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Simulation of Sound Field in a Classroom

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Introduction

Speaking and listening are the essential communication ways in education. The noise level and classroom characteristics should be such that speech produced by lecturers and students would be intelligible. Many researchers have shown the deleterious influence of excessive classroom noise (signal to noise ratio – SNR) and reverberation time (T_r) on speech-recognition [1-4]. If an acoustic environment allows +15 dB SNR throughout the classroom, students with normal hearing can fully receive the spoken message well enough [1].

There are many methods of speech intelligibility measurement that take into account the room sound reflections and noise level. They can be classified into three groups: 1) methods that use the acoustical energy ratio concept: the available acoustical energy (direct + reflected + noise) is sum of useful part (direct + earlier arriving reflected) and a detrimental part (later arriving reflected + noise) [1-2]; 2) An experimental based procedure that gives an expected articulation score as a function of the reverberation time and the signal-to-noise ratio [3]; 3) Measurement of the speech transmission index (STI) and use the concept of the modulation transfer function [4].

However, it is shown that the STI cannot be considered to correspond to intelligibility because it does not distinguish useful early energy from non-early energy, which does not contribute to intelligibility [5].

Although the methods of all groups are different, an experimental study showed their values to be strongly correlated [6].

The total sound field of room acoustics is divided into the direct sound field and the reverberant sound fields by a further subdivision into useful and detrimental parts. The useful part includes the sound waves which the human ear can identify as part of the direct sound and of the first sound wave respectively. According to experiments the reflected sound waves arriving at a listening place within same delay after the first wave front. Delay period was defined from T_1 (35 ms) to T_2 (95 ms) within which the intensity of the arriving sound waves is to be considered with decreasing weight. The detrimental share comprises all the reflected sound waves arriving later than T_2 [2]. Useful-to-detrimental sound ratio was called signal-to-noise ratio Rsn , and can be calculated according to

$$R_{SN} = 10\lg\left[\frac{\int_{0}^{0.095} p_r(t)\, p(t)\, dt}{\int_{0.095}^{\infty} p(t)\, d(t) + 10^{(L_n - L_r)/10}}\right], \tag{1}$$

where $p_r(t)$ is the fraction of the reflected sound energy, $p(t)$ is the room sound power density impulse response, L_n – noise pressure level, L_r reflected sound level

For the linear relation for $p_r(t)$ signal-to-noise ratio R_{SN} can be found in the form [3]

$$R_{SN} = 10g\left[\frac{1 + (r_h / r)^2 + 1.21 T_r \left(e^{-\frac{1.31}{T_r}} - e^{-\frac{0.48}{T_r}}\right)}{e^{-1.31/T_r} - 10^{(L_n - L_r)/10}}\right], \tag{2}$$

where T_r – reverberation time.

The ratio of useful to detrimental share in dB is called signal-to-noise ratio SNR and is a measure of intelligibility and articulation as well as of the clarity or clearness of musical reproductions in a room.

Several authors suggested that a 50-ms early time limit was appropriate for speech.[7]

The articulation loss of consonants Al_{cons} is based on tests. These tests resulted in the development of an empirical relationship, without background noise, where the articulation loss of consonants is expressed as a function of the distance to the source r, the room volume V, and the reverberation time T_r, in the form [3]

$$Al = (200\, r^2\, T_r^2)/V, \tag{3}$$

where Al is the articulation loss of consonants, %.

The impulse response completely defines the properties of a room. Thus, the steady-state response and the reverberant response can both be determined from the impulse response. If the sound field in the room is diffuse the sound power density describing the room impulse response for $t>0$ can by written in form [3]

$$p(t) = \frac{qP}{4\pi c}\left[\frac{1}{r^2}\delta(t) + \frac{k}{r_h}e^{-kt}\right], \tag{4}$$

where $\delta(t)$ is the Dirac delta function at $t=0$, q and P are the directivity index and the sound power of the acoustic source, respectively, and c is the sound velocity, r is the distance from the acoustic source to the receiver, and r_h is the reverberation distance, defined as the distance for which the reflected sound energy density equals the direct sound energy density.

It has been determined that for classrooms the reverberation time T_r that maximizes the speech intelligibility is in the range between 0.1 and 0.3 s. But in very quiet classrooms, 100% speech intelligibility can be achieved with a reverberation time of 0.4– 0.5 s. [3].

The Acoustical Society of America [1] recommends a classroom signal-to-noise ratio of + 15 dB.

Same formulas are used for T_r calculation in many cases of simulation. The Sabine equation is used when absorption of room surfaces is $\alpha \leq 0.2$ [8]

$$T_r = \frac{0{,}164\,V}{\alpha\,S}, \tag{5}$$

where V the room volume, S surface area.

The Eyring equation is used for calculation RT in low frequencies area [8]

$$T_r = \frac{0{,}071\,V}{-S\lg(1-\alpha)}. \tag{6}$$

The Kutruff equation is used when it is necessary take in the account different reflection coefficient from differences surfaces of room and attenuation in an air [9].

$$T_r = \frac{0.16\ V}{S\ [-\ \ln(1-\alpha)] + \Delta + 4mV}, \tag{7}$$

where Δ – the average reflection coefficient of surface area Sn m – absorption sound energy in the air of room.

Experiment results

The program CARA 2.2plus was used to model the sound field of auditorium.

When selecting the coating (perforated and no hardboard panels) [9] of the surface of one of the walls, the necessary values of T_r were received practically in the entire speech frequency range. When students are present in auditorium, T_r considerably decreases in the zone of the low frequencies due to the increased sound energy absorption in the low frequency range. Some illustrative modeling results are presented in Fig. 1.

When investigating the response to the pulsed sound signal in the modeled auditorium, the measured reverberation time T_r was 0.36 s, thus in essence it coincided with the modeling results in the medium frequency range.

According to the modeling results the total room absorption index in the entire frequency range varies from 0.19 to 0.24.

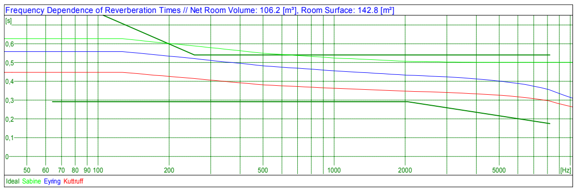
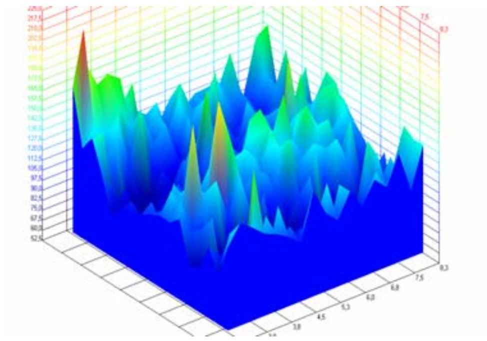


Fig. 1. Values of reverberation time T_r as function of frequency, calculated using three different formulas

Three variants of the sound sources were modeled: voice of a lecturer, speaker system of the types 5.1 and 7.1. The most even sound pressure level (SPL) was received for speaker system 7.1, but the natural voice of the lecturer is enough to obtain acceptable SPL in entire auditorium.

The distribution of SPL in respect of listener positions is one of the most important characteristic. Since student sitting places are fixed according to furniture construction, SPL distribution can be optimized by selecting acoustic systems. Two cases of modeling results are presented in Fig. 2 (here and further in the figures the positions of sound sources and 30 listeners are not indicated, so that modeling results in the pictures would be seen clearly).



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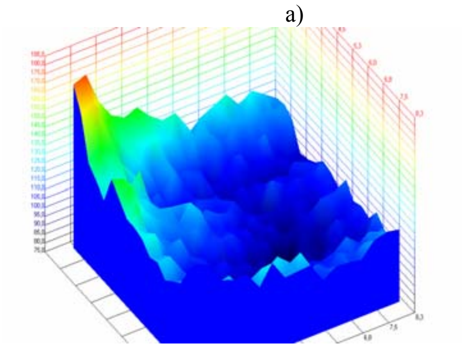


b)

Fig. 2. Sound pressure level: a) sound source voice of lecturer; b) loudspeakers system 7.1

When using loudspeaker system 7.1 the more precise sound addressing to the listeners is obtained (deflections in Fig. 2 show lesser deviation of the purpose function value from the maximum). By selecting the parameters of signals which are fed into loudspeakers even better addressing can be achieved.

Listener perceives the virtual sound source. An interesting modeling result was received when the voice of a lecturer is considered as sound source (Fig. 3).

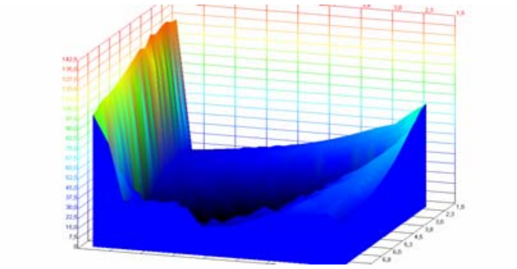
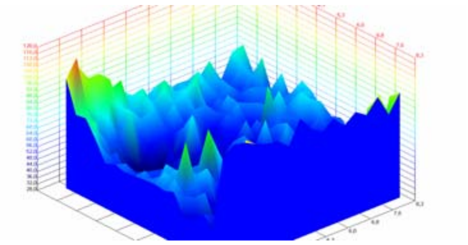


Fig. 3. The reception of a virtual sound source

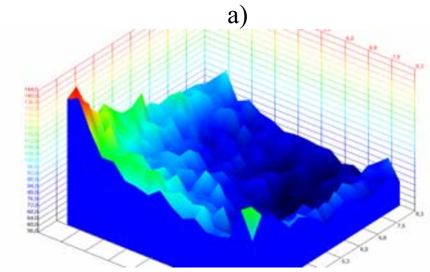
The virtual sound source is perceived worst near the walls, especially near the front wall; therefore listeners localize the sound source position incorrectly due to the reverberations. When using 7.1 systems, virtual sound sources are distinguished most evidently in the left corner, were subwoofer of the sound system is placed.

Speech intelligibility in the positions of listeners is one of the main characteristics of auditorium, and it is related to the signal-to-noise ratio (SNR).

Speech intelligibility modeling results are presented in Fig. 4 a, b, c.



a)



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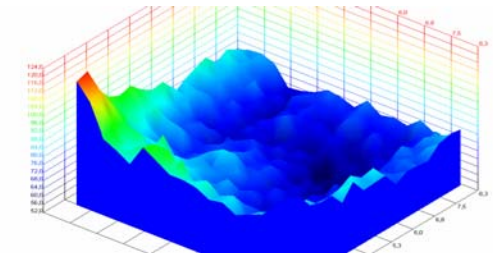


c)

Fig. 4. Speech intelligibility in the classroom: a) sound source voice of lecturer; b) loudspeakers system 5.1; c) loudspeakers system 7.1

The best speech intelligibility is achieved when the listener sits in 1.5 m distance from the speaking lecturer, and when using loudspeaker system 7.1 the areas of best speech intelligibility are considerably wider. Results indicate that in order to increase speech intelligibility it is required not only to optimize the placement of the speakers, but also optimize parameters of the signals fed into loudspeakers.

SPL distribution over the frequency range in all modeled cases significantly differs only below 180 Hz. More typical results are presented in Fig. 5 and Fig. 6.

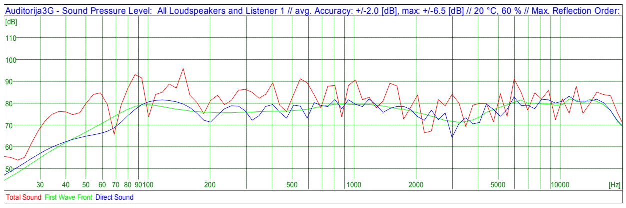


Fig. 5. Sound pressure level when sound source voice of lecturer.

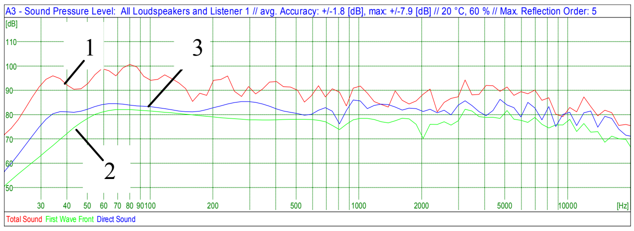
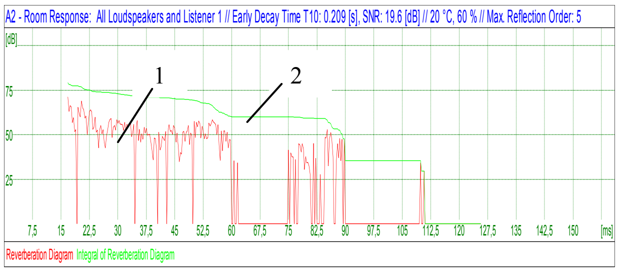
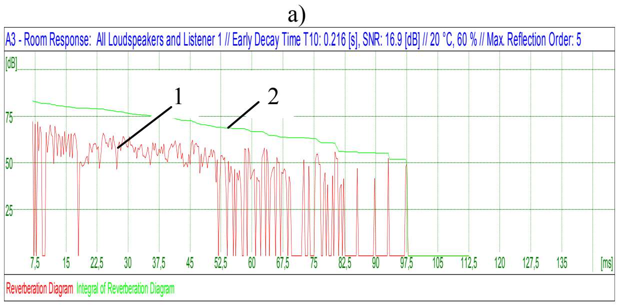


Fig. 6. Sound pressure level when sound source loudspeakers system 7.1
1 – total sound, 2 – first wave front, 3 – direct sound

Received results (Fig. 5 and Fig. 6) show that more even distribution of SPL is actually in the low frequency region when using complex loudspeaker system.



a)



b)

Fig. 7. Reverberation properties of classroom: a)) sound source voice of lecturer; b) loudspeakers system 7.1. Where 1 – reverberation diagram, 2 – integral of reverberation diagram

The reverberation diagram (energy density function in time) is calculated as the room response to the Dirac pulse. The integral of reverberation diagram shows the total energy density at the listening place as function of reverberation time.

The results of the modeling prove, that when the complex acoustic system is installed in auditorium, the total sound field level settles almost three times faster compare to lecturer voice without additional equipment.

Room reverberation parameters are also related to the nature of the sound source. Reverberation diagram and SNR significantly change when lecturer voice is changed (SNR = 19.6 dB) into the sound of acoustic system (SNR = 16.9 dB), as shown in Fig. 7 a, b. Early decay time increases from 0.209 s to 0.216 s. The calculated early decay time represents an average value over the whole frequency range.

At the same time no significant differences between loudspeakers systems type 5.1 and 7.1 according this point of view were not determined.

The results showed that the loudspeaker mounting and power can be optimized for the best intelligibility in many of the listening positions [10].

Speech intelligibility modeling results when calculating articulation loss of consonants *Al* is shown in Fig. 8. The results indicate that when lecturer speaks in the furthest corner of the 6x6 m auditorium, the intelligibility of consonants decreases almost by 30 %.

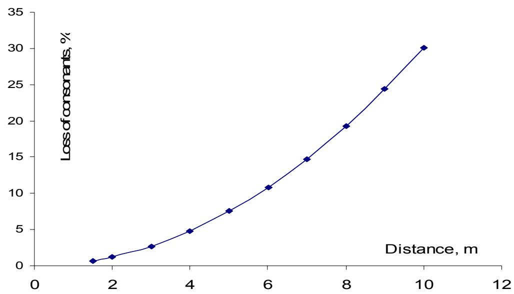


Fig. 8. The articulation loss of consonants

Conclusions

1. When performing the renovations of auditoriums, it is necessary to assess the acoustic properties of wall coatings and typically used gypsum panels.

2. In order to achieve better speech intelligibility, it is purposeful to install more complex acoustic systems in auditoriums.

3. In a small auditorium the lecturer voice as a sound source is sufficient in order to achieve good speech intelligibility and signal to noise ratio (SNR > 15 dB), provided that the auditorium is sufficiently good isolated from outer noise sources.

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A. Dumčius. Simulation of Sound Field in a Classroom // Electronics and Electrical Engineering. – Kaunas: Technologija, 2007. – No. 5(77). – P. 73–76.

Peculiarities of sound field in a classroom simulation are considered. The simulation of reverberation time, sound pressure level as function of distance and frequency, signal to noise ratio, location, speech intelligibility and articulation loss of consonants as function of distance from sound source are given. The simulation results of one speaker and two loudspeakers systems in frequency and spatial areas are shown. Ill. 8, bibl. 10 (in English; summaries in English, Russian and Lithuanian).

А. Думчюс. Моделирование звукового поля в аудитории // Электроника и электротехника. – Каунас: Технология, 2007. – № 5(77). – С. 73–76.

Рассматриваются аспекты моделирования звукового поля в учебном помещении при трех вариантах источника звука: голос преподавателя, акустические системы типа 5.1 и 7.1. Представлены результаты моделирования времени реверберации, уровня звукового давления как функции расстояния и частоты, отношения сигнал шумы, уровня разборчивости речи и артикуляции согласных. Результаты моделирования и измерения времени реверберации хорошо согласуются в области средних частот. Ил. 8, библ.10 (на английском языке; рефераты на английском, русском и литовском яз.).

A. Dumčius. Garso lauko modeliavimas auditorijoje // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2007. – Nr. 5(77). – P. 73–76.

Nagrinėjama garso lauko modeliavimo auditorijoje aspektai. Pateikta reverberacijos laiko, garso slėgio lygio priklausomybės nuo atstumo ir dažnio, signalo santykio su triukšmu, taip pat lokalizavimo, kalbos ir priebalsių artikuliacijos suprantamumo lygio modeliavimo rezultatai. Reverberacijos laiko modeliavimo ir matavimo rezultatai gerai sutampa vidutinių dažnių ruože. Parodyti lektoriaus ir garsiakalbių dviejų sistemų modeliavimo dažnių ruože ir erdvėje rezultatai. Il. 8, bibl. 10 (anglų kalba; santraukos anglų, rusų ir lietuvių k.).

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