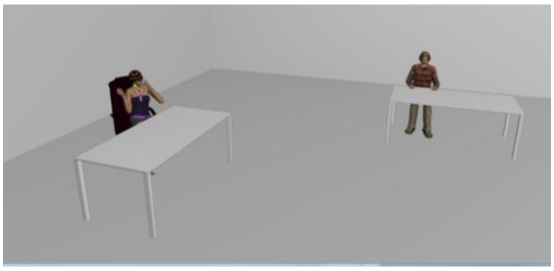


Figure 7: Sound source "telephone call" and receiver at a nearby workplace

In acoustic terms the woman at the telephone is a sound source with a certain sound power level defining its emission. It makes sense to relate this sound power level to the speech effort as it is expressed in the upper diagram of figure 8. If we assume a "normal, relaxed" verbal behavior we see that an A-weighted sound power level of 65 dB(A) can be assumed which can then be resolved into the seven levels of octave bands shown in the diagram below. For sound prediction with speech sources seven calculations for the octave frequency bands from 125 Hz up to 8000 Hz have to be performed generally for each single source.

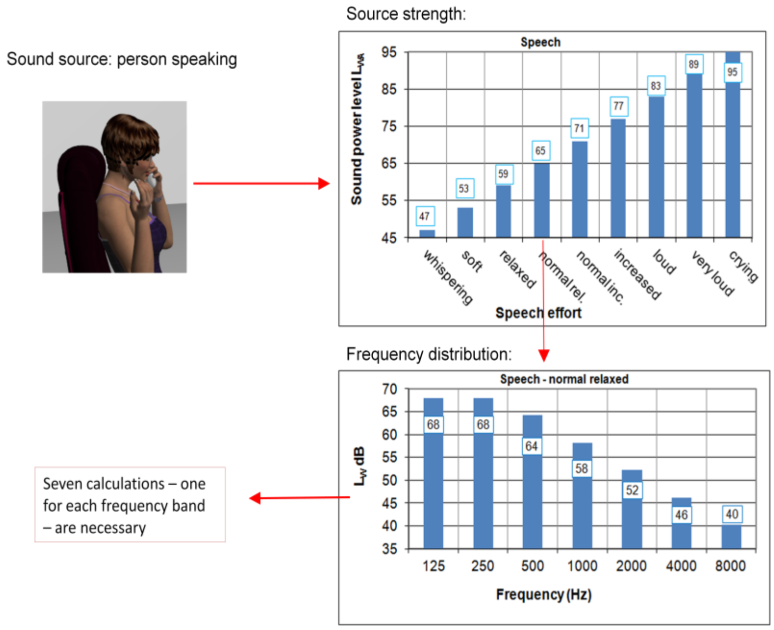


Figure 8: The speech signal expressed as emission values with sound power level and frequency spectrum

Simulating the sound propagation to predict the acoustic impact at the receiver's position the particle model is applied as shown before. Figure 9 highlights the different phases of sound propagation.

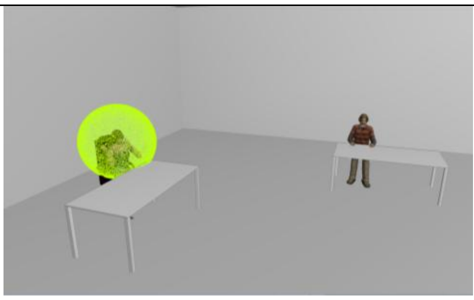


Figure 9a: Direct sound radiation

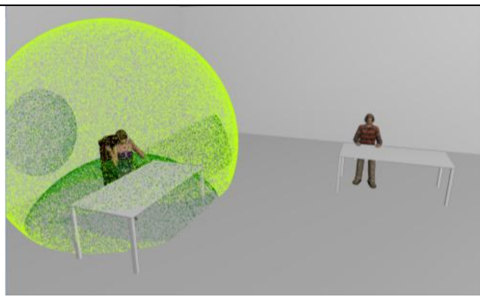


Figure 9b: After first reflection at a surface

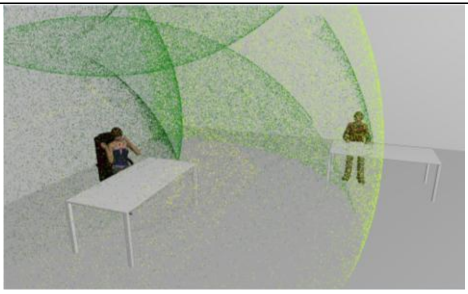


Figure 9c: Reaching the receiver for first time

Figure 9d: After many reflections and diffuse mixing

In simulations applying the SERT-method the propagation path of each particle or ray is determined up to a certain length - taking the speed of sound into account we then know the time shift between radiation of a particle and each impact on a detector volume. Even if the calculations are performed sequentially according to the computer-technique applied we can assume that they have been radiated at the same time and sort them in classes of the time needed until impacting the regarded detector volume. The result is - separated for each detector volume - the energy related impulse response as shown in figure 10.

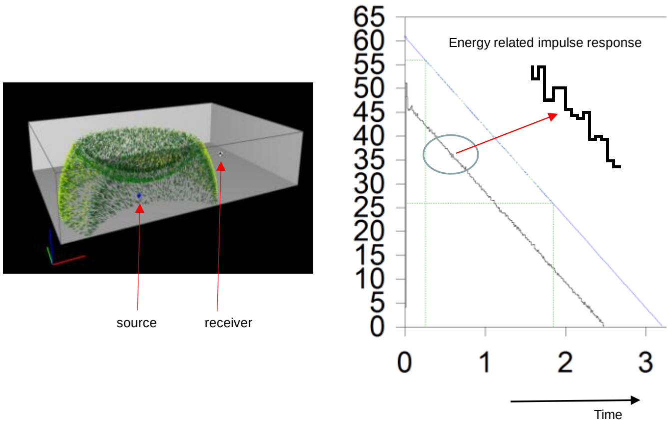


Figure 10: The energy related impulse response calculated for each detector volume from impacting particles and propagation time

The intelligibility of speech depends on the modulation of the sound signal due to the forming of syllables, words and sentences. The time shifted impact of the sound emitted at the same time represented by the energetic impulse response "smears" the speech signal and reduces the modulation depth - thus reducing its intelligibility. Therefore the speech signal emitted by the speaker on the left side in figure 11 is not only reduced in its intensity or level when it arrives at the listener on the right side but the modulation depth is decreased as well.

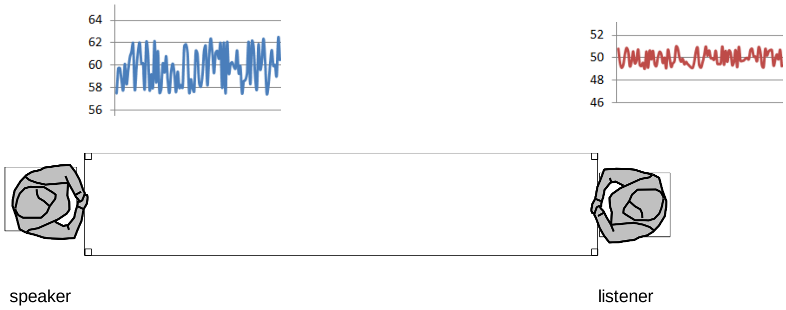


Figure 11: Decrease of modulation depth of speech due to reverberation at listeners position (simplified presentation)

This reduction in modulation depth can be calculated and qualified with the strategies of IEC 60268-16.

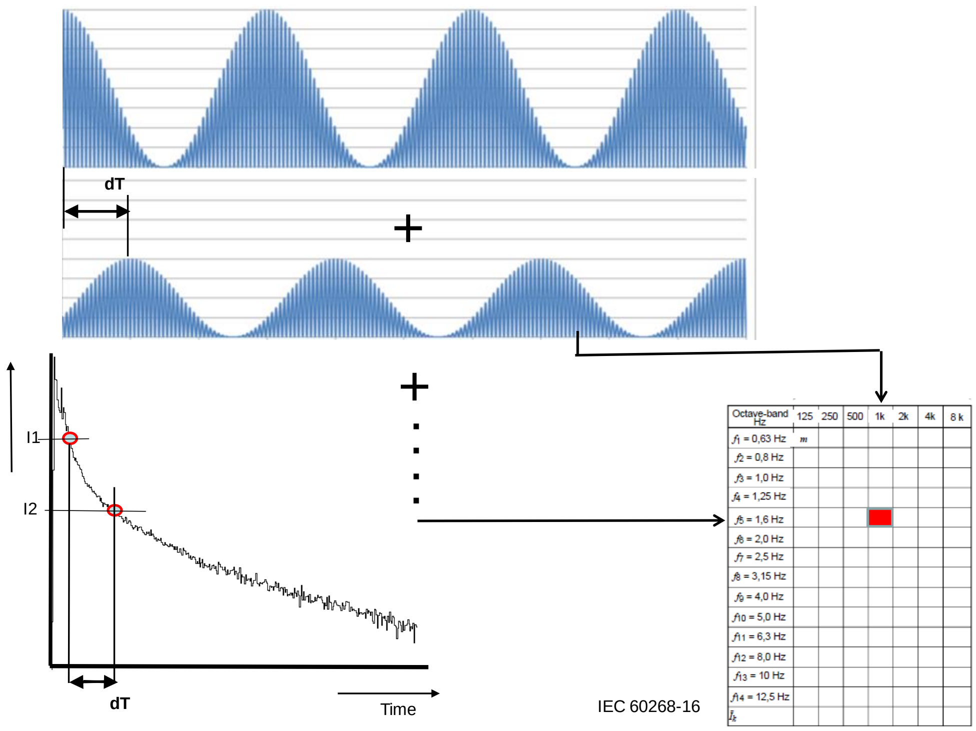


Figure 12: Superposition of time shifted signals with 1000 Hz octave band modulated with 1,6 Hz according to the energetic impulse response curve yields one of 98 Modulation Transfer Indices MTF

Figure 12 shows the principle. The reduction of modulation depth is calculated for each of the 7 octave bands from 125 Hz up to 8000 Hz taking into account a modulation with one of 14 frequencies from 0.63 Hz up to 12,5 Hz which means that 98 values in the MTF-matrix (Modulation Transfer Function) shown in the lower right are determined. If the energy related impulse response between a source and a receiver position has been determined using e.g. 2000 samples in 1 ms time steps for the 1k frequency band as shown in the lower left diagram, the MTF value for a modulation frequency 1,6 Hz is determined by superimposing this 1 k signal 2000 times modulated with 1,6 Hz but shifted in time and reduced in intensity according to the calculated decay curve. The resulting MTF value for this pair of a signal and a modulation frequency is the quotient of the remaining modulation depth at the receiver (therefore of the resulting summation of 2000 signals) and the original single signal radiated by the source. From these 98 MTF values between 0 (modulation completely destroyed) and 1 (modulation not reduced) the speech transmission index STI - a single number value between 0 and 1 - is determined as an objective rating of speech intelligibility according to IEC 60268-16.

The modulation depth of a speech signal is not only influenced by the acoustic response of the environment but also by existing background noise from other sources. If the sound pressure level of this background is known for the relevant 7 frequency bands the resulting further reduction of the STI can be determined with the procedure of IEC 60268-16.

The calculation of the single number STI from the 98 MTF values takes the signal to noise ratio in each of the 7 frequency bands (therefore also the speech effort in the upper diagram in figure 8), the masking of adjacent frequency bands and even the hearing threshold in each frequency band into account.

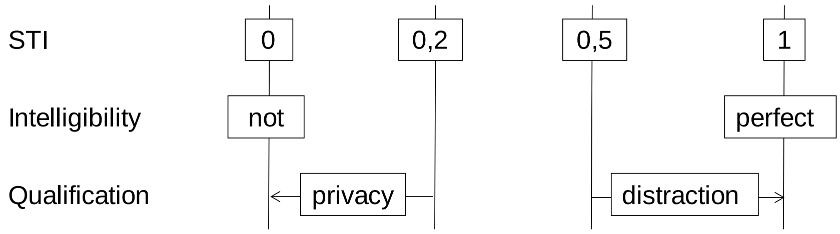


Figure 13: The intelligibility and its qualification in dependence from the STI value

In all applications where the intelligibility of speech sounds may be important the qualification can be oriented at the STI value determined between a speaker's and a listener's position. At distances where the STI is around or below 0.2, intelligibility is not given and privacy can be assumed. This should be checked between areas where confidential talks shall be possible and all other workplaces. If the STI is greater or equal to roughly 0.5 intelligibility is probable which may cause distraction and which has to be checked between different workplaces. If relaxed communication shall be ensured the goal is to reach STI values of 0.5 or higher.

It is important that the speech intelligibility and the STI depend on the impulse response of the room **and** on the signal to noise ratio of the speech signal at the listener's position.

In the special case of a pure diffuse sound field the impulse response is strongly related to the reverberation time of the room. Figure 14 shows this dependency - the STI at the vertical axis is shown as a function of the signal to noise ratio in dB and the reverberation time on the two horizontal axes.

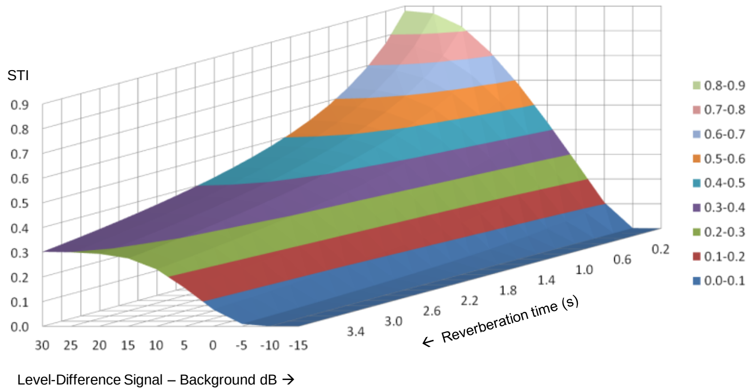
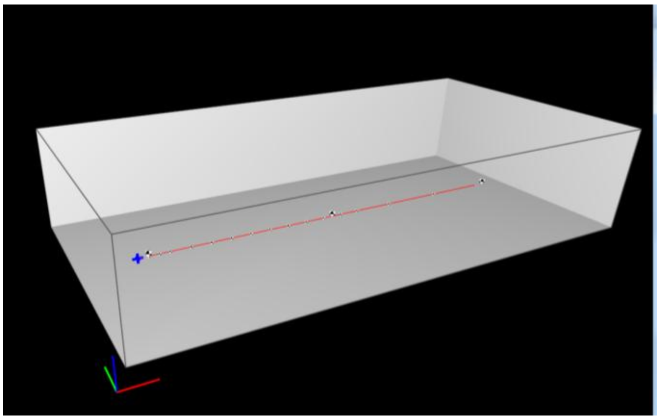


Figure 14: The Speech Transmission Index STI as a function of Signal to Noise ratio and of reverberation time

But this relation is only true if the listener is placed within a diffuse sound field which on the other hand can be described by a reverberation time - with adjacent workplaces in offices this is generally not the case.

A simple example to check these dependencies of the STI is shown with figure 15. The source is a speaking person on the left and the STI is calculated with the real impulse response along a straight path extended to the right side. The reverberation time of this room is roughly four seconds and from figure 14 an STI of 0.3 or even lower can be derived even if there is no background noise.



Room 50 m x 25 m x 10 m
Absorption all surfaces 0.3
A = 1200 m²
T60 -> 4s – 5 s
r_H = 5 m

STI from T60 → 0.3

Figure 15: Model of a large room to determine the STI along a straight path

Figure 16 shows the STI along this path calculated with the impulse response at each position - the dots correspond to the calculated values. The STI values predict a very good intelligibility at distances below 5 m and only at distances greater than 10 meters the value predicted from reverberation time is true. It shall be mentioned that the regression line in this diagram and the derived values of r_D and r_P according to VDI 2569 have no meaning in this case.

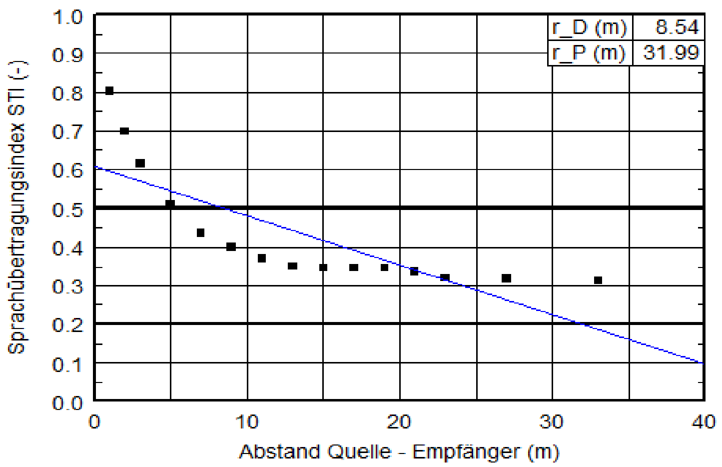


Figure 16: STI in dependence of the distance speaker - listener in a highly reverberant room

Another example where the STI calculated with the detailed impulse response is necessary to decide about intelligibility is shown exemplarily in figure 17. The room on the left side holding a speaker and a listener is treated with absorbing material but a large hard walled room - may be a storage room - is acoustically connected to the former by an opening. The impulse response on the lower right shows a typical concave shape which means a mean reverberation time makes no sense for this configuration. The STI calculated based on the impulse response is 0.87 and proves a very good intelligibility inside this left part even with a larger distance between speaker and listener. This shows that the first part of the impulse response determines the intelligibility - the later reverberation is of minor importance for the qualification of the room.

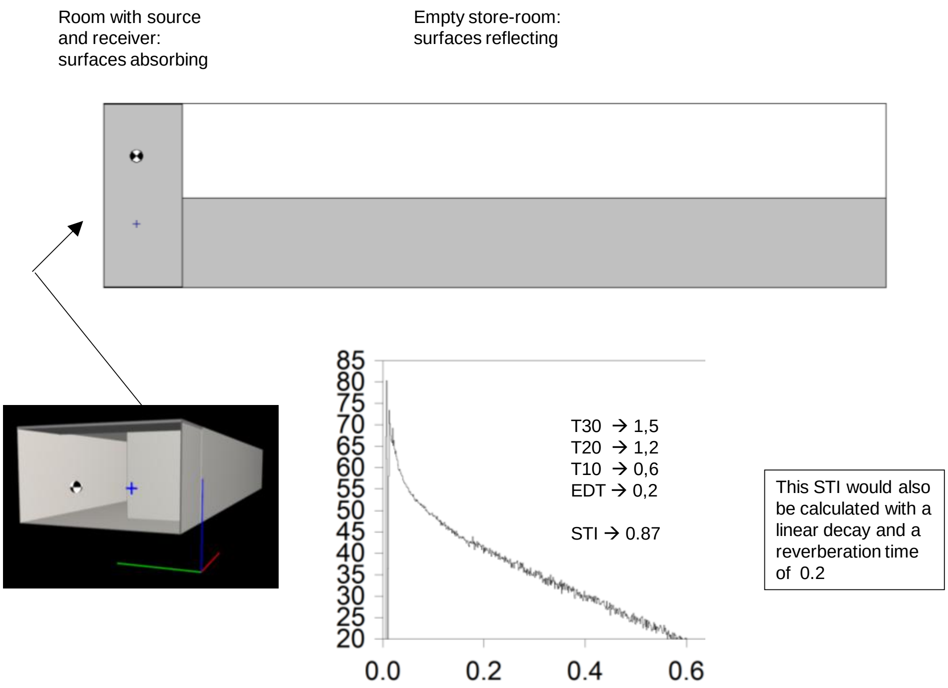


Figure 17: Acoustically damped room with coupled reverberant part

An estimation of the intelligibility of speech on the basis of the reverberation time is completely impossible in situations with coupled spaces like it is shown in figure 18. It shall be checked if a person speaking in one of these rooms can be understood in the adjacent rooms if the doors are open. The absorbing treatment and the geometry of the rooms shall be taken into account.

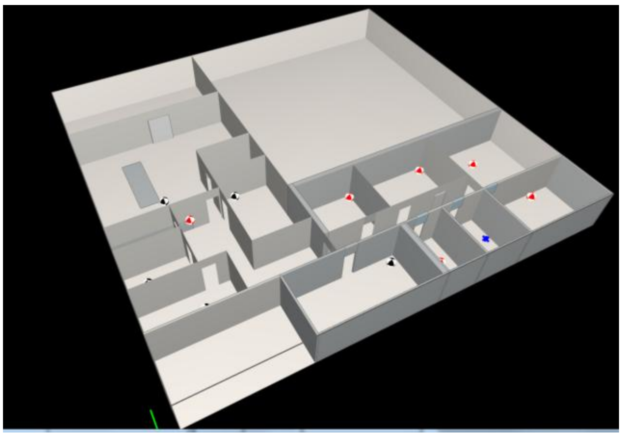


Figure 18: Many little office rooms acoustically linked by open doors

Applying the SERT calculation and deriving the STI from the impulse response in each room separately is the only way to get the correct answer according to figure 19. The speaker can be understood in all adjacent rooms with an STI above 0.5.

It shall only be mentioned that the situation with an unwanted good intelligibility can be improved with a background noise spectrum. Natural communication background or artificial masking sounds can easily be taken into account in such investigations.

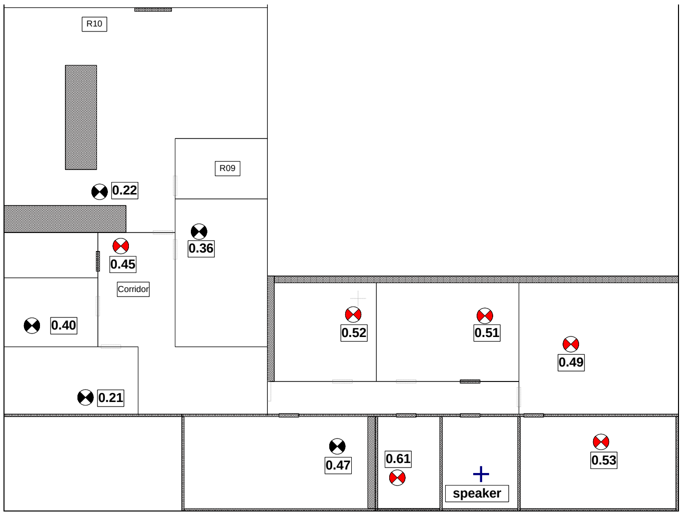


Figure 19: Calculated STI values in different rooms with the speaker in another room - doors are open

The planning procedure in a simple case is demonstrated with the model shown in figure 20.

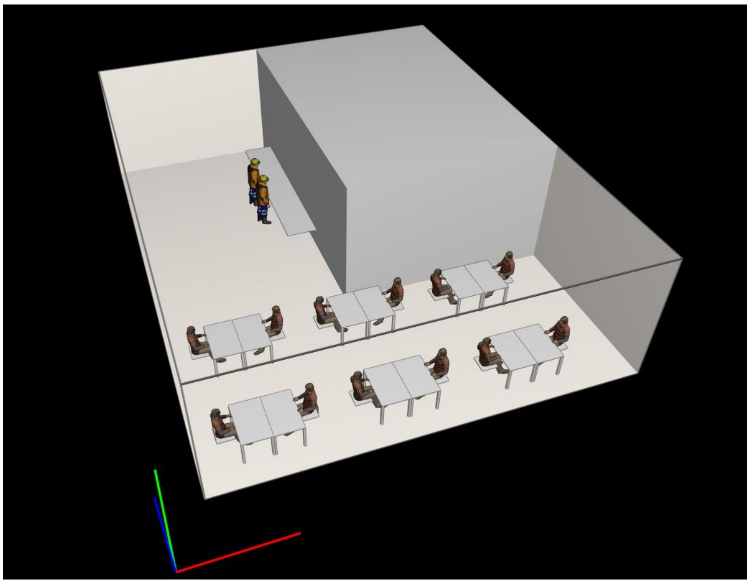


Figure 20: Model of a dining or office room with a food court area

The room is L-shaped, where one flank is a working or dining room with six tables for two persons sitting opposite to each other and the other flank is a basically empty room with two persons at the buffet. An acceptable acoustic planning should ensure that each pair of two people can communicate with normal relaxed speaking effort and that this communication cannot be understood at all other tables.

The prediction procedure is shown for one table - it can be repeated in the same way for all tables sequentially. Provided that the model is complete and includes all absorptions, screens and acoustically relevant furniture and fittings the assessment is performed in two steps.

The first step is to predict the background noise level due to the communication at all other tables at the position under test. This is done by taking one source at each table with an emission due to normal relaxed speech into account. This is generally the recommended procedure and independent from the number of persons at a table in restaurants or offices - one person is speaking while the others are listening. The SERT-calculation is applied and the level spectrum at the listener's position at the table under test is calculated.

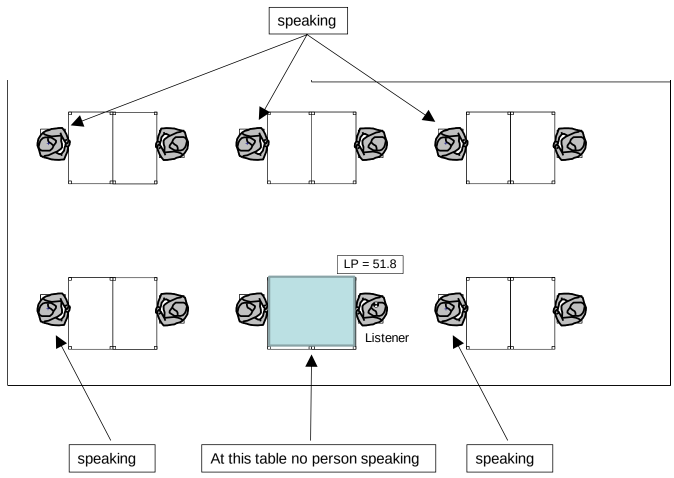


Figure 21: Calculation of background noise - at all other tables one person speaking normal relaxed

The resulting level at the table under test is 51,8 dB(A).

In the second step the frequency spectrum of the background noise determined in step one is now taken as background noise with only the person on the left side of the table under test speaking.

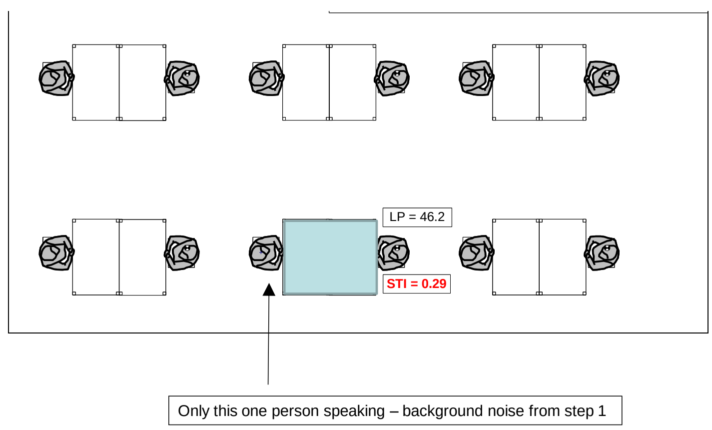


Figure 22: Calculation of the STI at the listener position with the background noise spectrum from step 1

As it is shown in figure 22 the signal level from the speaking person is 46 dB(A) and the resulting STI of 0,3 indicates a bad intelligibility. This is exactly the situation where the speaking person will increase the speaking effort to help the listener - this will increase the signal to noise ratio and the resulting STI - but because everyone will do so we end up with the same signal to noise ratio as before. This well known Lombard-effect can only be avoided by improving the acoustic fitting and layout in the room.

Such an improvement is indicated in figure 23 where absorbing barriers are inserted between the tables and the wall is covered with an absorbing panel up to a height of 1.6 meters to create an acoustically damped individual space around each table.

The previous steps are now repeated. As it is shown in figures 24 and 25 the background level from all other tables is now 38 dB(A) and using its frequency spectrum as background noise an STI of 0.7 indicates a good intelligibility.

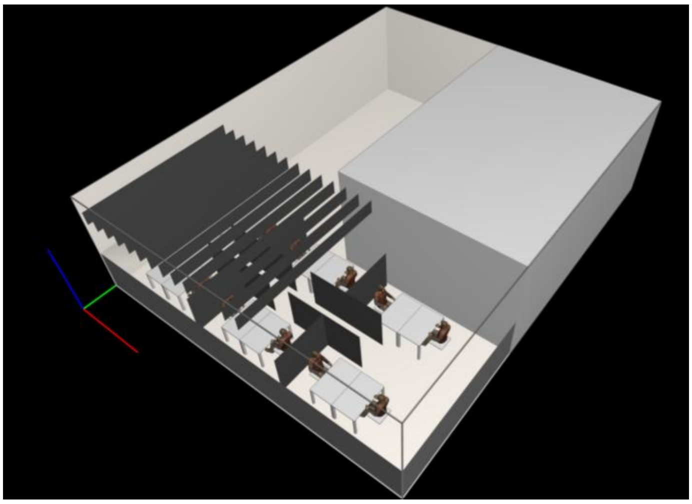


Figure 23: Absorbing screens and wall cover with baffle ceiling

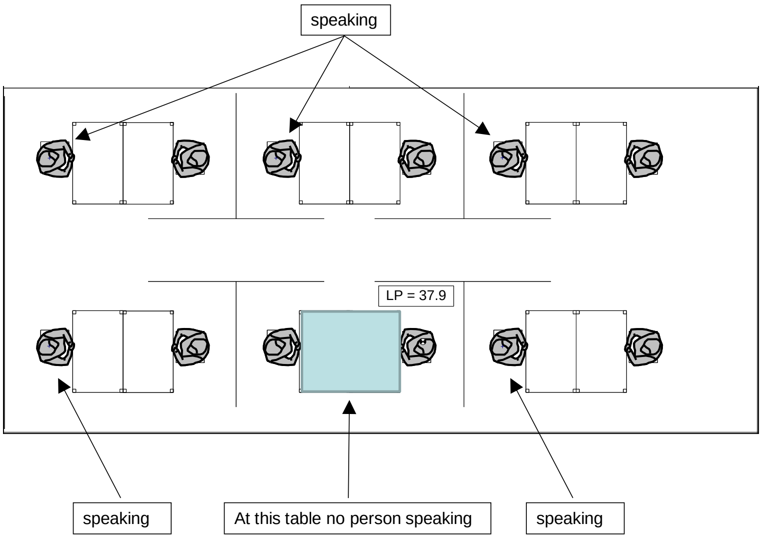


Figure 24: Determination of the background noise with 38 dB(A) in step 1

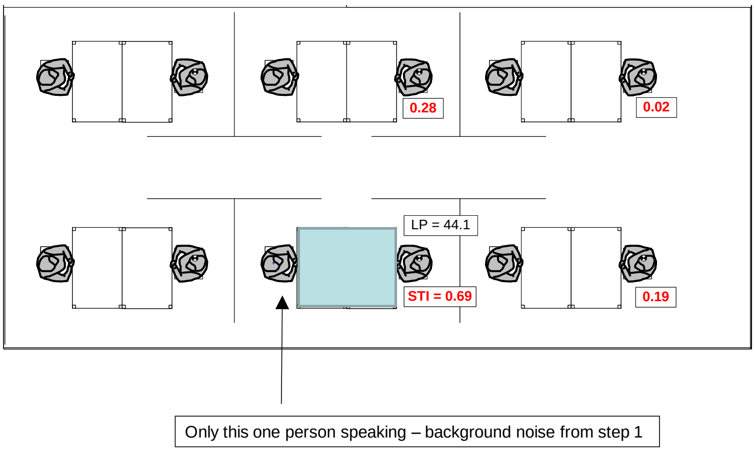


Figure 25: Calculation of the STI yields 0.7 due to the reduced background noise and better damped room response

This example was only presented to show the principles. The intended message is that our simulation techniques offer excellent means to assess the acoustic relations between subareas with different usage and requirements. Especially the concept of the STI according to IEC 60268-16 includes the influence of the existing background noise and of the energetic impulse response independent of the existence of a diffuse sound field - therefore all acoustic requirements based on the intelligibility of speech can be investigated. The next steps are to investigate the uncertainties related to this prediction process in typical office environments and to improve it where appropriate. Guidance rules should be developed for the planning phase of such office environments to support the creation of layouts that are in accordance with our requirements. It is also necessary to adapt standardization to this new techniques - the acoustic properties of all products, furniture and fixtures should be described in a way that the corresponding datasets can be applied in acoustic simulations.

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