Cepstral Deconvolution Method for Measurement of Absorption and Scattering Coefficients of Materials

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A method based on cepstral deconvolution technique is proposed for measurement of absorption and scattering coefficients of materials. The reverberation room method standardized by International Organization for Standardization (ISO) is taken as the reference for measurements. Several measurements are conducted in a physically scaled reverberation room and results are evaluated by these two methods. Two methods show slight differences in the estimation of specular parts of room impulse responses essential for determination of scattering coefficients. In the standard method, the specular part is found by synchronous averaging of impulse responses. However, the cepstral deconvolution method utilizes cepstral analysis to obtain the specular part instead of averaging. Results obtained by these two approaches are compared for different test materials. Both methods have yielded almost the same values for absorption coefficients. On the other hand, lower scattering coefficient values have been obtained for the cepstral deconvolution method.

1 INTRODUCTION

Acoustical properties like absorption and scattering are very crucial especially in the acoustical design of spaces which characterize acoustical comfort for a specific functional use. Accuracy in measurement of absorption and scattering coefficients is of vital importance to obtain reliable predictions in the area of room acoustics.

In 1965, Schroeder[1] introduced backward integration method to measure reverberation times in enclosures. The enclosure is excited by an impulsive noise and reverberation time is determined from corresponding decay curve obtained from integration of squared room impulse responses in reverse direction in time domain. Therefore, need for repeating measurements many times and ensemble averaging of decay curves to eliminate unwanted random fluctuations is no

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longer required. In reverberation time measurements utilizing integrated impulse response method, an excitation signal having flat spectrum should be employed. Schroeder[2] proposed to use maximum-length sequences (MLS) in measurement of impulse responses of linear systems. Chu[3] demonstrated advantages of using a periodic pseudorandom sequence i.e. MLS, in noisy environments together with experimental results. MLS was also employed in measurement of scattering on hard wall surfaces with highly diffuse behavior[4].

Vorländer and Mommertz[5] presented a new measurement method for determination of random-incidence scattering coefficients in 1995. The method was based on the separation of highly coherent specular part of the reflection from the total reflection by synchronous averaging of sufficient number of impulse responses obtained for different orientations of the sample surface. After this averaging procedure, it could be possible to evaluate the random-incidence scattering coefficient from the remaining part of the impulse response because averaging cancels out statistically independent scattered parts of the reflection. In 2004, this method was standardized by ISO for reverberation room measurements under standard number of ISO 17497-1[6]. This method was used in a development study of some scattering surfaces for concert halls by Jeon *et al.*[7] who employed the method in a scaled reverberation room for the evaluation of the effects of the size, and the structural density of different geometries located over a surface on the scattering properties of the surface.

In addition to measurement methods disregarding directional distribution of sound reflection, a method using directivity of reflected sound waves in determination of scattering coefficient was introduced by Mommertz[8].

An additional quantity to evaluate acoustical properties of surfaces is known as diffusion coefficient. It is used in evaluation of scattering uniformity of a surface for incident sound waves. A free-field measurement method has been proposed by Audio Engineering Society (AES) to evaluate both directional and random-incidence diffusion coefficients of surfaces[9]. Moreover, another free-field measurement method for absorption and scattering coefficients, namely, wave field synthesis method has been presented by Farina[10]. In this method, receiver (microphone) is moved on a straight line and specimen being tested is kept steady. Results obtained for this method and the method by Mommertz and Vorländer for scattering coefficient and AES method for diffusion coefficient were compared[10].

In this study, a method is developed to measure random-incidence absorption and scattering coefficients of materials in a scaled reverberation room[11]. Measurements for different materials are conducted and measurement results are evaluated for two different methods; the standard method proposed by ISO and the cepstral deconvolution method. These methods are compared in terms of results obtained for absorption and scattering coefficients. The new method differs from the standard method in decomposition of impulse responses in terms of their specular and scattered parts. Specular parts are separated from the rest of the impulse responses by employing cepstral analysis instead of time averaging of impulse responses. By this way, it is anticipated that specular parts could be obtained more accurately i.e. immune to noise caused by scattered parts of impulse responses.

2 CEPSTRAL DECONVULUTION

For an arbitrary composite signal $x(t)$ its cepstrum function describes the spectrum of a logarithmic spectrum of this signal. This function is very powerful because, if this signal is composed of a basic wavelet (such as an excitation signal) and a periodic component (e.g. its echoes), logarithm of the spectrum of this signal shows the delayed echoes as ripples. Then, the spectrum of this logarithmic spectrum gives the frequency of these ripples in the form of peaks at echo delay times[12]. Cepstrum analysis uses the same principles of superposition in algebraic operations. For example, the deconvolution procedure of a combined signal (superimposed in terms of convolution of the input signal and its echoes) is done by just addition and subtraction operations in cepstral domain.

Cepstrum techniques are widely in use for echo detection and signal decomposition. These two issues form the basic structure of a deconvolution process. Considering a signal which is composed of multiple signals overlapped in time domain, deconvolution process starts with determination of the echoes in the signal in terms of their relative amplitudes and epochs (arrival or delay times) by employing real cepstrum. Then, complex cepstrum is calculated for this signal and undesired parts of the signal are removed via filtering in cepstral domain. Finally, reconstruction of the remaining signal is done by transformation back to the time domain.

Real cepstrum of an arbitrary function $x(t)$ can be described as the inverse Fourier transform of the logarithm of the magnitude of the Fourier transform of this function. Real cepstrum of the function can be obtained by

$$c_{r,x}(\tau) = F^{-1}\left\{\log\left(\left|F\left\{x(t)\right\}\right|\right)\right\} = F^{-1}\left\{\log\left|X(\omega)\right|\right\}. \tag{1}$$

Delay times and amplitudes of echoes are determined by using real cepstrum easily. However, because all of the phase information of the signals is lost during the calculation, it is impossible to reconstruct the separated signals through real cepstrum. Complex cepstrum of an arbitrary function $x(t)$ may be defined as the inverse Fourier transform of the logarithm of the Fourier transform of this function.

$$c_c(\tau) = F^{-1}\left\{\log\left[F\left\{x(t)\right\}\right]\right\} = F^{-1}\left\{\log\left[X(\omega)\right]\right\}. \tag{2}$$

Complex cepstrum is employed for separation of the components of the composite waveforms, especially combined via convolution. Different from real cepstrum, phase information of the signals is kept in the complex cepstrum. So, it can be utilized in the reconstruction of the separated parts

Cepstral deconvolution for a composite signal $x(t)$ is performed in three steps, namely,

i) Estimation of the echo delay times in cepstral domain by employing real cepstrum of the signal,
ii) Evaluation of the complex cepstrum of the signal and removal of the unwanted parts in the cepstral domain via linear filtering,
iii) Restoration of the desired part of the signal in time domain by reversing the procedure followed in the calculation of the complex cepstrum.

Components forming an impulse response can be classified into two groups: scattered part and specular part. Specular parts of impulse responses for a particular measurement are highly correlated though their scattered parts are not correlated and even completely independent from each other. It is anticipated that such correlation properties of impulse responses can be used to separate the scattered parts from the rest of the impulse responses via cepstral deconvolution.

Waveforms of the specular parts of impulse responses remain unchanged throughout a particular measurement i.e. independent of orientation of the specimen being tested. Then, real cepstrum of a series of impulse responses gives specular part of every impulse response in a form of peaks in cepstral domain. The very first impulse response corresponds to basic signal and

specular parts of the rest of the impulse responses correspond to echoes of this basic signal. These specular components can be determined in cepstral domain by indicating evenly separated peaks because every impulse response has the same length and synchronous with each other.
In the application of the approach, impulse responses are added successively. Average of the corresponding series is used as the very first element of this sequence. This is because; average of a series includes all necessary information about the specular parts of impulse responses and also carries less distortion caused by the scattered parts. After applying real cepstrum to this sequence every impulse response after the first one cause a peak in the cepstral domain. Then deconvolution is achieved by liftering (i.e. filtering in cepstral domain) these peaks employing a linear notch filter. Liftered sequence is transformed into the time domain and this final signal gives specular part free impulse response sequence except the average one located at the beginning of the sequence. This deconvolved signal is subtracted from the original one and finally specular parts of the impulse responses are obtained. In the rest of the calculations, the average of these specular parts of the impulse responses is used in order to increase the signal-to-noise ratio and the accuracy of the measurement.

3 EXPERIMENTAL PROCEDURE

3.1 Experimental Setup

In order to conduct measurements a physically scaled reverberation room with a scale ratio of 1:10 was built (Fig. 1) according to requirements given in the standards ISO 17497-1 and ISO 354. Walls of the reverberation room are made of Plexiglas (20 mm in thickness) with internal dimensions of 0.72 m x 0.60 m x 0.465 m. The base plate of the turntable is produced using MDF with a diameter of 0.38 m. Rotational movement of the base plate is driven by a 12 V DC motor. An average value for reverberation time obtained from broadband signals can be used to make a rough estimation about the Schroeder frequency. With the values of $T = 0.6\,\mathrm{s}$ and $V = 0.201\,\mathrm{m}^3$, the limiting frequency becomes 3455 Hz in the reverberation room which corresponds to 345.5 Hz in full size measurement scheme.

A broadband noise signal generated from a maximum-length sequence (MLS) of order 16 is used as the excitation signal. The angular speed of the turntable and length of the MLS were taken into account to generate the signal by successive addition of 1+120 basic sequence (1st sequence will be dropped in the evaluation of the output signal due to the need for reaching a steady sound pressure level initially). This final sequence is sampled with frequency of 48 kHz and recorded as a sound wave. Recorded test signal is then sent to the loudspeakers by a 16 bit soundcard and an integrated amplifier. The signal involving the impulse response of the reverberation room is acquired by a ¼″ pressure microphone. Finally, this signal is recorded by the same soundcard after it is amplified through a preamplifier with a microphone power supply.

3.2 Test Specimens

In this study, different specimens of square shape with different surface pattern have been used as test specimen in the reverberation room[11]. These specimens are commercial acoustical panels produced by a local company for different acoustic treatment purposes. Their absorption performances are specified by the producer through measurements with respect to ISO 354. As the reverberation room used in the tests is a scaled reverberation room, the results obtained from the measurements correspond to the response of a material with larger dimensions than the measured one by a factor of scale ratio. These materials are shown in Fig. 2 with their commercial names.

3.3 Measurement Procedure

The method described in ISO 17497-1 is based on the assumption that scattered components of impulse responses for different positions of a material are statistically independent from each other[5,6]. According to this assumption, specular part of the energy reflected form the surface of a material can be extracted from the impulse response by using "synchronized averaging" of impulse responses obtained during a period of a measurement for different positions of the material[6]. By this averaging procedure scattered components of the impulse responses cancel out each other and the remaining part is thought to be just consisted of "highly correlated" specular components of the reflection. In practice, a test sample is placed on a turntable in a reverberation room and impulse responses are collected while the sample is rotated during the measurement. Then, the averaged impulse response is used in further calculations. If the sample shows a perfect specular reflection characteristic and also the sample is circular in shape, results of measurements for which turntable is rotating and not rotating show an excellent correlation. Deviations from this perfect correlation are directly related to the diffuse reflections (i.e. scattering) exhibited by the sample being tested[5].

The measurement procedure adopted in this study follows the requirements of ISO 354 and ISO 17497-1[6, 13]. First of all, reverberation times T_1 and T_2 "without and with test sample", respectively are obtained from the measured impulse responses while the turntable is not rotating. Then, this is repeated for a rotating turntable to get reverberation time T_3 without the test sample and reverberation time T_4 with the test sample. During the measurements, while the turntable is operating, excitation signals should be propagated in equal time steps for a measurement period. This means that the total time required for n number of periodic test signals to be radiated should be equal to the time elapsed in one revolution of the turntable at constant speed. The test signal should be the same for all measurements during the tests. This is required to get a time invariant impulse response obtained from the averaged n measurements. Averaging could be performed before or after calculating the impulse responses. The number of impulse responses to be averaged should be within the range from 60 to 120 [6]. This number is also equal to the number of excitation signals radiated during a measurement period.

The decay curve of the sound pressure level from the averaged impulse responses is obtained as stated in ISO 354. In this procedure Schroeder's backward integration technique is applied to the impulse response. The result is a linearly decreasing curve for T_1, T_2 and T_3 measurements and "two superimposed decay curves" for T_4 measurements where the reverberation time should be evaluated from the first one[6].

In the determination of the reverberation times from decay curves, the upper limit of the evaluation range should be 5 dB below the highest sound pressure level of the impulse response and the lower limit should be 10 dB above the background noise level. In order to obtain the reverberation time a least squares fit line algorithm is performed on the evaluation range and the resultant fit line is used to determine the corresponding reverberation time[13].

Measurement procedure should be repeated for at least six times. This is combination of at least two different sound source positions and at least three different microphone positions[6]. Then, evaluated reverberation times for the same sample and turntable configuration (e.g. six T_2 for different positions) are averaged arithmetically in spatial manner and this averaging is performed for each four different reverberation times to obtain their final value before further calculations.

4 RESULTS AND DISCUSSION

In measurements, a total of six combinations of two loudspeaker and three microphone positions are used for testing each specimen (see Fig. 3). Every specimen is tested for each loudspeaker/microphone position combination for four different reverberation times (T_1, T_2, T_3, T_4), i.e. 24 measurements are carried for each material. During the measurements, test codes given in relevant standards are followed. Random-incidence absorption and scattering coefficients of the specimens obtained according to both ISO standard method (S) and the cepstral deconvolution method (CD) are illustrated in Fig. 4 and Fig. 5, respectively.

Both methods gave almost the same results for random-incidence absorption coefficients for all of the specimens. This is reasonable, because, in the calculation of absorption coefficients only T_1 and T_2 values affect the resultant values. As a matter of fact, first three reverberation times have very similar characteristics. In measurements of these T values, impulse responses have not changed significantly throughout a measurement set. Therefore, these impulse responses happen to be almost identical with their average. This causes separation of the whole signal instead of its specular part in cepstral deconvolution. In conclusion, almost the same impulse response is used in the rest of evaluations and calculations.

On the contrary, results of random-incidence scattering coefficients have exhibited strong deviations from each other for two methods especially at higher frequencies. This is due to difference in T_4 values obtained for two methods. As shown in the Fig. 6 for longer evaluation ranges on decay curves the difference has become much more noticeable. The interpretation of this outcome is that the sound pressure level difference between the root-mean-square amplitude of signal and background noise. Hence, in turn signal-to-noise ratio is reduced due to the subtraction of specular part from the original impulse response performed in cepstral deconvolution method.

5 CONCLUSION

Cepstral deconvolution method yields very close results for absorption coefficients with differences less than 1 % in general and 3.49 % at most for reliable evaluation ranges when compared against the results obtained from ISO method. However, the proposed method produces scattering coefficients much lower than those measured by ISO method. In that respect these results of scattering coefficients are thought to serve as lower bounds for their corresponding values.

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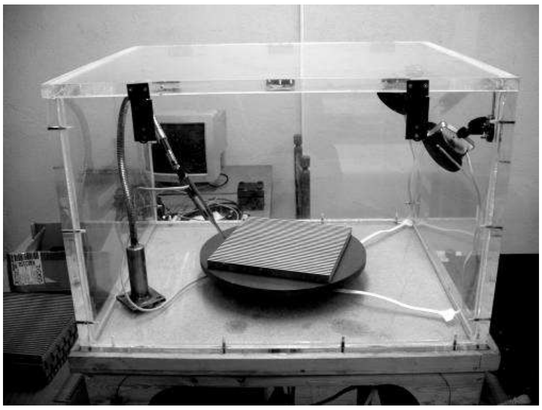
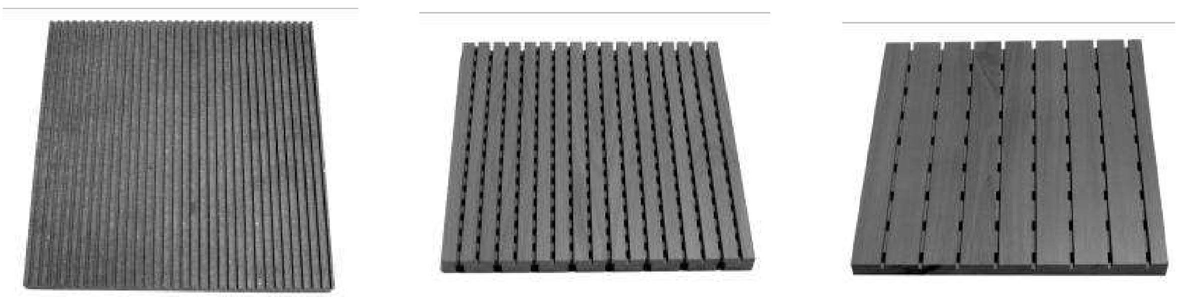


Fig. 1 – The reverberation room and experimental setup[11]



(a) **(b)** **(c)**

Fig. 2 – Test specimens ***(a)*** *No 1 - Reflector Panel YST-KNL/3x5 Lake,* ***(b)*** *No 2 - Mid-Frequency Panel KP4000DS****, (c)*** *No 3 - Mid-Frequency Panel LM1000DS*

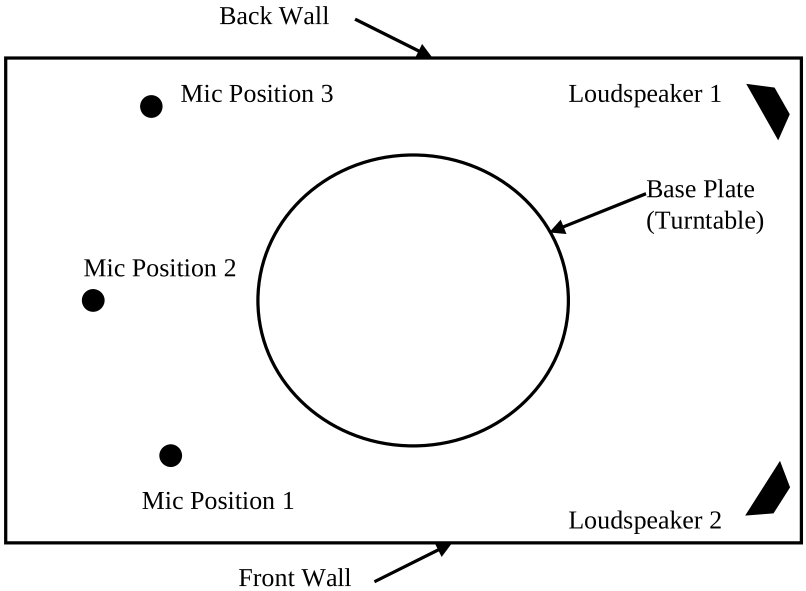
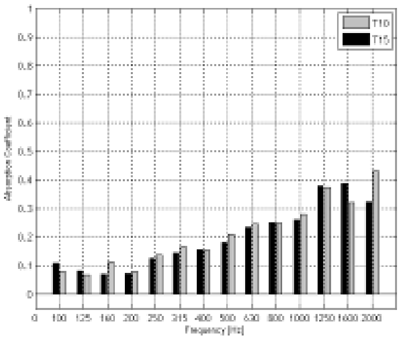
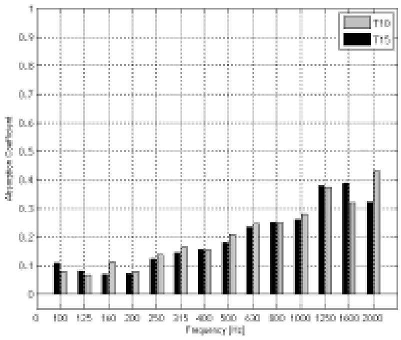


Fig. 3 – Measurement configurations[11]



(a)



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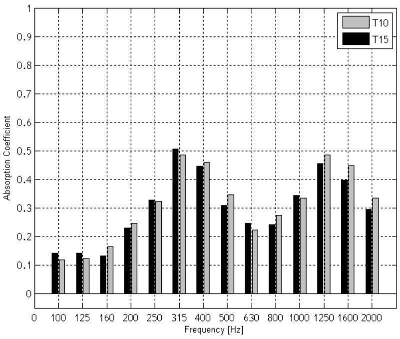

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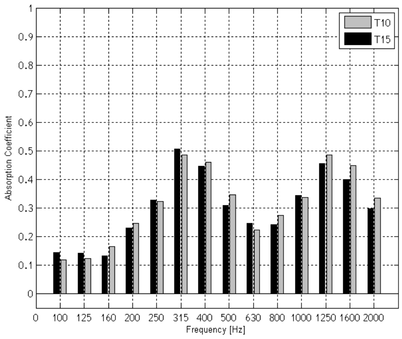
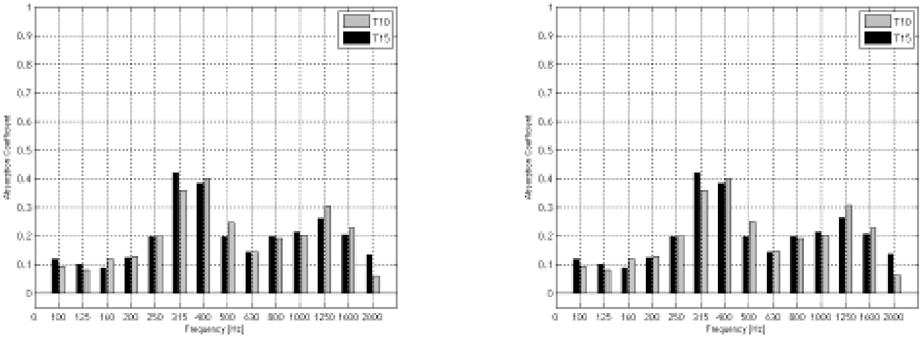
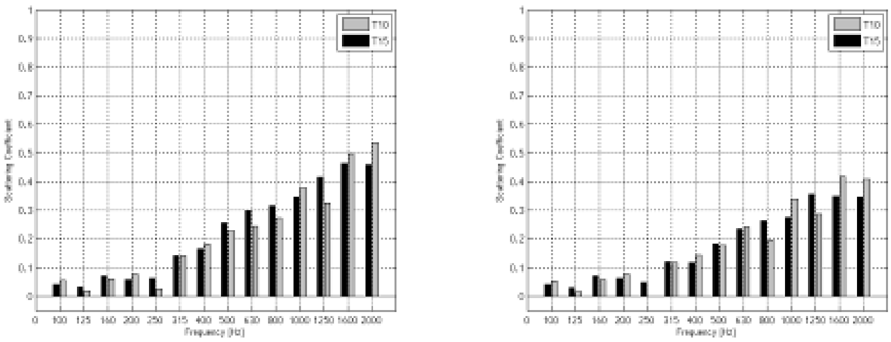


(d)

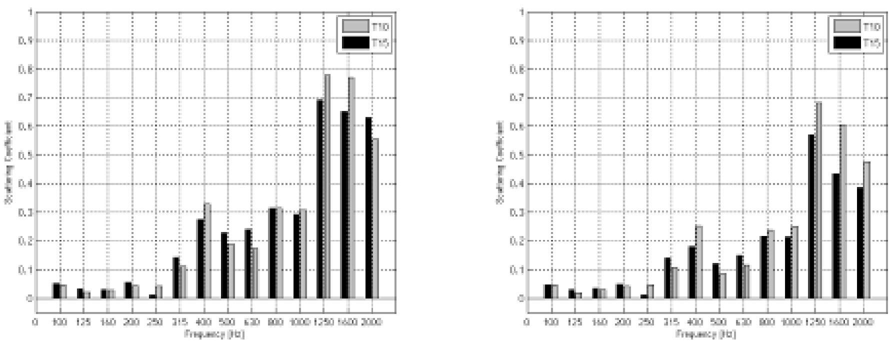


(e) **(f)**

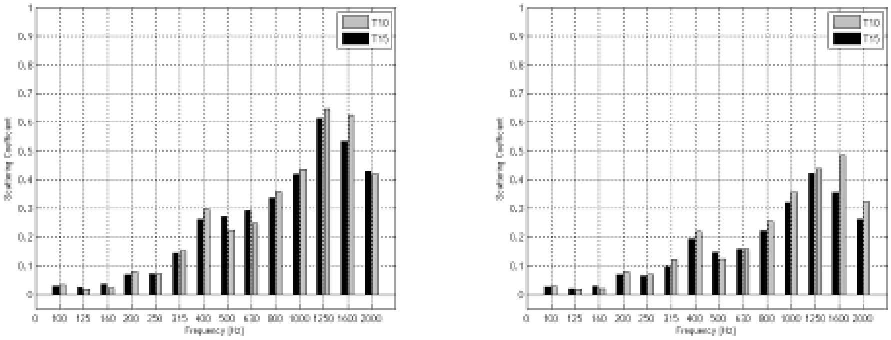
Fig. 4 – Random-incidence absorption coefficients obtained by using S (left column) and CD (right column) of specimens No-1 (a-b), No-2 (c-d), No-3 (e-f)11



(a) **(b)**



(c) **(d)**



(e) **(f)**

Fig. 5 – Random-incidence scattering coefficients obtained by using S (left column) and CD (right column) of specimens No-1 (a-b), No-2 (c-d), No-3 (e-f)11

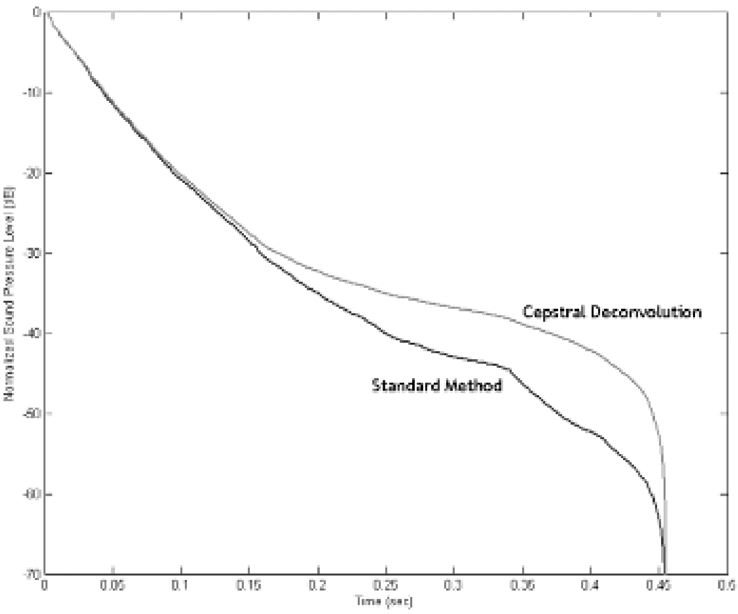


Fig. 6. − Decay curves for S and CD[11]