Author(s): Hongisto, Valtteri; Keränen, Jukka; Larm, Petra

Title: Simple model for the acoustical design of open-plan offices

Year: 2004

Version: Final published version

ACTA ACUSTICA UNITED WITH ACUSTICA
Vol. 90 (2004) 481 – 495

Simple Model for the Acoustical Design of Open-Plan Offices

Valtteri Hongisto, Jukka Keränen, Petra Larm
Finnish Institute of Occupational Health, Laboratory of Ventilation and Acoustics, Lemminkäisenkatu 14-18 B, FIN-20520 Turku, Finland. valtteri.hongisto@ttl.fi

Summary
According to questionnaire studies, speech is the most annoying sound source in open-plan offices. The most distracting speech comes from the nearest workstations. Therefore, speech privacy between workstations should be as high as possible. This presupposes that the speech is properly attenuated and masked. The aim of this study is to present a simple and validated model that can predict the speech transmission index, STI, between two nearby workstations. The speech level propagating different paths and masking noise level from different sources are calculated using conventional room acoustical principles. Speech through the screen, reflected speech via ceiling, diffracted speech over the screen and reverberant speech were involved. Masking noise sources comprise, e.g., ventilation, office equipments, and artificial masking system. Speech intelligibility was determined from the speech-to-noise ratio using the Modulation Transfer Function method, which gives the speech transmission index and the rapid speech transmission index, $RASTI$. The model was validated using 30 randomly selected workstations *in situ*. The prediction accuracy was -0.03 ± 0.04. Because the standard error of $RASTI$ measurements was approximately ± 0.02 in constant background level, the prediction accuracy of the model was found satisfactory.

PACS no. 43.55.Hy, 43.55.Ka, 43.66.Dc

1. Introduction

More than 50% of workers in Western Europe work in offices. The number of office workers is growing continuously. By using open-plan offices instead of single rooms, the worker density can be easily increased by 100…200%. This reduces rent expenses, at least in the short term. In open-plan offices, the most important source of distraction is usually the speech from neighbouring workstations [1]. It is not the level of speech that causes the distraction but its uncontrollability, information content and personal attitude to the speaker. It is obvious that when the speech intelligibility is reduced, the distraction is reduced as well [2].

The most important individual phase in acoustical design is the selection of proper workroom according to the need for concentration and privacy. Room acoustical remedies should not be designed before this process has been discussed properly because room acoustical remedies can seldom change dramatically people's impressions about the acoustical conditions of the room.

Building and room acoustics can be controlled technically by three main factors in open-plan offices. The first is room absorption, which prevents reverberation and early reflections. The second is screens, which cut the direct sound. The third is artificial masking sound, which gives a stable sound environment and masks the speech from nearby workstations. Good open-plan acoustics cannot be obtained unless these three factors are simultaneously considered.

It is typically not possible to reach satisfactory speech privacy only by absorption and screens. The speech level from the neighbouring workers should fall below background noise level. Screens and absorbents are never so effective that the speech level from the nearest desk could be attenuated below 40 dB(A), assumed that the distance between nearest workstations is 2…3 m. Because sound levels caused by ventilation are, typically, 35…40 dB(A) because of building regulations, extra masking sound systems are necessary to create satisfactory speech privacy in open-plan offices.

Beranek presented a well-known suggestion for the masking noise spectrum. At high frequencies, it has almost the same shape as that of standardized speech (Figure 1). Although some air handling devices produce very similar sound spectra, the use of special masking systems were recommended. The density of air supply devices is usually too low to produce smooth distribution of masking sound. The sound powers and spectra of individual devices are also impossible to adjust. The sound power level of ventilation can also vary depending on thermal conditions

Received 20 May 2003, Revised 22 December 2003, accepted 16 February 2004.

and the current application of the room. Therefore, ventilation is not the best possible masking sound source in a long term. Modern artificial masking sound is produced by small loudspeakers installed to the ceiling using 3…4 m dense grid. The loudspeakers are driven by a central amplifier, which produces pseudorandom noise or other sound having the long-term spectrum of Figure 1. The spectrum produced by any loudspeaker, or a group of them, can be equalized to compensate local differences in, e.g., room absorption or loudspeaker position. Loudspeakers are preferrably installed under the raised access floor or above the suspended ceiling to avoid visibility and misuse.

The use of artificial masking noise has been studied e.g. by Keighley and Parkin [3]. The highest level of satisfaction was found in offices with relatively high background levels. People rated masking noise levels above 46 dB(A) effective but unacceptable. Warnock [4] made two experiments with masking noise in a large office of floor area 1000 m^2. In the first experiment, the workers were informed about the research, where the acceptability of masking was studied with three levels, 45, 48 and 51 dB(A). Masking was unexpectedly rejected at all levels. According to interviews, the need for privacy was small because their work was of such a nature that intruding speech sounds did not distract them. Therefore, masking sound was not necessary in that workplace. The experiment indicates that people have to be disturbed by peoples conversations before acoustical consultant should consider masking as an acoustical remedy. After three months without masking noise, the masking level was increased from ambient noise by $1\frac{1}{2}$ dB per week, without telling to the workers about the experiment. People noted the noise and started to make complaints, when the level reached 47 dB(A).

Hegvold [5] studied which masking noise level workers will choose, if they can adjust the level themselves. The self-adjusted level after a long term experiment was approximately 48 dB(A). This coincides with the fact that people inherently start to raise their voice levels at normal conversation distances when the ambient level exceeds 48 dB(A) [6].

The first approach to the speech privacy design in offices was described by Beranek [7]. The first nearly completed model was presented by Moreland [8]. The model contained two nearby workstations separated by a screen and surrounded by a room with reflecting or absorbing surfaces. The propagation of speech from the speaker to the listener was possible through the screen, diffracted above the screen and reflected path via the nearby wall or ceiling. Also the effect of sound leaks around the screen was studied. Articulation index, AI, defined at that time in ANSI S3.5-1969, was used as the key design parameter [9]. The articulation index was determined from the signal-to-noise ratio of speech. The signal was considered as the transmitted speech level at the listener workstation and the noise contained all other sounds (ventilation, office machines, masking sound, etc.) reaching the workstation. Unfortunately, Moreland did not present comparison with experiments. Neither was the effect of reverberant speech considered. In addition, the parametric examples of different office parameters were given at a presumption that the background noise level follows PNC-35, e.g. 43 dB(A). According to our unpublished field studies, the average noise levels at open-plan workstations can vary between $L_{A,eq,8h} = 40$ and 57 dB(A), depending of the amount and level of speech. Thus, the background noise level is the most important parameter of acoustical design.

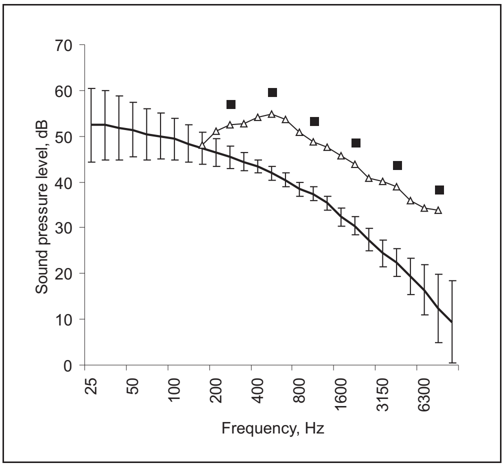

Figure 1. The masking noise spectrum suggested by Beranek [7] and the spectra of a normal effort male speech at a distance of 1 m in free field according to ANSI S3.5. The linear octave band values are 57.2, 59.8, 53.5, 48.8, 43.8 and 38.6 dB (re 20 μPa). ——: Masking noise spectrum, 48 dB(A); –△–: Third octave band spectrum of speech, 59.2 dB(A); ■: Octave band spectrum of speech, 59.2 dB(A).

Wang and Bradley studied recently the modelling of open-plan offices [10, 11, 12, 13] The first two studies presented a model for predicting the insertion loss of a single screen using the image source technique. It included the effect of reverberation between the ceiling and floor, multiple diffraction over the screen, the effect of ceiling height, screen and floor absorption as a function of sound incidence angle, and the dimensions of the room. The third and fourth study took also the effect of backwall screens and the position of passage opening into account. They used the speech intelligibility index, SII, in describing the acoustical performance between workstations. In our opinion, this model lacks practicability because it falls into unnecessary details. The model was validated in laboratory conditions in a workstation of a cubic form, which makes its application questionnable in an open-plan office. They also assumed that the workstations were empty, in which strong flutter reverberation can occur especially in vertical direction. However, in practical workplaces, the flutter echo is weak because of tables, chairs, monitors, shelves, books, things on the table and worker. The background noise spectrum conformed the room criteria spectrum RC-40 dB, which corresponds to 48 dB(A). The same level was used in all measurements. Because background noise is the most important individual factor affecting the signal-to-noise ratio of speech in open-plan offices, we cannot understand why so high and constant value was used. They only pointed out the combined importance of screen height, screen absorption and ceiling absorption. If a negligible background noise level were used, the full dependence of SII on room acoustical parameters would have been revealed. Now, the SII never exceeded 0.60 because of high background noise. On this basis, it is difficult to say how their model works in real open-plan offices where SII is typically higher than 0.60.

In real office design, only few parameters are known accurately. The model should be rapid to use and contain as little parameters as possible but also allow more specific modelling if needed. Such models have not been presented yet. Existing models require too much physical background to become understood by real practitioners, like architects or interior designers. The aim of this paper is to present a simple and field-validated model that can predict the speech transmission index, STI, between two workstations in an open-plan office. It should be easy to use considering the data available in practical workspace design. The results can be used by, e.g., an acoustical consultant, an architect, a material manufacturer, a system provider or a teacher.

2. Modelling methods

2.1. The model

The authors' experience is that the input and output parameters should be as simple as possible to quarantee wide general acceptance and high degree of usability. The model is aimed to open-plan offices, but it can be applied to conventional offices as well. The most serious limitation of this model is that it concentrates on two nearby workstations. Speech at distant workstations is not yet considered. However, concentration on two nearby workstations is justified in the first place because the speech sounds from the nearest workstations are the loudest and also most disturbing. Good design of nearby workstations, and application of this acoustical design throughout the office space, will guarantee good acoustics also in the far field. Therefore, far field acoustics has, obviously, secondary importance.

The principle of the model is shown in Figure 2. Seven separate sound "paths" or "sources" are considered. Paths 1–4 constitute different propagation paths of speech from nearby workstations. Paths 5–7 represent the most important background noise sources. The heights are marked by h_i, partial distances by x_i and path distances by d_i. Basic trigonometric formulas were used to solve the distances $x_3 - x_{14}$ and they are not presented here.

2.2. Path 0 – Direct speech

Path 0 produces the sound level at a distance of d_1 without a screen in a reflection-free field. This is used to calculate the incident energy on the screen and the received energy if screen is absent. Normal voice level according to ANSI S3.5 was used as the source level of a talker [14]. The sound level in front of a talker at 1 m distance in a free field is $L_{p,1m} = 59.23$ dB. The spectrum was shown in Figure 1. The free field sound levels were converted into sound power levels (re 1 pW) by

$$L_W = L_{p,1m} + 10 \log S = L_{p,1m} + 11 \text{ dB}, \quad (1)$$

where S (m^2) is the area of a spherical measurement surface at a distance of $r = 1$ m from the talker, i.e., $S = 4\pi r^2 = 12.56\,\text{m}^2$ (full space). According to our investigations in a semianechoic room, the effect of floor reflection on sound levels was between $0.1 \ldots 0.8$ dB at a distance of 1 m from a sitting person so that the assumption of full space was justified.

There is very recent evidence that voice levels in open-plan offices are lower than normal voice levels [15]. We have also observed that typical speech levels are 3-5 dB smaller than ANSI S3.5 expects (Figure 1). Most people probably adapt their voice effort depending on the expected need for privacy in the environment although individual differences in speech levels are large. Therefore, the ANSI sound level presented in Figure 1 is suggestive but the best available at the moment. The spectrum and level agreed well with our experiments in laboratory for a male talker.

The sound level of direct path depends on the sound power level of speech, distance between the talker and the listener and the sound reduction index, R, of the screen. The sound level of direct sound without a screen, $L_{p,0}$, is calculated as follows:

$$L_{p,0} = L_W + 10 \log \frac{Q}{\Omega d_1^2} = L_W - 10 \log \left(4\pi d_1^2\right), \quad (2)$$

where Q is the directivity factor, $d_1 = x_1 + x_2$, and Ω is the space angle of sound radiation. Radiation into full space was assumed here, i.e., $\Omega = 4\pi$.

Flanagan has studied the directivity pattern of a manikin talker, which has very similar directivity as a human talker [16]. The directivities of speech and hearing are close to a cardioid. Within $\pm 45°$ from the front of mouth, the emitted sound intensity is almost constant. At high frequencies, the intensity is 15 dB lower in the back of the head than in the front of it. At low frequencies, the difference between front and back radiation is less than 5 dB. The directivity of speech may have significance in certain office layouts where people work very close to each other, the space is very absorbing and the relative positions and orientations of worker's heads are almost fixed. That could be, e.g., in a call center, where the positions of worker's heads during phone calls are closely defined. In such situations, the speech directivity factor can have some significance. Also the directivity of hearing should be considered in such cases. Unfortunately, the consideration of directivities for hearing and speech are beyond the scope of this paper. The model is for average direction of the talker and

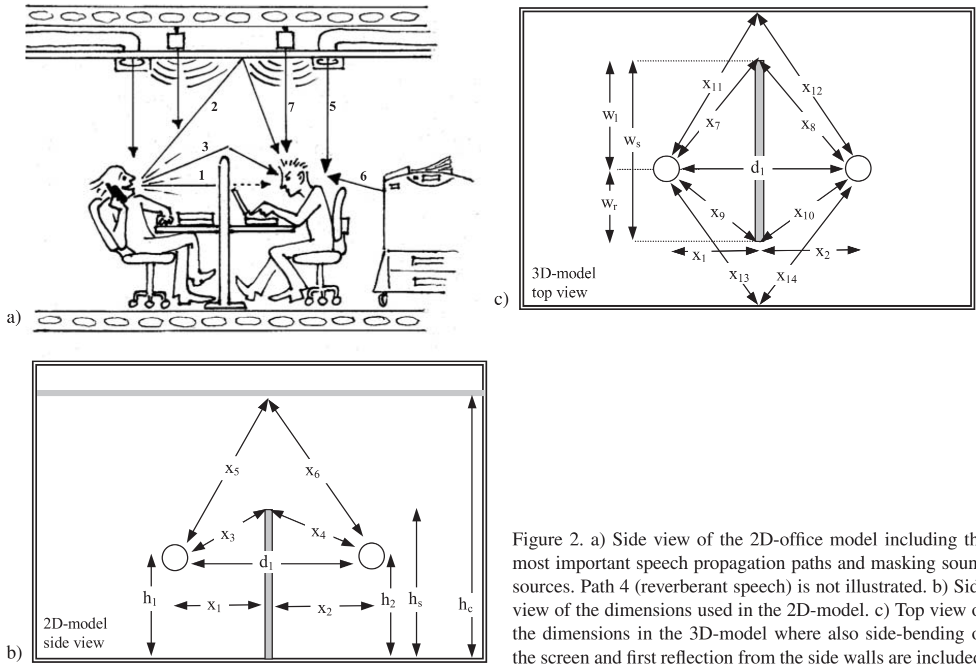

Figure 2. a) Side view of the 2D-office model including the most important speech propagation paths and masking sound sources. Path 4 (reverberant speech) is not illustrated. b) Side view of the dimensions used in the 2D-model. c) Top view of the dimensions in the 3D-model where also side-bending of the screen and first reflection from the side walls are included.

listener so that the directivity patterns of speech and hearing are not considered. Therefore, we use $Q = 1$.

2.3. Path 1 - Speech through the screen

The speech through screen is usually negligible compared with other paths since the SRI of a screen is generally larger than 15 dB as a consequence of structural requirements. However, in some cases the sound insulation can be important, e.g. when high lightweight screens are used in highly absorbing rooms. Therefore, the transmitted path is included in the model. The sound level of speech penetrating through the screen is approximated by

$$L_{p,1} = L_{p,0} - R. \tag{3}$$

In the absence of screen, $R = 0$ is used. The sound reduction index R (dB) is based either on measurements or prediction. The determination of the absolute screen sound reduction index is not necessary in most cases because the strength of path 1 is usually considerably lower than that of paths 2–4. Measurement results are also seldom available for screens. Therefore, approximation of R by the field incidence mass law is the easiest approach:

$$R = 20 \log \left(mf \right) - 47 \text{ dB}, \tag{4}$$

where m is the surface mass of the screen (kg/m^2) and f is the frequency of sound. Sound leaks are not considered in this model because sound via other paths will usually be much stronger. Sound leaks can have some contribution in total transmission only when other paths (2–3) are well damped. Sound leaks can be modelled by simple equations, which have been validated, e.g., by Hongisto [17] and Hongisto *et al.* [18].

2.4. Path 2 - Speech reflected from ceiling or walls

The reduction of sound intensity due to reflection from a surface is given by

$$\Delta L = 10 \log \frac{1}{1 - \alpha_c}, \tag{5}$$

where α_c is the absorption coefficient of the surface. In practice, absorption coefficient depends on the sound incidence angle. Angle-dependent absorption coefficients were not considered because such data are seldom available. Field incidence absorption coefficients determined in a reverberation room are recommended [19].

The most important individual room reflection takes place in the ceiling above the screen. It is obvious that reflected intensity cannot be considerably reduced without highly-absorbing ceiling absorbents. Absoption coefficient of $\alpha_c = 0.5$ will reduce the intensity of reflected sound only by 3 dB while $\alpha_c = 0.9$ reduces the level even by 10 dB.

The sound level of the reflected path, $L_{p,2}$, is obtained by:

$$L_{p,2} = L_W - 10 \log \left(4\pi d_2^2 \right) - \Delta L, \tag{6}$$

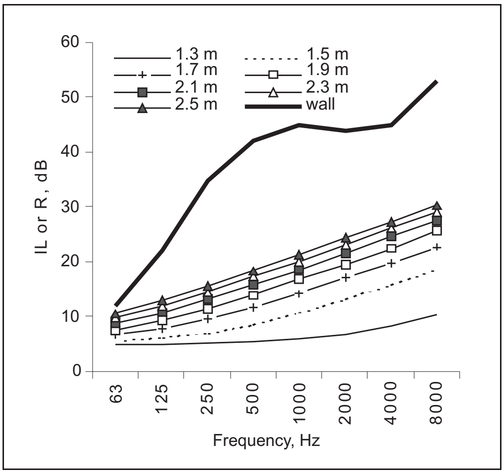

Figure 3. Screen insertion loss as a function of screen height in a reflection-free room. The sound insulation of the wall corresponds to a double plasterboard wall measured *in situ*. It was assumed that $x_1 = x_2 = h_1 = h_2 = 1.2$ m.

Table I. A-weighted sound level L_S [dB(A)] of standard speech behind the screen of height h_s [m] or in a nearby room separated by a double wall (compare to Figure 3).

h_c	0.0	1.3	1.5	1.7	1.9	2.1	2.3	2.5	wall
L_S	52	46	43	40	37	35	34	33	13

where the path length is $d_2 = x_5 + x_6$.

The estimation of ceiling reflection can be very difficult for certain ceiling types. If the ceiling contains strong forms like high beams, arches, tooth forms or niches, specular reflection is weak. This can be taken into account either by ignoring the reflected path totally or by increasing, in a sophisticated manner, the ceiling absorption coefficients so that the strength of specular reflection is reduced.

In the 2D-model, the ceiling is the only reflecting room surface. This approach is valid only in the middle of the room. In the 3D-model, reflection from the sidewall can be included. The calculation is similar to that of ceiling but the dimensions x_5 and x_6 are replaced by either x_{11} and x_{12}, or x_{13} and x_{14}. Backwall reflection was ignored in the calculations.

2.5. Path 3 - Speech diffracted over the screen

The insertion loss, IL, of the screen is defined as the difference of sound level at a certain receiver's position measured without and with the screen. During the measurements, a specified sound source at a certain position is used. That is, the insertion loss of the screen depends on the relative positions of source, receiver and screen.

Diffraction over a screen was thoroughly dealt with by Wang and Bradley [10]. In this study, the IL model of ISO 17624 was applied [20]

$$IL = 10\log\left(3 + 40\frac{z}{\lambda}\right), \tag{7}$$

where $z = x_3 + x_4 - d_1$ is the path length difference between the diffracted path and the direct path between source and receiver, and $\lambda = c_0/f$ is the wavelength of sound, f is the frequency and c_0 is the speed of sound (343 m/s).

Equation (7) is for free field. ISO 17624 gives also an alternative formula, which is applicable for cases where room reflections are present. However, in this model, room reflections are considered separately so that equation (7) is appropriate.

The speech level caused by path 3 is calculated by

$$L_{p,3} = L_{p,0} - IL. \tag{8}$$

The behaviour of equation (7) is presented in Figure 3. It is emphasized that the insertion loss values of Figure 3 are not obtained *in situ* because of room reflections.

The predicted free field speech levels behind the screen are presented in Table I. In principle, the speech levels can be reduced down to 35 dB using a screen but, in practice, speech levels below 40 dB were rare.

The 2D-diffraction model expects that the screen is infinitely wide. If the width of the screen, w_s, is of the same order than the effective screen height ($= h_S - h_1$), side bending can have a significant contribution to the total speech level on the receiving side.

The side-bending is included in the 3D-model. We divide here the screen width w_s into two parts along the shortest line between the source and the listener. Thus, the screen has a left and right part having widths w_L and w_R, respectively. The insertion loss via these paths can be calculated in a similar way as above for top-bending but x_3 and x_4 are replaced by x_7 and x_8 (left bending), or x_9 and x_{10} (right bending). Equations for side bending are not presented here because they can be easily derived from equation (7).

Sound absorbing screens were studied by, e.g., Butler [21]. When the surface is hard, standing waves can occur between the screen and backwall, which increases sound level in the source side. Absorbing screen can improve attenuation in these cases at most 6 dB. The effect of screen absorption was not considered in this study because the absorption coefficients of commercial office screens seldom exceed 0.5, even at very high frequencies. In addition, backwalls are seldom hard and smooth in real offices so that standing waves are not strong.

2.6. Path 4 – Reverberant speech

In speech analysis, there is a coarse rule that room reflections arrived within 50 milliseconds from the direct sound are considered as useful (early reflections) and reflections arrived later (late reflections) are considered as harmful for speech intelligibility. The paths 0–3 deal with the most important early reflections. Because their path difference is small, they amplify each other, and, thus, improve the speech intelligibility. If room reverberation is negligible, paths 1–3 constitute most of the measured speech levels. However, if room reverberation is not negligible, reverberant speech can have a significant contribution to the total speech level.

In this study, the Modulation Transfer Function Method is used to estimate the effect of reverberation on speech intelligibility. In this method, the total speech level is considered as the sum of direct and reverberant speech components. Direct sound was already determined in path 0/1. What we still need to know is a simple method that estimates the sound level caused by reverberant speech at a receiver's position.

E.g. Hodgson [22] has compared empirical simplified models for the prediction of sound propagation in industrial workrooms. However, the distance between source and receiver is small in typical open-plan offices, typically below 3 m. Therefore, propagation models are not implemented here because they primarily aim at good accuracy at long distances.

For a point source, the sound level as a function of distance, r, in diffuse rooms follows equation [23]

$$L_p = L_W + 10\log\Big(\frac{Q}{\Omega r^2} + \frac{4(1-\alpha_R)}{S\alpha_R}\Big), \tag{9}$$

where S is the total room surface area (m^2) and α_R is the average absorption coefficient of the room. The first term in parenthesis of equation (9) corresponds to direct sound and the second to reverberant sound. The direct path was already dealt with in equation (2). Here, reverberant speech level, $L_{p,4}$, was calculated by using $r = \infty$.

The easiest way to obtain α_R is to measure the reverberation time T (s) and room volume V (m^3). The Sabine's equation relates reverberation time and total room absorption area, $S\alpha_R$:

$$T = 0.163\frac{V}{A} = 0.163\frac{V}{S\alpha_R}. \tag{10}$$

If measurement is not possible, the average absorption coefficient can be predicted from the surface absorption coefficients, α_i, of surfaces i having area S_i by

$$\alpha_R = -\frac{1}{S}\sum_i S_i \ln\big(1-\alpha_i\big), \tag{11}$$

where S (m^2) is the total area of room surfaces.

The behaviour of these equations is presented in Figure 4. The increase of reverberation time from 0.3 to 0.9 seconds increases the total speech level by 2 dB at a distance of 2.4 m from the speaker because of the increased proportion of reverberant speech. This example is valid when screen is absent. If there is a screen between the talker and a listener, we should look at the change in the reverberant speech level. It increases even by 8 dB when the reverberation time increases from 0.3 s to 0.9 s.

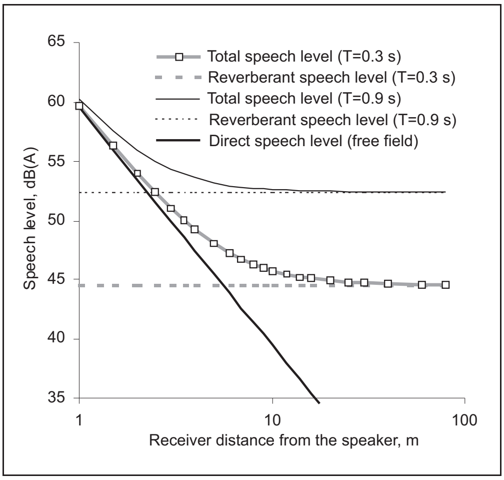

Figure 4. Propagation of sound from a point source according to equation (9) with two different reverberation times T. The dimensions of the office were $20\times20\times23$ m^3.

There are three difficulties in the determination of reverberant speech level. Firstly, it is difficult to estimate the effective volume of an office room. Effective volume means the room space, which is coupled freely to the receiver point. The total room volume can be much larger than the effective volume. For instance, if the office is very low and absorbing, and screens are high, isolated cubicles are formed which are are weakly coupled to the rest of the room. Secondly, the determination of reverberation time is difficult. Normal reverberation time of the room, T_{20}, can be longer than the local early reverberation time, T_{10}, because screens eliminate effectively horizontal room modes. Thirdly, office rooms are not absolutely diffuse, which violates with equation (9). Therefore, reverberation does not follow exponential decay and spatial distribution disagrees with Figure 4 at large distances.

In this study, there was a possibility to measure the reverberation times so that the average room absorption coefficients, α_R, could be determined directly by equation (10). The exact room volume was used in calculations except when the workstation was not coupled to the rest of the room. In some cases, the screen was very high so that an isolated cubicle was formed having only a weak coupling to the rest of the room. In that case, reverberant speech was ignored.

2.7. Paths 5, 6 and 7 – Noise caused by ventilation, office equipment and masking system

Sound levels caused by ventilation, office equipment and masking system are denoted by $L_{p,5}$, $L_{p,6}$, and $L_{p,7}$, respectively. In field conditions, paths 5–7 have to be measured simultaneously because, in most cases, separation of masking sound sources is neither possible nor necessary. This was also the situation in our field measurements.

Table II. Subjective rating of rapid speech transmission index, $RASTI$, as a descriptor of speech intelligibility and speech privacy.

$RASTI$	Speech intelligibility	Speech privacy	Examples in office
0.00…0.05	very bad	confidential	Between two single person office rooms, high sound insulation
0.05…0.20	bad	good	Between two single person office rooms, normal sound insulation
0.20…0.40	poor	reasonable	Between workstations in a high level open-plan office Between two single person office rooms, doors open
0.40…0.60	fair	poor	Between desks in a well-designed open-plan office
0.60…0.75	good	very poor	Between desks in an open-plan office, reasonable acoustical design
0.75…0.99	excellent	no	Face-to-face discussion, good meeting rooms Between desks in an open-plan office, no acoustical design

2.8. Calculation of speech intelligibility using MTF method

The objective of the prediction model is to express the output simply, preferably in a single-number form. We used speech intelligibility as the main quantity because the most distracting sound source in open plan offices is speech, and the acoustical design should aim at adequate speech privacy.

There are two main approaches to calculate the speech intelligibility. The first is Modulation Transfer Function (MTF) method, which gives the speech transmission index, STI [24]. The second is the audiologic method described in ANSI S3.5, which gives the speech intelligibility index, SII [14]. In this study, we use only the MTF method because ANSI method gives equal results with the MTF method in unechoic rooms but MTF method takes the effect of reverberation into account. The subjective meaning of speech intelligibility, $RASTI$ and speech privacy is presented in Table II. In the following, the MTF method is summarized. It does not take the directivity of the speaker or the distance between the source and the speaker into account. It applies for a point source in a diffuse sound field. The speech-to-noise ratio is defined by

$$L_{SN} = L_S - L_N. \tag{12}$$

The total speech level, L_S, at the listener's location is the sum of paths 1–4:

$$L_S = 10 \log \sum_{i=1}^{4} 10^{L_{p,i}/10}. \tag{13}$$

The total masking level, L_N, is calculated with the same equation but $i = 5, 6, 7$.

The first step is to calculate the modulation reduction factor $m(F, f)$ on 14 modulation frequencies F_i (0.63, 0.80, 1.00, 1.25, 1.60, 2.00, 2,50, 3.15, 4.00, 5.00, 6.30, 8.00, 10.00 and 12.5 Hz) and at seven octave bands f_j (range 125, 250, 500, 1000, 2000, 4000 and 8000 Hz) by

$$m(F, f) = \frac{1}{\sqrt{1 + \left(T(f)\, 2\pi F/13.8\right)^2}} \cdot \frac{1}{1 + 10^{-L_{SN}(f)/10}}, \tag{14}$$

where $T(f)$ is the local reverberation time determined from the early part of the reverberation curve (Early Decay Time, T_{10}). The apparent signal-to-noise ratio is calculated for all 98 m values by

$$SN_{app} = 10 \log \frac{m}{1 - m}. \tag{15}$$

If $SN_{app} > 15$ dB, we use $SN_{app} = 15$ dB. Similarly, if $SN_{app} < -15$ dB, we use $SN_{app} = -15$ dB. The speech transmission index is calculated by

$$STI = \frac{15 + \sum_{j=1}^{7} k_j \left(\frac{1}{14} \sum_{i=1}^{14} SN_{app}(F_i, f_j) \right)}{30}, \tag{16}$$

where band weighting constants k_j are 0.13, 0.14, 0.11, 0.12, 0.19, 0.17 and 0.14 at 7 octave bands 125 …8000 Hz, respectively. The shortened version of STI, rapid speech transmission index, $RASTI$, uses only two frequency bands and nine modulation frequencies in total [25]:

$$RASTI = \frac{1}{2} + \frac{1}{270} \sum_{i=3,6,9,12} SN_{app}(F_i, 500\,\mathrm{Hz}) + \frac{1}{270} \sum_{i=2,5,8,11,13} SN_{app}(F_i, 2000\,\mathrm{Hz}). \tag{17}$$

In the literature, slightly different modulation frequencies are used for the determination of $RASTI$ but the expected error is negligible.

To facilitate the concept of speech intelligibility index, Figure 5 was drawn. It describes schematically the dependence of $RASTI$ on reverberation time with different speech-to-noise ratios. It was assumed in Figure 5 that both masking spectrum and transmitted speech spectrum have the same shape and reverberation time is the same in all frequency bands.

It should be noted that the increase of reverberation time has two effects: modulation reduction factor, m, decreases which decreases $RASTI$ and the reverberant speech level, $L_{p,4}$, increases, which increases $RASTI$. These effects partially cancel each other.

Table III. Room parameters, measurement results and prediction results in 30 test subjects. Legends: d_1 = distance between workers [m], w_s = width of the screen [m], h_s = screen height [m], surf = surface material of screen (S=soft, H=hard), h_c =room height [m], V = effective room volume [m^3], α_C = ceiling absorption coefficient (average of 500–4000 Hz), L_N = measured background noise level (sum of paths 5–7) [dB(A)], T_{10} = early decay time, $RASTI$ = rapid speech transmission index, L_{SN} = speech-to-noise ratio [dB], error = difference between predicted and measured $RASTI$.

Measured values											Predicted values		
case	d_1	w_1	h_s	surf	h_c	V	α_c	L_N	T_{10}	$RASTI$	L_{SN}	$RASTI$	error
1	3.0	1.6	1.50	H/S	2.2	440	0.35	42.1	0.30	0.81	9.2	0.73	-0.08
2	3.0	2.1	1.50	S	2.3	440	0.05	37.1	0.30	0.82	12.1	0.76	-0.06
3	3.0	0.0		–	2.3	440	0.05	37.1	0.30	0.89	15.0	0.78	-0.11
4	3.0	6.0	2.13	H	2.9	–	0.05	28.7	0.40	0.81	18.5	0.78	-0.03
5	2.4	4.2	1.27	S	3.2	1280	0.05	41.6	0.30	0.76	8.2	0.75	-0.01
6	2.4	2.4	1.27	S	3.2	1280	0.05	41.6	0.30	0.75	8.1	0.75	0.00
7	2.4	2.0	1.27	S	3.2	1280	0.05	41.6	0.46	0.75	8.1	0.75	0.00
8	2.4	8.0	1.27	S	3.2	1280	0.05	40.2	0.46	0.75	9.2	0.69	-0.06
9	2.4	1.6	1.25	H	3.2	1280	0.05	40.2	0.46	0.74	9.7	0.70	-0.04
10	2.4	1.6	1.27	S	2.6	260	0.70	37.6	0.34	0.84	11.2	0.79	-0.05
11	2.4	0.8	1.27	H	2.6	260	0.70	37.6	0.34	0.82	12.1	0.79	-0.03
12	2.4	2.4	1.59	S	2.4	480	0.50	38.4	0.29	0.81	9.3	0.76	-0.05
13	2.4	2.0	1.59	H	2.4	480	0.50	38.4	0.29	0.79	9.4	0.76	-0.03
14	2.4	2.4	1.59	S	4.5	630	0.05	38.4	0.61	0.66	7.3	0.65	-0.01
15	2.4	17.6	1.59	S	4.5	630	0.05	38.4	0.61	0.63	6.8	0.65	0.02
16	2.4	2.0	1.59	H	4.5	630	0.05	38.4	0.61	0.68	7.3	0.65	-0.03
17	3.0	3.3	1.65	H	2.5	560	0.50	31.7	0.18	0.89	14.3	0.89	0.00
18	3.0	3.3	1.65	H	2.5	560	0.50	31.7	0.18	0.87	14.3	0.89	0.02
19	3.0	5.2	1.65	H	3.0	670	0.75	31.7	0.18	0.88	11.6	0.84	-0.04
20	3.0	2.7	1.66	H	3.0	670	0.75	31.7	0.18	0.84	11.9	0.84	0.00
21	2.4	3.2	1.33	S	2.8	280	0.55	36.8	0.24	0.90	13.9	0.86	-0.04
22	2.4	4.0	1.33	S	2.8	280	0.70	37.4	0.23	0.85	10.3	0.84	-0.01
23	2.4	5.5	2.05	S	2.8	280	0.40	38.4	0.26	0.79	7.8	0.79	0.00
24	2.4	5.5	2.05	H	2.8	280	0.70	37.9	0.29	0.77	6.1	0.79	0.02
25	3.2	3.0	1.18	H	2.8	280	0.55	38.1	0.29	0.81	12.2	0.83	0.02
26	2.4	1.6	1.33	S	5.1	2040	0.80	42.1	0.32	0.65	4.9	0.59	-0.06
27	2.4	3.2	1.65	S	5.1	510	0.80	35.9	0.32	0.71	6.8	0.71	0.00
28	2.4	3.0	1.68	H	5.1	510	0.80	36.1	0.32	0.75	6.4	0.73	-0.02
29	2.4	1.6	1.33	S	5.1	2040	0.80	36.6	0.26	0.71	10.4	0.68	-0.03
30	2.4	6.0	2.08	H/S	5.1	510	0.80	36.2	0.39	0.72	6.2	0.71	-0.01
												Average error:	-0.024
												Standard deviation of the error:	0.031

2.9. Application of the model to single-person office rooms

The model can be applied between two single-person office rooms. In that case, path 1 is the most important speech propagation path. The source room speech level can be calculated by equation (9). The sound reduction index, R, of the partition wall should be based on measurements or a more sophisticated prediction model than mass law of equation (5). E.g. Hongisto has summarized and validated simple prediction models for lightweight double walls [17, 18].

Speech level transmitted to receiving room is

$$L_{p,2} = L_{p,1} - R + 10 \log \frac{S_{12}}{A_2}, \tag{18}$$

where S_{12} is the partition wall area (m^2) and A_2 is the absorption area of receiving room determined by equations (10)–(11).

In equation (18), $L_{p,2}$ contains paths 1–4. It should be noted that the same wall results in different STI-values in different masking conditions.

3. Subjects

To validate the model, 30 different workstations were studied *in situ*. The workstations were located in 7 different premises. The modelled workstations were selected randomly in the workplaces to provide as versatile acoustical conditions as possible. The subjects represent well typical Finnish offices so that the validation material should be acceptable for the evaluation of the model. The room parameters are presented in Table III.

4. Measurement methods

The following room parameters were gathered in the offices: screen height h_s, screen width w_s, screen material,

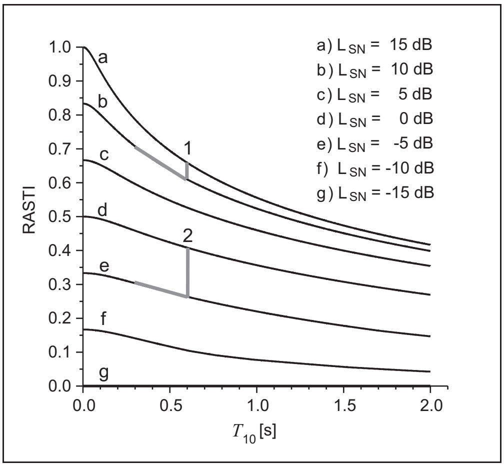

Figure 5. The dependence of $RASTI$ on early decay time, T_{10}, and speech-to-noise ratio, L_{SN}, in a diffuse far field of a point source. Typical values of open-plan offices are 0.30 s$< T <$1.0 s and -5 dB$< L_{SN} <$15 dB. Reverberation time was set equal in all frequency bands. Transitions 1 and 2 are explained in section 6.5.

Figure 6. Measurement of $RASTI$ between two workstations. The screen is situated behind the palm tree. The source apparatus is on the right side.

room height h_c, ceiling absorption material, and percent of ceiling absorbent covered by fixtures, like lighting and ventilation ducts.

The acoustical measurements were performed between 5 p.m. and 10 p.m. to minimize the disturbance of working activities. Source and receiver points were arranged, whenever possible, so that $h_1 = h_2 = x_1 = x_2 = 1.2$ m. This corresponded to normal positions of workers in most situations.

The dominating background noise source was ventilation. It was measured in the receiver point as an equivalent sound level ($L_{eq,15s}$) in 1/1 octave bands. This level represented paths 5–7.

The absorption coefficient of ceiling was determined by observing the material type, thickness and suspension height, and finding the declared absorption coefficient, α_d, from manufacturer's material databases. The degree of ceiling area covered by lighting, ventilation supplies and hard frame panels surrounding these elements was also estimated. The effective ceiling absorption coefficient *in situ* was calculated by

$$\alpha_c = (1 - C)\alpha_d, \tag{19}$$

where C is the ratio between uncovered absorbent area and total absorbent area. Typically, 10–30% of ceiling area was covered. Therefore, higher values than $\alpha_c = 0.8$ were not used.

The measurement of rapid speech transmission index, $RASTI$, was carried out by a standard device (Brüel & Kjær 3361). During the measurements, the only background noise was caused by the HVAC systems. The averaging time was 32 seconds. The method is very sensitive to random changes in background noise. Therefore, the measurements were repeated until the variations between successive measurements were between 0.05. The measurement arrangement is presented in Figure 6.

Reverberation time was measured in workstations with a real time analyzer (Norsonic 840-2) using MLS-method and backwards integration. One measurement was performed per point. The placement of the microphone and the source was the same as above in the $RASTI$ measurements. Reverberation time was determined from the early part of decay (early decay time, T_{10}, EDT) but also T_{20} was measured for comparison.

All frequency-dependent results are presented as an average of octave bands within 500…4000 Hz. Sound levels are presented as A-weighted values. Speech-to-noise ratios are given as an average of octave bands 500…4000 Hz.

The prediction accuracy of the MTF method is defined as the difference between predicted $RASTI$ and measured $RASTI$. The prediction errors of $RASTI$ were averaged over 30 samples. This method gives the trend of the error. The standard error gives the range within which the prediction error will be with a probability of 68%.

5. Results

The summary of room parameters, measurement results and prediction results for 30 subjects are presented in Table III. The MTF method underestimated the $RASTI$ value by 0.024. The standard error was ± 0.031. The relations between the measured and predicted $RASTI$ values are shown in Figure 7.

6. Discussion

6.1. Prediction accuracy of the model

The model underestimates $RASTI$, on average, by 0.024. The measured value will be, with a probability of 68%, within ± 0.031 from the predicted value. Example: If the prediction result is 0.68, measurement result will be, with a probability of 68%, within 0.67 and 0.73. This is sufficient accuracy regarding field applications. It seems that

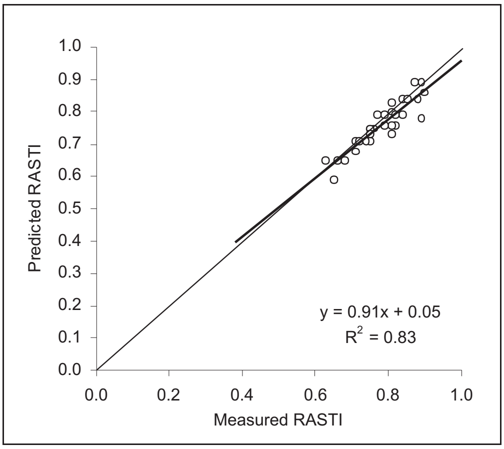

Figure 7. Relation between measured and predicted $RASTI$ values. The prediction accuracy of the model was -0.03 ± 0.04.

the model could be useful in workplace design and, especially, in comparison of different acoustical designs both in existing and new workplaces. Abovementioned prediction accuracy holds for absolute $RASTI$ value when the workstation parameters of Table III are known. It can be assumed that the model is more reliable in predicting the changes in $RASTI$ than its absolute value. Once the model is calibrated to a certain acoustical environment, changes in $RASTI$ can be predicted even more reliably.

In Figure 7, linear fitting curve fits quite well with the data in the $RASTI$ range 0.63–0.90 with satisfactory small bias and very close to the intentional unity slope. Extrapolation down to $RASTI$=0.40 was justified because there should be no confounding variables that could deflect the model. Below $RASTI$=0.40, the increase of masking noise is the only factor that could further decrease $RASTI$ values. The MTF method reacts monotonically to the increase of L_N (see Figure 5). Therefore, there are no serious reasons to doubt the validity of the model in the $RASTI$ range 0.00–060.

Unfortunately, we could not find nearby workstations with $RASTI$ <0.60. The subjects of this study represent typical Finnish open-plan offices where ventilation levels are nearly always less than 40 dB. Therefore, $RASTI$ values below 0.50 cannot exist at typical speaker-listener distances ($d_1 = 2.4\,\mathrm{m}$), despite the amount of ceiling absorption or height of screens. Of course, we could have added some constant background noise during test situations with loudspeakers but that was not found meaningful.

We are going to publish later data on laboratory tests where the most important room acoustical parameters, screen height, ceiling absorption and masking level, could be controlled and the full range of STI values could be validated. We did not want to present laboratory values in this study because the simultaneous analysis of field and laboratory data would be very incoherent.

The measurement of $RASTI$ can be difficult in field conditions if the background noise level is varying. The standard deviation of repetitive measurements in silent laboratory conditions was 0.02. In field conditions, the variations were 0.02…0.06 in nearly constant conditions. Measurements *in situ* were carried out repeatedly until a stable background noise could be achieved. Finally, measurements were carried out 2–5 times from which the average was taken. With these arrangements, we assume that the standard deviation of field measurements was below 0.04.

6.2. Sensitivity analysis of the model

It is very difficult to present a qualitative sensitivity analysis of the model because of great number of prediction variables and their nonlinear interactions. The prediction accuracy does not depend only on the partial errors of the input variables but also on the range of L_{SN} and T_{10}. Therefore, it was found easier to present a short mathematical sensitivity analysis. The aim of this analysis is to present the dependence of $\Delta RASTI$ on the main input parameters and their absolute errors.

The speech level in the receiver point was calculated by (no wall reflections)

$$\begin{aligned}
L_S = {} & 10\log\Big[10^{\left(L_W - 10\log(4\pi d_1^2) - R\right)/10} \\
& + 10^{\left(L_W - 10\log(4\pi d_2^2) + 10\log(1-\alpha_c)\right)/10} \\
& + 10^{\left(L_W - 10\log(4\pi d_1^2) - 10\log(3+40(x_3+x_4-d_1)/\lambda)\right)/10} \\
& + 10^{\left(L_W + 10\log 4(1-\alpha_R)/S\alpha_R\right)/10}\Big] \\
= {} & 10\log\Big[10^{L_W/10}10^{-R/10}\frac{1}{4\pi d_1^2} \\
& + 10^{L_W/10}\frac{1}{4\pi d_2^2}(1-\alpha_c) \\
& + 10^{L_W/10}\frac{1}{4\pi d_1^2}\frac{1}{3+40(x_3+x_4-d_1)/\lambda} \\
& + 10^{L_W/10}\frac{4(1-\alpha_R)}{S\alpha_R}\Big].
\end{aligned} \tag{20}$$

The absolute error of $RASTI$, $\Delta RASTI$, can be derived from equations (15) and (17) to the form

$$\begin{aligned}
|\Delta RASTI| \le {} & \sum_{i=1}^{9}\left|\frac{\partial RASTI}{\partial m_i}\Delta m_i\right| \\
= {} & \frac{1}{9\cdot 3\cdot \ln(10)}\sum_{i=1}^{9}\left|\frac{1}{1-m_i}\frac{\Delta m_i}{m_i}\right|.
\end{aligned} \tag{21}$$

The relative error of each modulation reduction factor, $|\Delta m_i/m_i|$, can be derived from equation (14). It depends on reverberation time, T, absolute error of it, ΔT, modulation frequency, F, speech-to-noise ratio, L_{SN} and the absolute error of it, ΔL_{SN}, at octave band j:

$$\left|\frac{\Delta m_i}{m_i}\right| \le \left|\frac{\partial \ln(m)}{\partial T_j}\Delta T_j\right| + \left|\frac{\partial \ln(m_i)}{\partial L_{SN,j}}\Delta L_{SN,j}\right| = \frac{T_j}{(13.8/2\pi F_i)^2 + T_j^2}\Delta T_j + \frac{\ln 10}{10}\frac{1}{1+10^{L_{SN,j}/10}}\Delta L_{SN,j}. \quad (22)$$

Because the relative error of each modulation reduction factor depends on the modulation frequency, F, $\Delta RASTI$ was determined over all 9 modulation frequencies used in the determination of $RASTI$.

The absolute error of speech-to-noise ratio is

$$\left|\Delta L_{SN}\right| \le \left|\Delta L_S\right| + \left|\Delta L_N\right|. \quad (23)$$

We assume now that the error in path 3 (diffraction) is negligible because its partial error depends only on the measurement error of distances from the screen and floor. Although the error of sound insulation, ΔR, can be significant, the effective error of path 1 is presumably negligible because its contribution on the speech level is, usually, negligible. Thus, the absolute error of speech level depends mainly on the partial errors of α_c and α_R:

$$\left|\Delta L_S\right| \le \left|\frac{\partial L_S}{\partial \alpha_c}\Delta\alpha_c\right| + \left|\frac{\partial L_S}{\partial \alpha_R}\Delta\alpha_R\right|. \quad (24)$$

(It should be noted that the error of T affects not only on ΔT but also on $\Delta\alpha_R$ because it is usually defined by equation (10).)
From equations (5) and (6), the first partial error of path 2 is

$$\left|\frac{\partial L_S}{\partial \alpha_c}\Delta\alpha_c\right| = \frac{10}{\ln 10}\frac{1}{4\pi d_2^2}10^{(L_W-L_S)/10}\Delta\alpha_c. \quad (25)$$

After equation (9), the second partial error of path 2 is

$$\left|\frac{\partial L_S}{\partial \alpha_R}\Delta\alpha_R\right| = \frac{10}{\ln 10}10^{(L_W-L_S)/10}\frac{4}{S\alpha_R^2}\Delta\alpha_R. \quad (26)$$

To facilitate the analysis, we use the following initial values: $d_2 = 3.5\,\mathrm{m}$, $L_W = 70.23\,\mathrm{dB}$ (speech spectrum), $L_S = 55\,\mathrm{dB}$, $\Delta\alpha_c = 0.1$, $\alpha_R = 0.5$, $\Delta\alpha_R = 0.1$ and $S = 118\,\mathrm{m}^2$. Then we can estimate the partial errors of speech level caused by the errors of ceiling absorption coefficient and average room absorption coefficient:

$$\left|\frac{\partial L_S}{\partial \alpha_c}\Delta\alpha_c\right| \le 1\text{ dB} \quad \text{ja} \quad \left|\frac{\partial L_S}{\partial \alpha_R}\Delta\alpha_R\right| \le 2\text{ dB}.$$

This means that the absolute error of speech level is $\Delta L_S \le 3\,\mathrm{dB}$. In field measurements, one can assume that the measurement error of background noise is $|\Delta L_N| \le 2\,\mathrm{dB}$. This implies that the total error of the speech-to-noise ratio is, in the worst case, $|\Delta L_{SN}| \le 5\,\mathrm{dB}$. In further analysis we assume that $|\Delta L_{SN}| \le 3\,\mathrm{dB}$.

The absolute error of rapid speech transmission index, $\Delta RASTI$, as a function of reverberation time, can be calculated using equations (21) and (22). Five curves with different initial values of L_{SN} and ΔT are presented in Figure 8. It shows several features of the prediction uncertainties. The larger the reverberation time is, the smaller is $\Delta RASTI$. This implies that the errors are more probable in rooms with short reverberation time. It is also evident that the smaller L_{SN} is, the smaller is the influence of changes in T. That is, if we have a small speech-to-noise ratio, the less accurately we need to know the reverberation time.

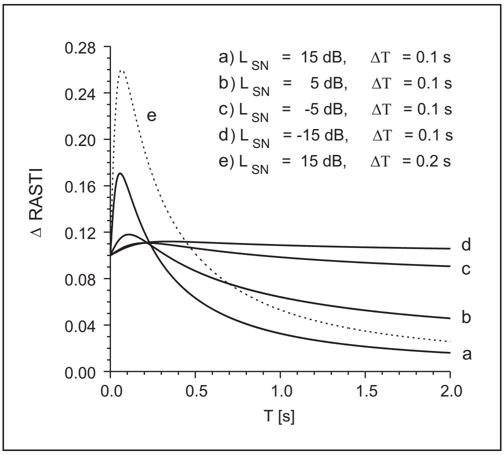

Figure 8. The dependence of the absolute error of speech intelligibility index, $\Delta RASTI$, on the reverberation time, T, and signal-to-noise ratio, L_{SN}. The error of the speech-to-noise-ratio was ΔL_{SN}=3 dB.

The reverberation time is within 0.3 and 1.0 seconds and L_{SN} between adjacent workstations is above -5 dB in most open-plan offices. According to Figure 8, the mathematical error is, then, within $\Delta RASTI = \pm 0.04 \ldots 0.11$. The largest error occurs with short reverberation time. The mathematical error is larger than the prediction errors of Table III. The mathematical error evaluation gives the highest possible error (worst case estimate) provided that the partial errors are correctly selected.

6.3. Early decay time

In this study, we had an opportunity to measure the early decay times. It is preferred in the prediction of STI instead of T_{20} because it gives information about the early reflections, which are advantageous for the speech intelligibility. In practical design this is not always possible. It is possible to estimate the reverberation time, T_{20}, by equations (10)–(11) by using absorption material databases. However, the estimation of local early decay time can be more difficult because it is typically lower than T_{20} in complex and highly absorbing rooms like open-plan offices. MTF method will underestimate the STI value if too large reverberation times are used. Therefore, we recommend that, in open-plan offices, reverberation times, in general, should always be measured between two workstations at a height

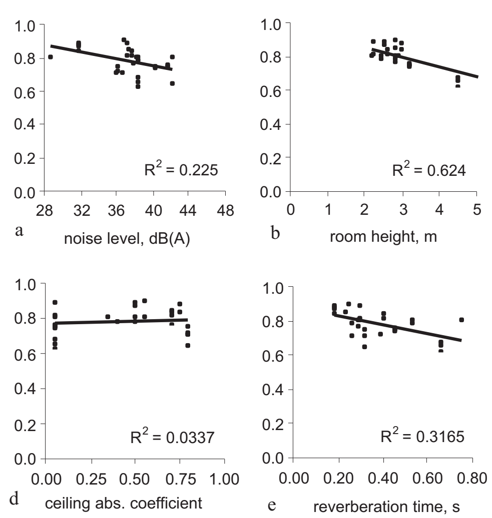

Figure 9. Dependence of measured $RASTI$ on individual room acoustical variables. These relations hold only for the subjects used in this study ($n = 30$). The slopes shall not be applied generally because $RASTI$ depends on all these parameters simultaneously.

of 1.2 m, and not in open areas at a height of 1.5 m as recommended in ISO 3382.

It was interesting to compare the early decay time T_{10} and "normal" decay time T_{20} in different rooms. In open rooms having smooth distribution of absorption, the difference between T_{10} and T_{20} was negligible. However, when the ceiling was strongly absorbing and the rest of the room was reverberant, T_{10} could be even 0.3 seconds shorter than T_{20}. This means that local reverberation ceases rapidly (vertical modes) but after that the rest of the room (horizontal modes) starts to feed sound energy to the measurement point. This increases T_{20} but not T_{10} because the end of the decay curve was bended, not straight. It seems that the difference between T_{10} and T_{20} is the larger the weaker is the coupling between the listener point and the rest of the room. However, a model to predict T_{10} would be useful for office applications because Sabine's method gives, in principle, correct values only for T_{60}.

6.4. Screen performance in open-plan offices

Performance of screens is a very popular topic in all applications of noise control. Insertion loss, IL, is determined as the sound level difference without and with the screen. ISO 10053 describes a laboratory method for free field conditions. Correspondingly, ISO 11821 describes a method *in situ* giving no restrictions for reverberation. It would have been very laborious to determine the IL in workplaces because it is usually impossible to remove screens from their fixed locations. We see that insertion loss has no practical significance in the acoustical design of offices because the screen performance depends strongly on room reflections. It is much more informative to measure the speech level, L_S, in the receiver point (field attenuation of speech) using a sound source calibrated to the standard speech level (see Figure 1), and compare it to speech level without a screen.

6.5. Effect of individual room parameters on STI

Correlations between independent room acoustical design variables and measured $RASTI$ values are presented in Figure 9. They were drawn to demonstrate that individual room acoustical design parameters are inappropriate predictors of speech privacy. According to Figure 9a, $RASTI$ decreases with increasing background noise level. This is an obvious consequence of masking. According to Figure 9b, $RASTI$ decreases with increasing room height. This is also obvious because intensity of reverberant speech decreases with increasing room height. This means also that it is not so important to have highly absorbing ceilings in high rooms than in flat rooms. According to Figure 9c, screen height and $RASTI$ do not correlate at all. The reason to this odd result is that ceiling absorption and masking noise levels were not high enough in most rooms. It seems that it is not meaningful to increase screen heights until other room acoustical parameters are properly designed. According to Figure 9d, the increase of ceiling absorption coefficient slightly increases $RASTI$ values. The same holds for decreasing reverberation time in Figure 9e. This is a natural consequence of the MTF theory. High reverberation always decreases the $RASTI$ value. This is evident from Figure 5.

It is interesting to note from Figure 5 that simultaneous reduction of L_{SN} and T can result in the increase of $RASTI$ when L_{SN} is large but to the reduction of $RASTI$ when L_{SN} is small. E.g., when reverberation time is reduced from 0.6 to 0.3 seconds and speech-to-noise ratio is reduced by 5 dB, the $RASTI$ is increased (see transition 1 in Figure 5), when L_{SN} is initially larger than 10 dB,

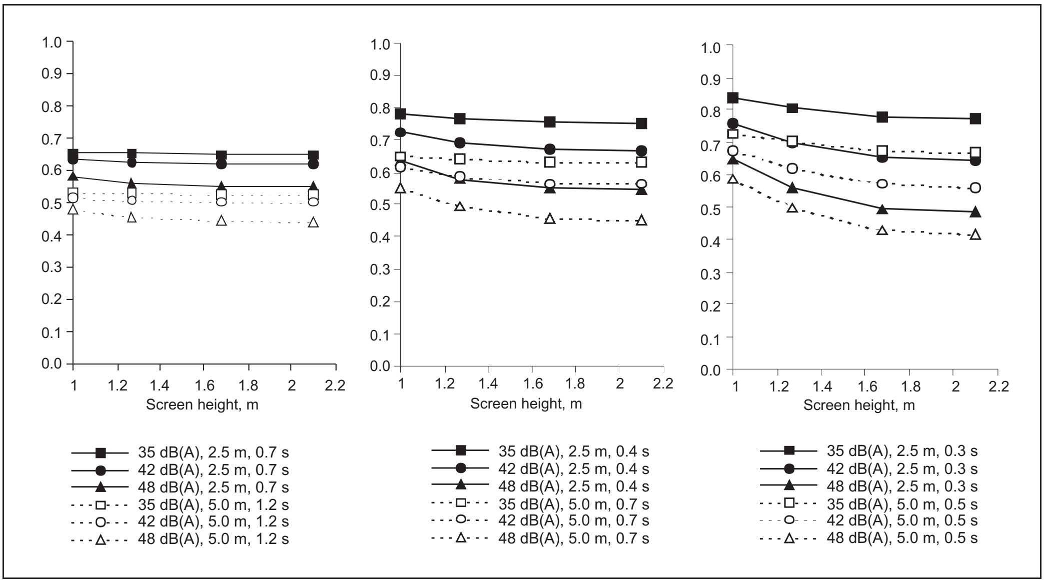

Figure 10. The effect of screen height on $RASTI$ in three different cases of ceiling absorption: $\alpha = 0.10$, $\alpha = 0.60$, and $\alpha = 0.90$. In each figure, the calculation was made on three different levels of background noise, 35, 42 and 48 dB. Whole lines represent room height $h_C = 2.5$ m and dotted lines $h_c = 5.0$ m. The last number in the label is the reverberation time, which is calculated by a hypothetic room of area 400 m^2.

but $RASTI$ is increased when L_{SN} is initially below 5 dB (see transition 2 in Figure 5).

The installation of ceiling absorption reduces path 2 but it reduces also ventilation noise. As a result, the $RASTI$ value typically increases between nearest workstations because of added absorptive panels, if masking levels are initially low. However, if the masking levels and screens were sufficiently high, also the increase of ceiling absorption would cause the decrease in $RASTI$. Without sufficient masking noise, absorbents can cause nonfavourable changes in local speech intelligibility. Figures 9d and 9e do not imply that ceiling absorbents were useless. The reduction of speech level is always important. Ceiling absorbents are the most important means to reduce speech levels at large distances.

Figure 9 is very valuable information for material manufacturers and workplace designers. It should motivate them to take the all-inclusive acoustical design seriously and to support co-operation between system providers. However, the trendlines of Figure 9 should not be generalized. Figure 9 is just to demonstrate the importance of the mutual relations of all design parameters. The slopes are based on a certain population of 30 workstations representing typical acoustical conditions in Finnish offices. E.g., if background noise levels had been considerably higher at all workplaces, the slopes would have been different for screen height, reverberation time, and ceiling absorption. In that case, the slopes of Figures 9d and 9e would have different signs than now, as will be shown next in Figure 10.

6.6. Conversion of the model into simple design charts

The aim of acoustical design of open-plan offices is proper attenuation and masking of speech. So far, simple design charts, which could help designers to understand the importance of all-inclusive design, have not been published. The main factors affecting the speech privacy, e.g., ceiling absorption, screens, room height and masking sound, were summarized to Figure 10. Figure 11 indicates the corresponding speech levels. It should be noted that Figure 10 must not be used alone. One can think, on the basis of Figure 10, that the use of hard ceiling, low screens and high background noise would be the cheapest way to obtain reasonably low STI values. However, this kind of design would lead to complaints regarding reverberation, overall noisiness and poor speech intelligibility.

Figures 10a, 10b and 10c correspond to hard ceiling ($\alpha_c = 0.1$), semi-absorbing ceiling ($\alpha_c = 0.6$) and highly-absorbing ceiling ($\alpha_c = 0.9$), respectively. This division is justified because there are clear product families available for these groups. Higher values than $\alpha_c = 0.9$ are difficult to obtain *in situ* because of the cover of lighting and ventilation. The calculation was made independent of frequency so that the same value of α_c, L_N and T was given for every frequency band. Two different room heights were also used.

It was necessary to give an estimate for the reverberation time in the three cases of Figure 10. As mentioned before, the prediction of early reverberation time is difficult and cannot be made reliably using equations (10) and

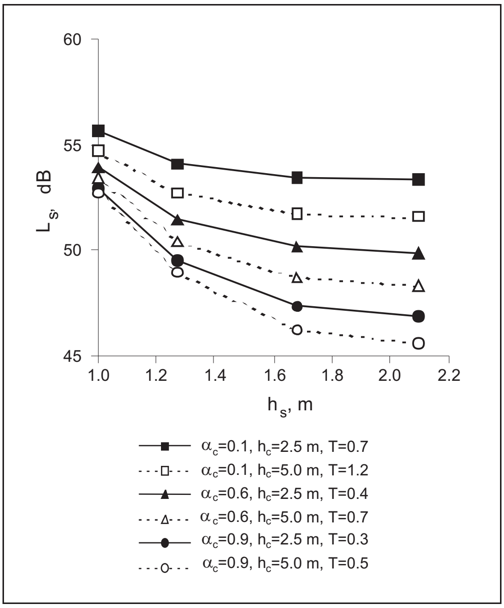

Figure 11. The effect of screen height, h_s, room height, h_c ceiling absorption coefficient, α_c, and reverberation time, T_{10}, on speech levels on the listener side of the screen, L_S.

(11). Typically, local early decay times are shorter than the average room reverberation times. However, in Figure 10, the reverberation time was calculated using a hypothetic room of area 20 m×20 m. The walls were reasonably hard ($\alpha = 0.15$) and the floor was hard ($\alpha = 0.04$). The ceiling absorption was changed according to α_c. The room was full of furniture with an area of 400 m^2 and absorption coefficient $\alpha = 0.40$.

In open-plan offices, one should aim at low $RASTI$ values, which leads to high speech privacy. According to Figure 10, $RASTI$-values below 0.50 are possible but it requires 48 dB(A) masking level, 1.6 m screens, room height of 5 m and full ceiling absorption ($\alpha_c = 0.9$). Lower $RASTI$ values can be obtained by increasing the distances between workers, which was assumed to be 2.4 m in Figure 10.

According to Figure 10, ceiling absorption has unfavourable influence on $RASTI$ value when masking is weak. The less ceiling absorption, the lower $RASTI$ value we get. This would lead to a false conclusion that the more reverberation, the better speech privacy. The problem of Figure 10 is that $RASTI$ does not tell the absolute speech level. Therefore, Figure 10 should be used together with Figure 11 giving the predicted speech levels in the same conditions. One can see that ceiling absorption coefficient has much stronger importance on speech levels than screen height. The higher reverberation time we have the higher speech level is. The variations of speech levels can be even 10 dB.

Figure 10 contains certain carefully selected assumptions: sidewall reflections are absent, screen sound insulation is infinite (no leaks), masking and speech spectra follow Figure 1 and distances are $h_1 = h_2 = x_1 = x_2 = 1.2$ m. However, we believe that the assumptions agree sufficiently with most offices so that the selection of correct acoustical parameters can be facilitated in early stages of office design.

6.7. Limitations of the model

There are numerous factors of the interior acoustical design, which are not dealt with in this study, e.g., tapping noise caused by people walking in the corridors, telephones ringing, traffic noise through the facade, sounds from nearby meeting rooms and structure borne sound caused by, e.g., cooling unit or ventilation plant upstairs. Although they all can be sources of a major annoyance, they cannot be seen as a part of a general design tool because they have to be solved structurally.

6.8. Extension of the model to far distances

The prediction model presented in this study does not deal with longer distances than the nearest work desk. It would be valuable to predict also the effect of acoustical remedies at longer distances, within an area of radius 20 m. For decision makers, it would be very interesting to know the office area disturbed by a single speaker in square meters because it is more easily convertible to space economy measures than speech intelligibility quantities. Although the principle of $RASTI$ and Table II is easy to understand, the $RASTI$ quantity lacks the space economy examination. We have made preliminary work on sound propagation in different types of open-plan offices. Measurements of sound propagation at distances 5…20 m have been made over areas with and without screens. The decay of sound per distance doubling, denoted by DL_2 in ISO 14257, has varied within 3 and 10 dB depending on frequency, room height, absorption, locations of absorbents, and screen height. This agrees with previous studies. It seems that the model for predicting DL_2 should be based more or less on empirical data. The speech disturbance area around a talking person could be defined, e.g., by the condition $L_{SN} > 0$. The distance, at which this equation is no longer valid, can be easily calculated by using the previous model behind the screen (point 2.4 m) and with a decay curve for more distant workstations (range 5…20 m). Preliminary tests showed that this method could be a very effective in the design of the whole office room. The extension of the model to larger distances will be made in the future.

7. Conclusions

A reliable and simple method was developed, which can be applied in the room-acoustical design of open-plan offices. The model serves as a valuable tool in demonstrating the importance of all-inclusive design. We believe, that this model will, in a long run, improve the quality of acoustical environments in open-plan offices. We also see the power of the model in proving that single rooms are still necessary when high speech privacy is required or when job complexity places high requirements for the work environment. It is very difficult to find solutions to open-plan offices which were acceptable both by an acoustician and an interior designer.

Acknowledgement

I would like to thank Ms. Riikka Helenius, Dr. Jarkko Hakala and Mrs Jaana Jokitulppo for their help in field measurements. This study was a part of national research programme Productive Office 2005 (Tuottava Toimisto 2005) funded by Tekes (National Technology Agency), Finnish Work Environment Fund and several participating companies. Finally, thanks are due to the companies and contact persons who let their premises at our disposal, project managers, who gave valuable comments for the applicability of the model, and guards who gave us the best possible company during long evenings.

References

[1] J. Nemecek, E. Grandjean: Results of an ergonomic investigation of large-space offices. Human Factors **15** (1973) 111–124.

[2] A. Kjellberg, U. Landström: Noise in the office: Part II - The scientific basis (knowledge base) for the guide. Inter. J. Industr. Erg. **14** (1994) 93–118.

[3] E. C. Keighley, P. H. Parkin: Subjective responses to sound conditioning in a landscaped office. J. Sound Vib. **64** (1979) 313–323.

[4] A. C. C. Warnock: Acoustical privacy in the landscaped office. J. Acoust. Soc. Am. **53** (1973) 535–1543.

[5] L. W. Hegvold: Experimental masking noise installation in an open planned office. Technical Note No. 563, National Research Council Canada, Division of Building Research, Canada, 1971.

[6] R. N. Hamme, D. N. Huggins: The problem of acoustical specifications for office landscape ceilings. J. Acoust. Soc. Am. **52** (1972) 120.

[7] L. L. Beranek: Criteria for noise and vibration in communities. Buildings and vehicles, Ch. 18. – In: Noise and Vibration Control. L. L. Beranek (ed.). McGraw-Hill Book Company, New York, USA, 1971.

[8] J. B. Moreland: Role of the screen on speech privacy in Open Plan Offices. Noise Con. Eng. J. **30** (1988) 43–56.

[9] ANSI S3.5-1969: Methods for the calculation of the articulation index. American National Standards Institute, New York, USA, 1969.

[10] C. Wang, J. S. Bradley: A mathematical model for a single screen barrier in open-space offices. Appl. Acoust. **63** (2002) 849–866.

[11] C. Wang, J. S. Bradley: Prediction of the speech intelligibility index behind a single screen in an open-plan office. Appl. Acoust. **63** (2002) 867–883.

[12] C. Wang, J. S. Bradley: Sound propagation between two adjacent rectangular workstations in an open-plan office – Part I: mathematical modeling. Appl. Acoust. **63** (2002) 1335–1352.

[13] C. Wang, J. S. Bradley: Sound propagation between two adjacent rectangular workstations in an open-plan office – Part II: effects of office variables. Appl. Acoust. **63** (2002) 1353–1374.

[14] ANSI S3.5-1997: Methods for calculation of the speech intelligibility index. American National Standards Institute, New York, USA, 1997.

[15] A. C. C. Warnock, W. T. Chu: Speech levels in open plan offices. J. Acoust. Soc. Am. **110** (2001) 2664.

[16] J. L. Flanagan: Analog measurements of sound radiation from the mouth. J. Acoust. Soc. Am. **32** (1960) 1613–1620.

[17] V. Hongisto: Sound insulation of doors - Part 1: Prediction models for structural and leak transmission. Journal of Sound and Vibration **230** (2000) 133–148.

[18] V. Hongisto, J. Keränen, M. Lindgren: Sound insulation of doors - Part 2: Comparison between measurement results and predictions. Journal of Sound and Vibration **230** (2000) 149–170.

[19] ISO 354-1985 (E): Acoustics - measurement of sound absorption in a reverberation room. International Organization for Standardization, Geneva, 1985.

[20] Draft international standard ISO/DIS 17624: Acoustics - guidelines for noise control in offices and workrooms by means of acoustical screens. International Organization for Standardization, Geneve, 2001.

[21] G. F. Butler: A note on improving the attenuation given by a noise barrier. J. Sound Vib. **32** (1974) 367–369.

[22] M. Hodgson: Experimental evaluation of simplified models for predicting noise levels in industrial workrooms. J. Acoust. Soc. Am. **103** (1998) 1933–1939.

[23] L. L. Beranek (editor): Noise and vibration control.

[24] T. Houtgast, H. J. M. Steeneken: A review of the MTF concept in room acoustics and its use for estimating speech intelligibility in auditoria. J. Acoust. Soc. Am. **77** (1985) 1069–1077.

[25] IEC 268-16:1988: Sound system equipment, Part 16: The objective rating of speech intelligibility in auditoria by the RASTI method. International Electrotechnical Commission, Geneva, Switzerland, 1988.