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 is a view to a 3D-model of a bottling plant created to predict the noise levels at work-places on the basis of emission values declared by the machine suppliers and the acoustic properties of the room.

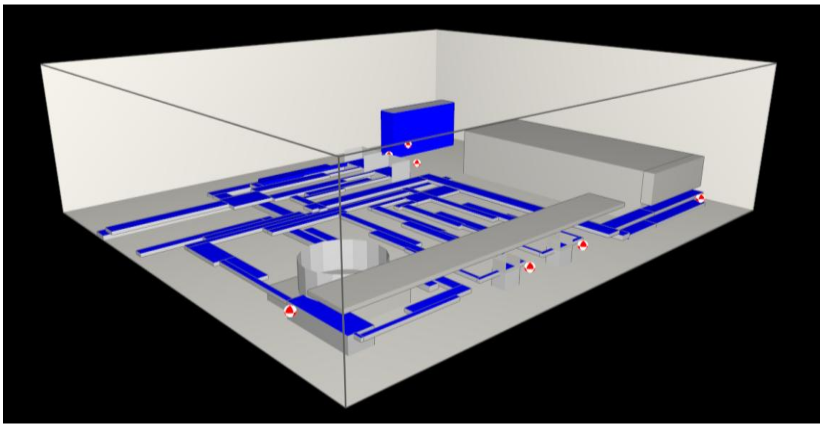


Figure 1: "Virtual" model of a bottling plant

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

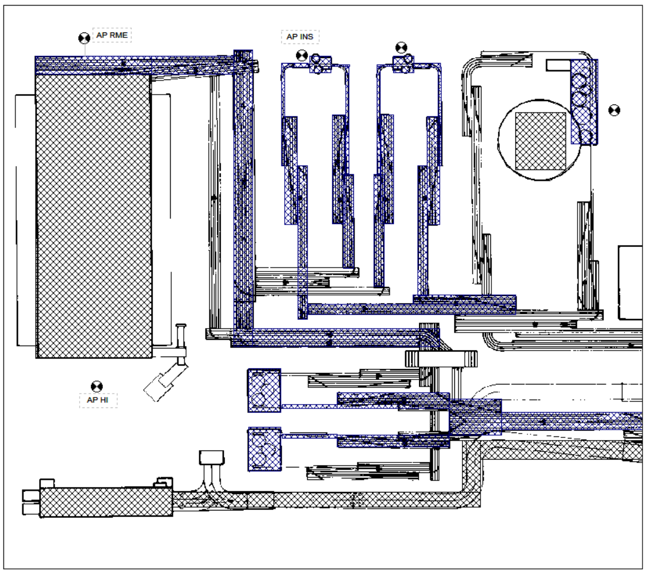


Figure 2: 2D layout plan with work places

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 Lr with the plant according to the first step (figure 1) and with levels after improvement (figure 3).

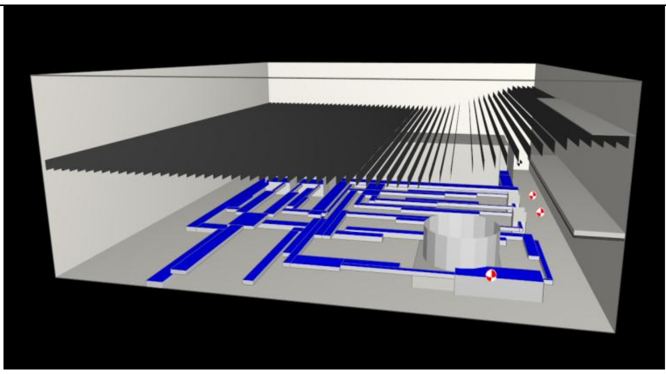


Figure 3: Bottling plant with inserted absorbing baffle system

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

Mirror image method (specular reflection)

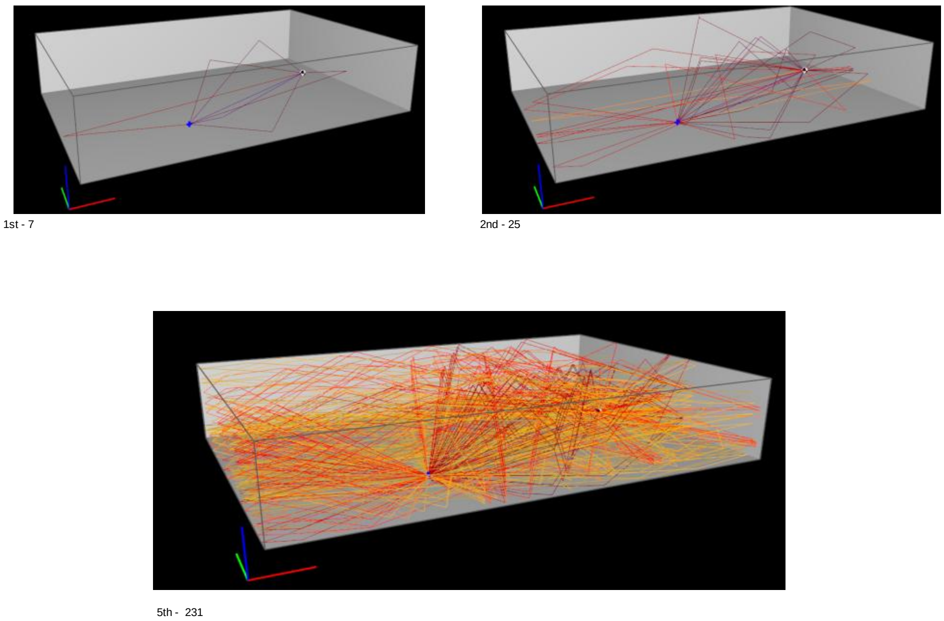


Figure 4: The calculated rays between a point source and a receiver with mirror image method

With mirror image methods the ray paths between source and receiver are calculated deterministically - figure 4 show a calculation with 7 rays up to 1st order, with 25 rays up to 2nd order and with 231 rays up to 5th order. This number of necessary calculations explodes if we take into account higher reflection orders and longer propagation times as it is needed 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 in three steps the sound rays or particles emitted with equal distribution in all directions from the point source, then after being reflected by the room surfaces and finally after many reflections representing something like a diffuse sound field. Depending on accuracy-aspects about 1 million particles are radiated from each source.

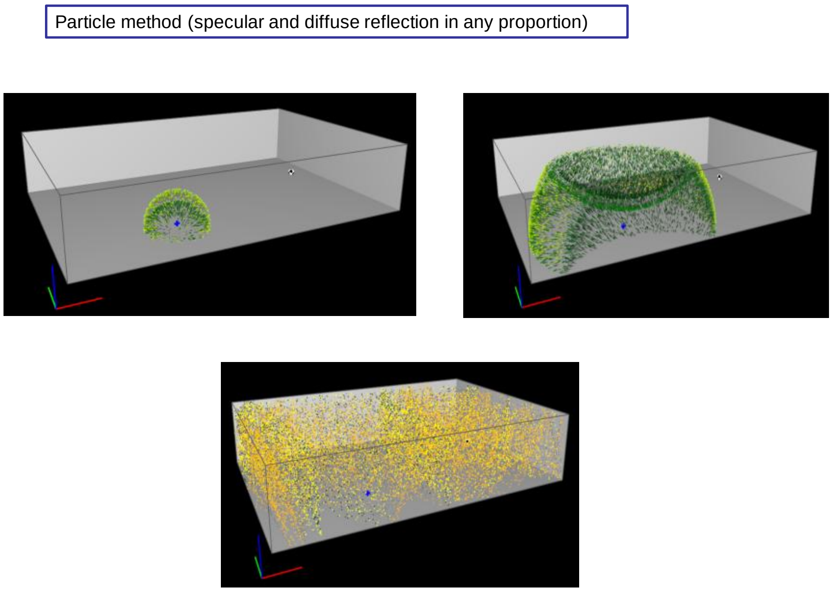


Figure 5: Sound "particles" being just radiated, then after being reflected and finally forming a diffuse sound field

The room of any shape is subdivided in smaller spherical or box shaped detector volumes, and each particle crossing such a volume contributes to the finally summed up energy density related to its center-point.

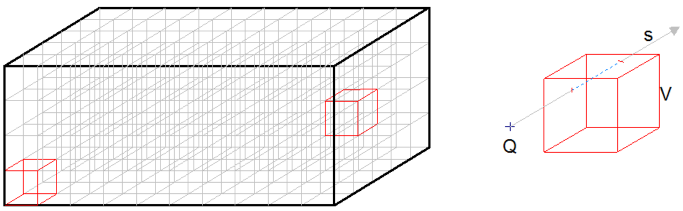


Figure 6: Subdivision of a room in detector volumes

From various studies we know that the human voice is the most relevant sound source in 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 another office workplace. This is in a certain way the simplest possible situation and therefore an "atom" of any complex layout in open plan offices.

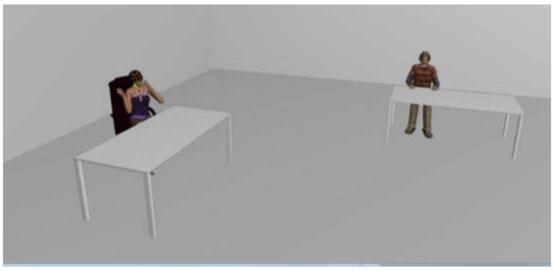


Figure 7: Sound source "telephone call" and receiver at another workplace

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 figure 8. If we assume a "normal relaxed" talking we see that an A-weighted sound power level of 65 dB(A) can be assumed and this again can be resolved into the seven levels in 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.

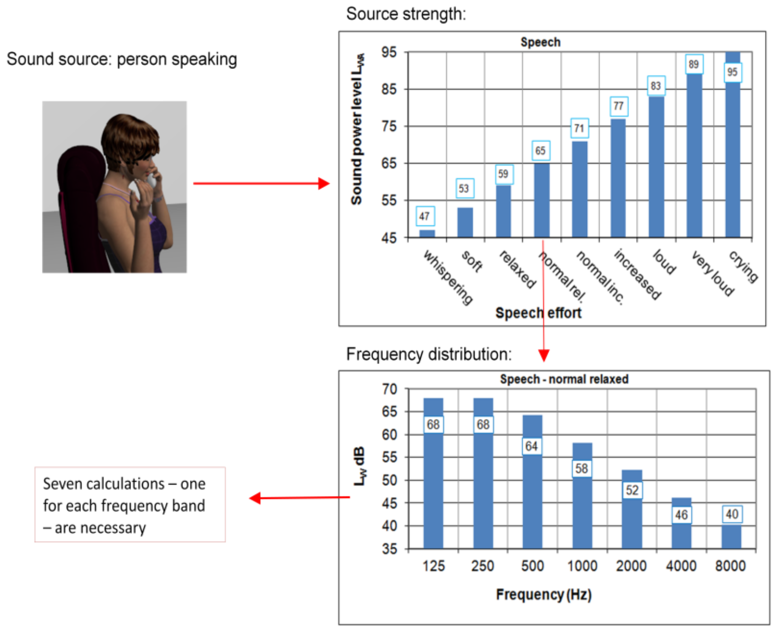
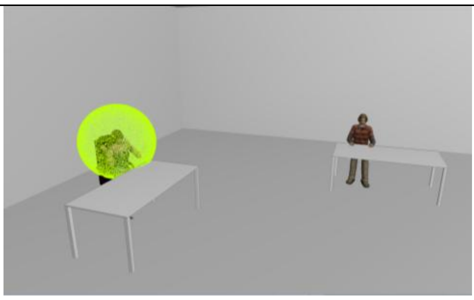
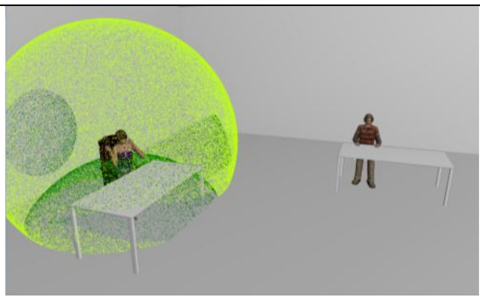


Figure 8: The speech signal expressed as emission values with sound power level and frequency spectrum

Simulating the sound propagation to predict the acoustic impact at the receiver position, the particle model is applied as it was shown before. Figure 9 shows the different phases of sound propagation.

 Figure 9a: Direct sound radiated	 Figure 9b: After first reflection at surfaces

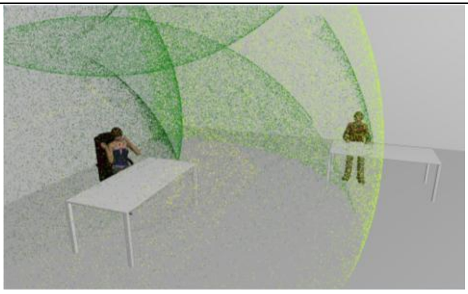


Figure 9c: Reaching the receiver first time

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 into account the speed of sound we then know the time shift between radiation of a particle and each impact in 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 till impact in the regarded detector volume. The result is - for each detector volume separate - the energy related impulse response as shown in figure 10.

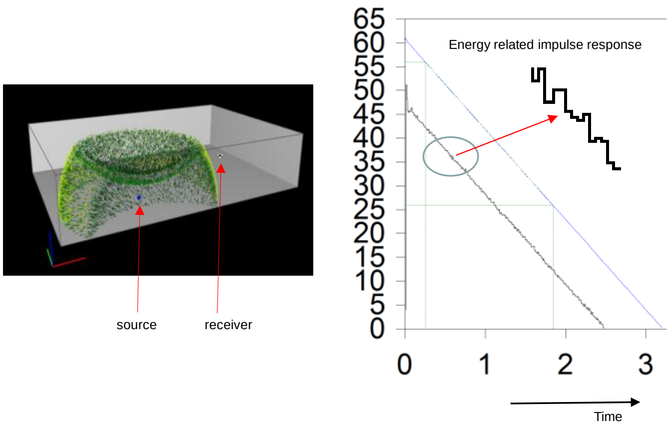


Figure 10: The energy related impulse response calculated for each detector volume from impacting particles and propagation time

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 left side in figure 11 is not only reduced in its intensity or level when it arrives at the listener right side, but also the modulation depth is decreased.

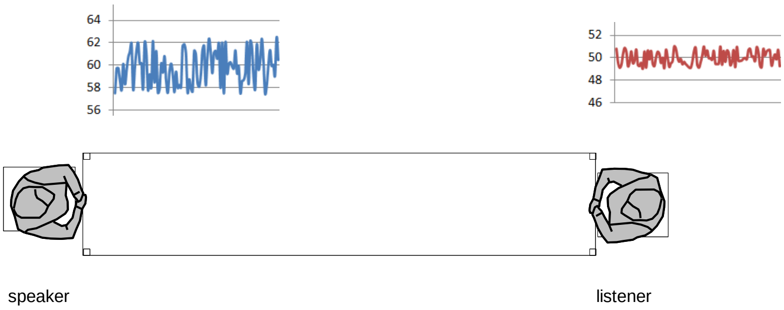


Figure 11: Decrease of modulation depth of speech due to reverberation at listeners position (simplified presentation)

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

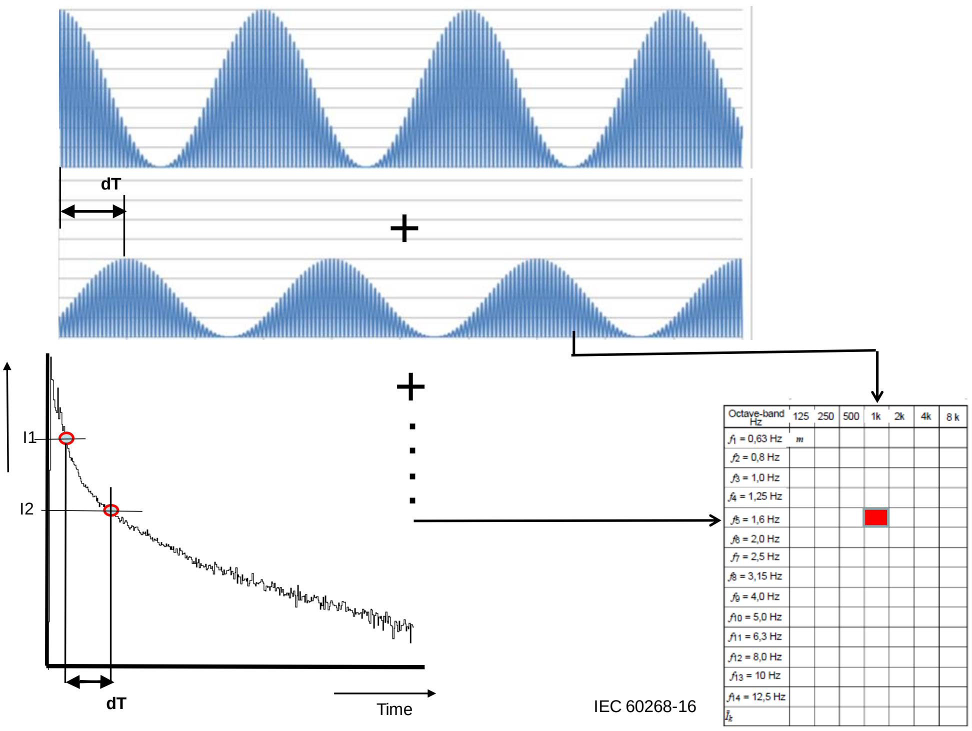


Figure 12: Superposition of time shifted signals with 1000 Hz octave band modulated with 1,6 Hz according to the energetic impulse response curve yields one of 98 Modulation Transfer Indices MTF

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 - this means that 98 values in the MTF-matrix (Modulation Transfer Function) shown lower right are determined. If the energy related impulse response between a source and a receiver position has been determined with 2000 samples in time steps of 1 ms 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 superposing 2000 times this 1 k signal 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 signal and 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 - as an objective rating of speech intelligibility according to IEC 60268-16 is determined.

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 into account the signal to noise ratio in each of the 7 frequency bands (therefore also the speech effort in the upper diagram figure 8), the masking of adjacent frequency bands and even the hearing threshold in each frequency band.

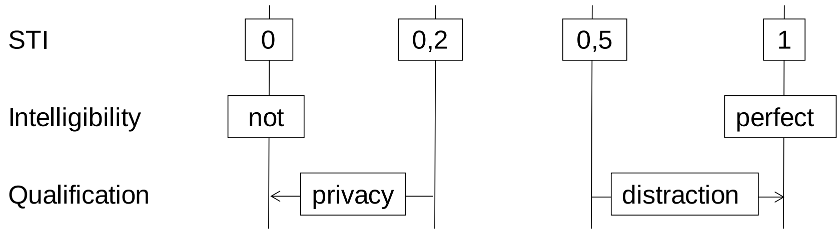


Figure 13: The intelligibility and its qualification in dependence from the STI value

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 and a listener 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 or other positions of persons. If the STI is about 0.5 or larger then intelligibility is probable. This may cause distraction and has to be checked between different workplaces. If relaxed communication shall be ensured then values of the STI at or above 0.5 are the target.

But it should always be taken into account that 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 listeners 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 of the reverberation time at the two horizontal axes.

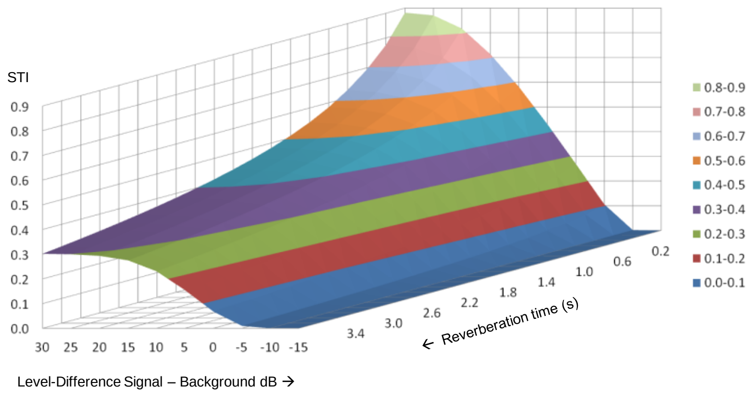
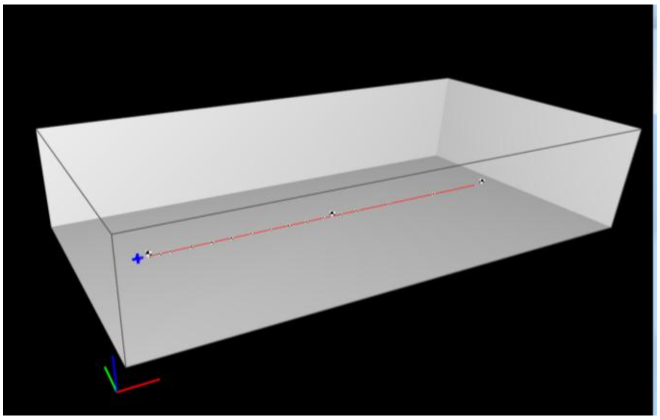


Figure 14: The Speech Transmission Index STI as a function of Signal to Noise ratio and of reverberation time

But this relation is only true if the listener is in the diffuse sound field and this 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 left side and the STI is calculated with the real impulse response along a straight path extended to the left side. The reverberation time of this room is about 4 seconds and from figure 14 can be derived an STI of 0.3 or even lower if there is no or negligible 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 show the calculated values. The STI values predict a very good intelligibility at distances below 5 m, and only at distances larger than 10 m 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 sense in this case.

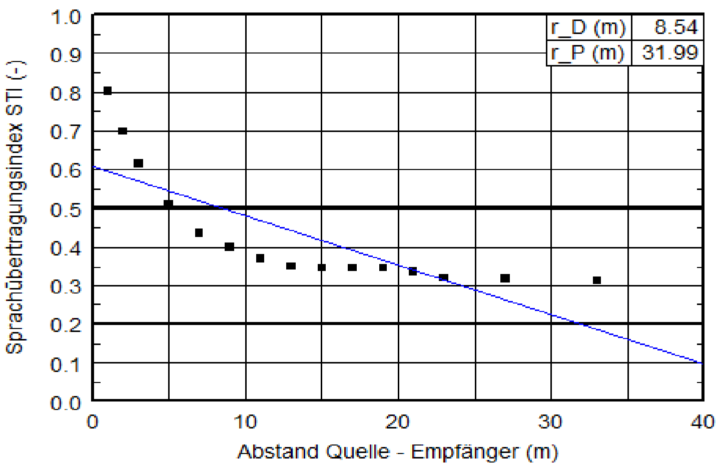


Figure 16: STI in dependence of the distance speaker - listener in a highly reverberant room

Another example where the STI calculated with the detailed impulse response is necessary to decide about intelligibility is shown with example figure 17. The room at the left side with speaker and listener is treated with absorbing material, but a large hard walled room - may be a storage room - is acoustically coupled with it by an opening. The impulse response lower right shows a typical concave shape - 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 speaker - 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.

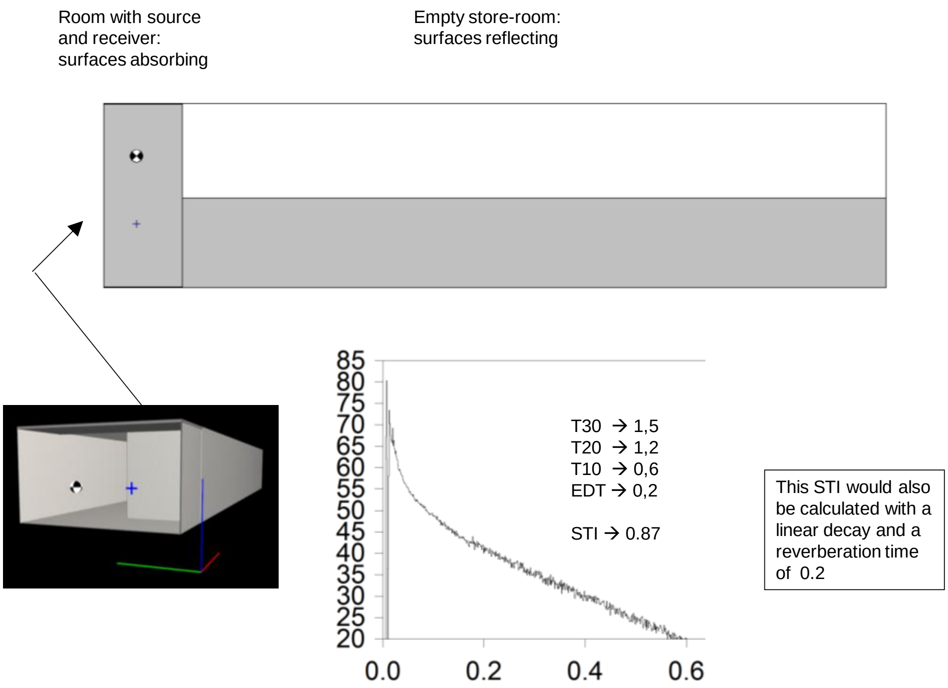


Figure 17: Acoustically damped room with coupled reverberant part

An estimation of the intelligibility of speech on the basis of 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.

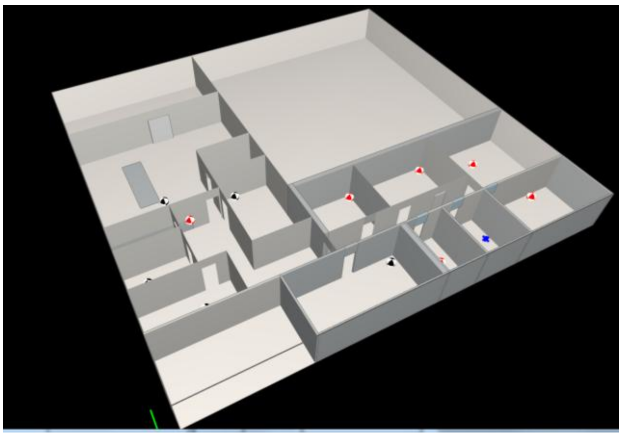


Figure 18: Many little office rooms acoustically linked by open doors

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. It can be seen, that the speaker will 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.

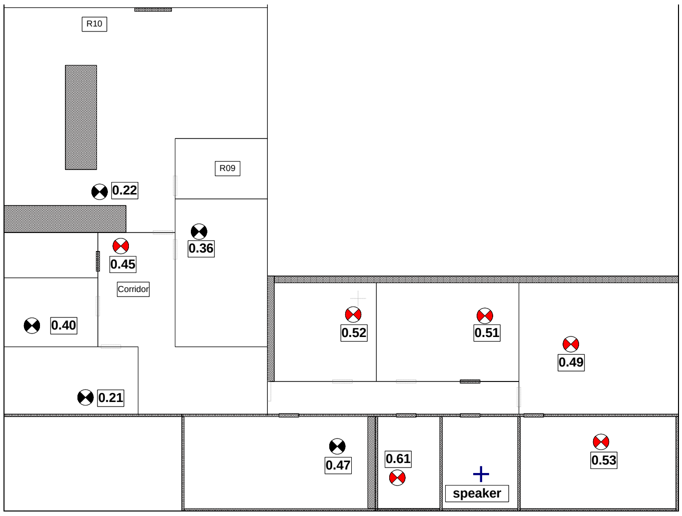


Figure 19: Calculated STI values in different rooms with the speaker in another room - doors are open

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

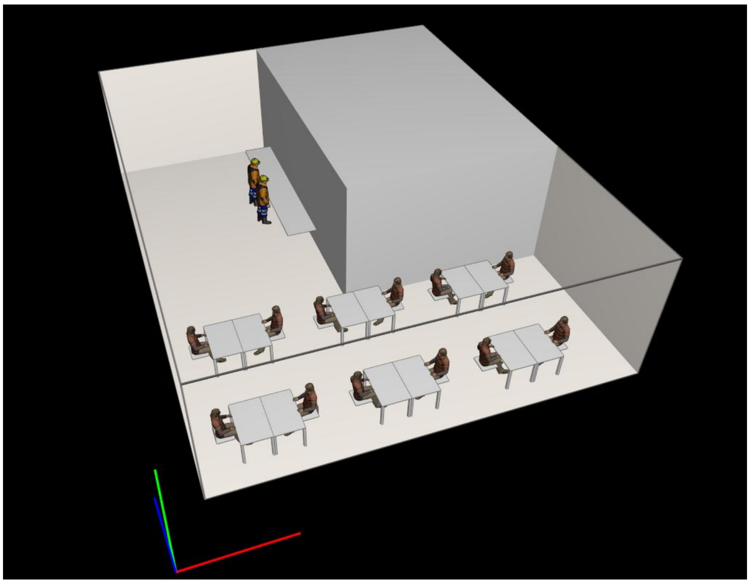


Figure 20: Model of a dining or office room with a food court area

The room is L-shaped, where one flank is a working or dining room with six tables for two persons sitting opposite and the other flank is a basically empty room with two persons at the buffet. An acceptable acoustic planning should ensure that each of the two people sitting opposite can communicate with normal relaxed speaking effort and that this communication is not 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 into account one source at each table with an emission due to normal relaxed speech. This is the recommended procedure generally 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 position at the table under test is calculated.

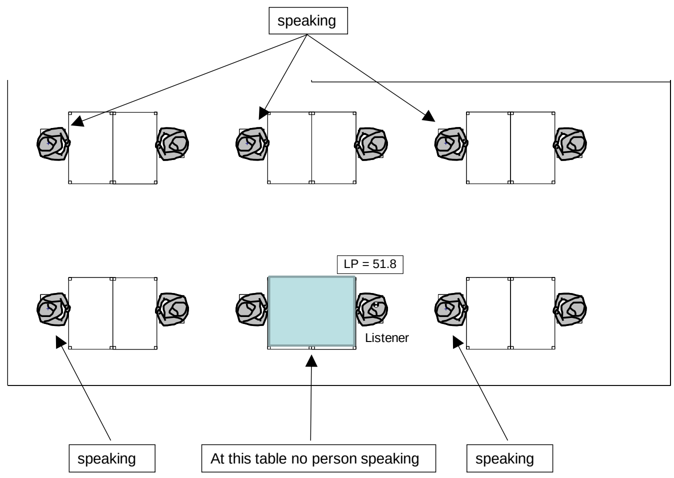


Figure 21: Calculation of background noise - at all other tables one person speaking normal relaxed

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

In step 2 the frequency spectrum of the background noise determined in step 1 is now taken as background noise with only the person left side of the table under test speaking.

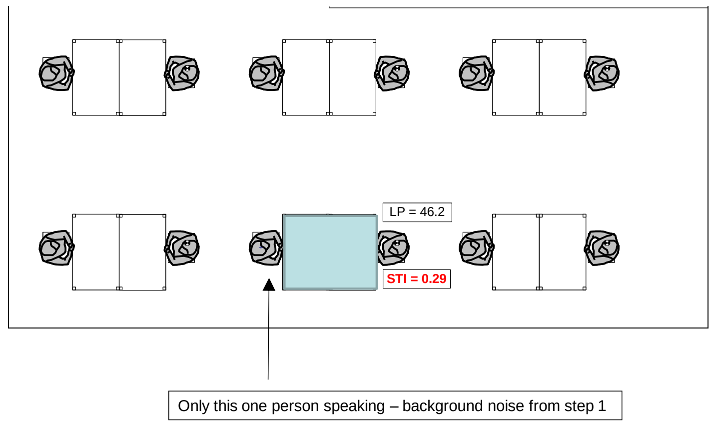


Figure 22: Calculation of the STI at the listener position with the background noise spectrum from step 1

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 the same do the others ending 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 - absorbing barriers are inserted between the tables and the wall is covered with an absorbing panel up to 1.6 m height to create around each table an acoustically damped individual space.

The two 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.

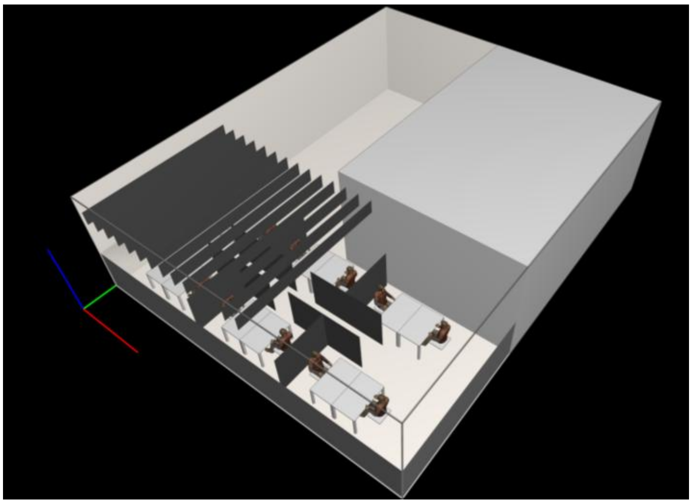


Figure 23: Absorbing screens and wall cover with baffle ceiling

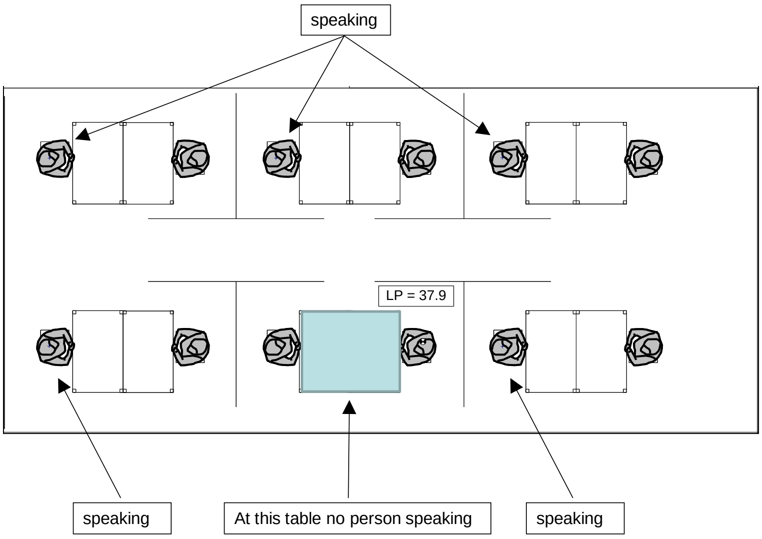


Figure 24: Determination of the background noise with 38 dB(A) in step 1

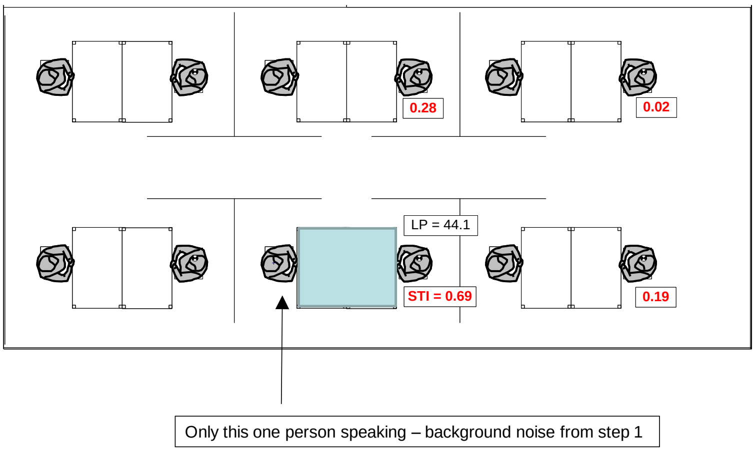


Figure 25: Calculation of the STI yields 0.7 due to the reduced background noise and better damped room response

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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