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A METHOD FOR DETERMINING THE EQUIVALENT LEVEL L_{eq}

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The constantly growing traffic noise caused, among others, by the rapid development of motorization, is a considerable menace to the human environment. From the view point of acoustics the rating scale generally used to evaluate the environment is the equivalent level L_{eq}. In this paper the relationships between such parameters as the traffic flow, the speed and sound power of individual vehicles, and parameters describing the paths of the moving noise sources are derived.

1. Introduction

The equivalent sound level L_{eq} is one of the many rating schemes of traffic noise now used. The investigations carried out in Sweden [27], France [2], the Federal Republic of Germany [20], Australia [11] and in Poland [23] have shown that L_{eq} is a valid rating scales for the evaluation of noise. It is also widely used in the U.S.A. [3]. It serves, among others, for the determination of the "day-night" equivalent level [5, 23]. The results of the investigations performed in Great Britain [4, 24] indicate that the Traffic Noise Index – TNI [7] and the Perceived Noise Level – PNL [21] are both superior to L_{eq} as noise evaluation indices, since they are a better "measure" of the fluctuation of noise level. These fluctuations have a decisive effect on the noise exposure. Because of measurement difficulties the TNI and PNL are not so widely used as the equivalent – L_{eq}, which can be measured directly using meters such as those produced by Brüel-Kjaer and RFT.

The definition of L_{eq} is

$$L_{eq} = 10\log\frac{1}{T}\int\limits_{-T/2}^{T/2} 10^{0.1\,L(t)}\,dt, \tag{1}$$

where $L(t)$ denotes the sound intensity level measured in dB(A) and T the averaging time.

According to the definition of sound intensity level, to each value of the sound intensity $L(t)$ measured in dB(A) can be assigned the corresponding sound intensity

$$I(t) = I_0 \cdot 10^{0.1}L(t),$$

where I_0 denotes the reference intensity.

Substituting this relation into the definition of the equivalent level (1) we obtain

$$L_{eq} = 10 \log \frac{\langle I \rangle}{I_0}, \tag{2}$$

where

$$\langle I \rangle = \frac{1}{T} \int_{-T/2}^{T/2} I(t)\, dt$$

is the average sound intensity.

The purpose of this paper is to find the relationship between the equivalent level L_{eq} and such parameters as the traffic flow, the speed, those describing the route of moving noise sources, and sound power of the individual source types.

2. Method for the calculation of L_{eq} in urban areas

Among others, Kuttruft, Lyon, Lindquist, Malchaire, Shaw, Olson and Thiesse have been engaged in the problem of calculating some of the quantities characterizing urban noise. In this paper it is assumed that the parameter characterizing the urban noise, the rating scale of this noise, is the equivalent level L_{eq}. The proposed method for predicting the magnitude of L_{eq}

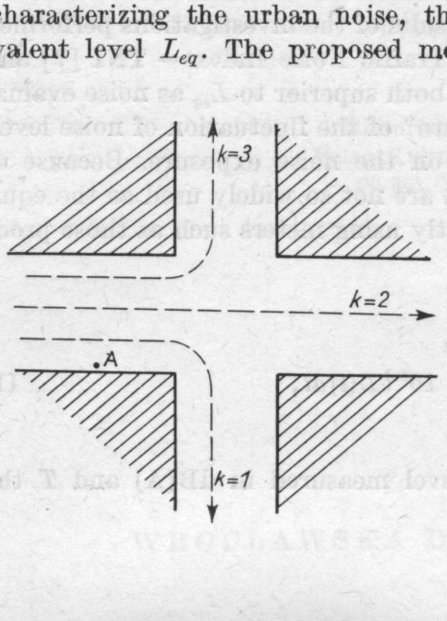

Fig. 1. Vehicle routes near the crossroad

differs from previous methods in that it only requires the measurement of $L_i(t)$, the intensity level of the sound emitted by a single passing vehicle.

Let us assume that we want to calculate the equivalent level L_{eq} at a point A located as shown in Fig. 1 which represents a partial view of central building development.

For further calculations let us form several classes of vehicles that differ by their sound intensity $I(t)$. We take into consideration:

(a) The power of the noise emitted while travelling at a constant speed determined by standard recommendations.

The vehicles considered: lorries, passenger cars, tramway, motorcycles etc. will have different values of this parameter which in the following text will be marked by the letter j.

(b) Travelling route.

In Fig. 1 several possible routes are marked, which will be given the symbol k.

(c) Speed.

We shall assume that there exist "discrete realizations" of the travel $V_l(t)$, e.g. travel at a constant speed, travel at a speed $V_1(t)$, where the function $V_1(t)$ is strictly defined, etc. This will be denoted by the subscript l.

In this manner the class jkl contains the vehicles of type j, which are moving at speed l ($V_l(t)$), along route k. Each of the possible combinations j, k, l will be denoted by the letter i. In this manner the class i ($i = 1, 2, \ldots, n$) comprises the vehicles that are characterized at a point A by the same sound intensity as a function of time. The moment at which vehicle m belonging to class i passes the point of observation is a stochastic quantity. Thus the intensity of sound coming from this source is a function of the form

$$I_{m,i} = I_{m,i}(t - \gamma_{m,i}),$$

where $\gamma_{m,i}$ is the moment at which the source noise is passing the point of observation. The maximum value of sound intensity is then recorded (Fig. 2). We assume that γ is a stochastic quantity with a probability density function $p(\gamma)$.

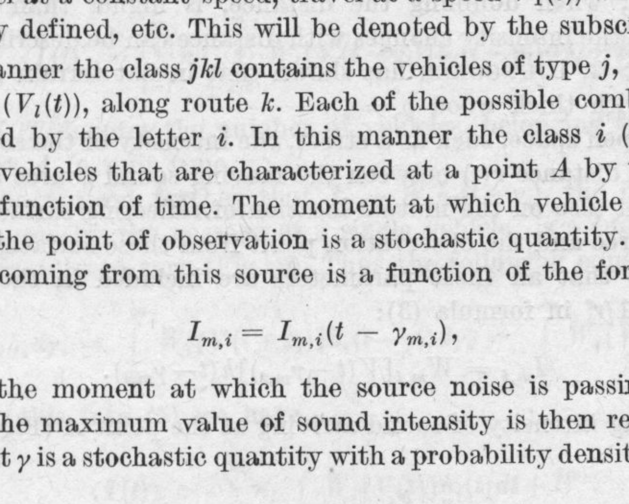

Fig. 2. Changes of the intensity of noise caused by i-class vehicle — $I_i(t)$

The vehicle, travelling at a speed $V(t)$ in open space, is a source of sound intensity

$$I_{m,i} = \frac{W_{m,i}[V(t-\gamma_{m,i})]}{[r(t-\gamma_{m,i})]^{\varrho}}, \tag{3}$$

where r denote the distance of the vehicle from the point in relation to which the computations are made. Movement of the source implies that this is a quantity which changes with time, while the sound power depends on the speed V.

In order to explain more closely the physical meaning of the exponent ϱ let us notice that, under the assumption of a loss-free medium, a sound intensity at the distance r from a spherical sound source is (in an open space) given by the formula

$$I = \frac{W}{r^2}.$$

Because of the absorption of the energy in air, and also the absorption observed at the reflection from the earth's surface, the level decrease of sound intensity, when doubling the distance, is higher than 6 dB ($\varrho = 2$). This means that the intensity changes with distance can be described by formula (3) assuming $\varrho > 2$. LJUNGGREN has shown that in flat terrain $\varrho = 3$ (for terrains covered with thick wood $\varrho = 4$).

In a semi-open space, such as a street, the intensity of the sound I depends not only on the distance $r(t)$ between the moving sound source and the observation point, but also on the mutual location of reflecting surfaces, on the distance of the source and the observation point from these surface, etc.

We assume that all these parameters are included in the function $h(t)$ which replaces $1/r^{\varrho}$ in formula (3):

$$I_{m,i} = W_{m,i}[V(t-\gamma_{m,i})]h(t-\gamma_{m,i}). \tag{3a}$$

The resulting intensity of sound arriving at the point A (Fig. 1) is the sum of these forms,

$$I(t) = \sum_{i=1}^{n} \sum_{m=1}^{N_i} W_{m,i} h(t-\gamma_{m,i}) + I^{(0)}(t), \tag{4}$$

where $I^{(0)}$ denotes the intensity of the background noise from other streets, while N_i is the number of vehicles of class i passing close to the point A in a time interval T.

The sound power $W_{m,i}$, like the $\gamma_{m,i}$, is a random quantity. Writing the probability density for the travel of the noise source of power W_i as $p(W_i)$, OLSON states that $p(W_i)$ is a long normal distribution for a certain type of vehicle travelling at a constant speed.

Since $I(t)$, according to (4), is dependent upon the random variables γ_i and W_i, the mean intensity

$$\langle I(t)\rangle = \int_{-\infty}^{\infty}\int_{-T/2}^{T/2} I(t, W_i, \gamma_i)p(W_i)p(\gamma_i)\,dW_i\,d\gamma_i + \bar{I}^{(0)},$$

where $\bar{I}^{(0)}$ denotes the average intensity of the background noise.

Consequently, from (4), we have

$$\langle I(t)\rangle = \sum_{i=1}^{n}\sum_{m=1}^{N_i}\int_{-T/2}^{T/2}\Big[\int_{-\infty}^{\infty} W_i(V)p[W_i(V)]\,dW_i\Big]h_i(t-\gamma_i)p(\gamma_i)\,d\gamma_i + \bar{I}^{(0)}$$

and

$$\langle I(t)\rangle = \sum_{i=1}^{n}\sum_{m=1}^{N_i}\int_{-T/2}^{T/2}\overline{W}_i[V(t-\gamma_i)]h_i(t-\gamma_i)p(\gamma_i)\,d\gamma_i + \bar{I}^{(0)},$$

where $\overline{W}_i$ denotes the mean sound power of a vehicle belonging to the class which is moving at a speed V.

If we assume that the passing of any vehicle is random in a time T, then $p(\gamma_i) = 1/T$, and we obtain

$$\langle I(t)\rangle = \sum_{i=1}^{n} n_i \int_{-T/2}^{T/2}\overline{W}_i[V(t-\gamma_i)]h_i(t-\gamma_i)\,d\gamma_i + \bar{I}^{(0)}, \tag{5}$$

where $n_i = N_i/T$ denotes the number of vehicles, belonging to class i, passing near the point A in unit time.

It may be noticed further that $I_i = \overline{W}_i h_i(t-\gamma_i)$, the intensity changes of a sound caused by the passage of a single vehicle, is a function which decreases very rapidly to zero (Fig. 2). Thus the following equalities occur:

$$\int_{-T/2}^{T/2}\overline{W}_i h_i\,d\gamma_i = \int_{-\infty}^{\infty}\overline{W}_i[V(t-\gamma_i)]h_i(t-\gamma_i)\,d\gamma_i = \int_{-\infty}^{\infty}\overline{W}_i[V(t)]h_i(t)\,dt.$$

Inserting this into (5) we have

$$\langle I(t)\rangle = \sum_{i=1}^{n} n_i \int_{-\infty}^{\infty}\overline{W}_i[V(t)]h_i(t)\,dt + \bar{I}^{(0)}. \tag{6}$$

In this manner the equivalent level L_{eq} is a function of the form

$$L_{eq} = 10\log\frac{1}{I_0}\Big[\sum_{i=1}^{n} n_i a_i + \bar{I}^{(0)}\Big], \tag{7}$$

where

$$a_i = \int_{-\infty}^{\infty} I_i(t)\,dt = \int_{-\infty}^{\infty}\overline{W}_i[V(t)]h_i(t)\,dt, \tag{8}$$

in which $I_i(t)$ denotes the intensity change of a sound (with time) during the passage of a single vehicle belonging to class i (Fig. 2), $\bar{I}^{(0)}$ is the average intensity of the background, while n_i is the number of vehicles of class i passing near the point A in unit time (e.g. in hours). In other words, this is the traffic flow of vehicles of class i.

The average intensity of the background can be measured or calculated with the aid of the so-called *homogeneous model* for an ideal city [26, 28].

In many cases in the proximity of motorways or in high traffic flow, the magnitude $\bar{I}^{(0)}$ can be neglected. Formula (7) then permits prediction of the value of L_{eq} at any point of observation, since we know:

(a) the signal $L_i(t)$ at this point, caused by the movement of a single source of class i (Fig. 2) and

(b) the number of sources n_i of class i passing the observation point in unit time.

The already existing urban situations causes that the measurement of $L_i(t)$ can only be done at night when the sum amount of traffic permits the recording of a signal coming (without doubt) from a single source of a determined class. Model investigations are thus much more convenient.

If the point A (Fig. 1) is located in a one-way street and there are three possible travel routes along which lorries or passenger cars can move at equal speeds, then we can record six various signals coming from six different sources, as shown in Fig. 2. According to the previous definition of the source class, we have in this case $k = 1, 2, 3$ – three different travelling times, $j = 1, 2$ – two types of vehicles, $l = 1$ – the travelling speed – is the same for all vehicles, and it can be seen that the number of sources $n = 6$.

An advantage of the method proposed, which is expressed by relation (7), is the possibility of determining the equivalent level L_{eq} (for an acoustic field generated by any number of moving sources) on the basis of the sound intensity level $L_i(t)$ (Fig. 2) of a single source and traffic flows – n_i.

3. Method for the calculating L_{eq} close to the motorway

The problem of propagation of the noise, generated by vehicles moving along a motorway, is much simpler than that connected with the noise propagation in urban situation, for the simple reason that the only reflecting plane is the earth's surface. This permits of a theoretical determination of the sound intensity $I_i(t)$, and thus the magnitude a_i in formula (7). In this case it is necessary to determine experimentally the average sound powers of vehicles of various types $\overline{W}_j$, and also the magnitude of ϱ (formula (3)), which describes the sound attenuation in open spaces. For many types of vehicles the magnitude $\overline{W}_j$ and values of the exponent ϱ can be found in the literature.

The problem of calculation of L_{eq} and of the mean sound intensity $\langle I \rangle$ in the proximity of motorways has engaged many authors, including, ANDERSON, GORDON, JOHNSON and SAUNDERS, KURZE, LJUNGGREN, MARCUS, RATHE, SCHREIBER. The results of this paper are in some cases more general than the results of these authors.

Let the point A, at which L_{eq} is calculated, be located as shown in Fig. 3. The presence of woods or of compact building developments, lining an area, determines the fact that during the passage of a vehicle belonging to class i, the noise reaches the point A only in a time ϑ_i. Formally, it means that $\overline{W}_i = 0$ outside this time interval. If the point A is in entirely free space, then one should accept $\vartheta_i = \infty$.

The motorway is a road on which the traffic moves at constant speeds, except for such cases as, for instance, a change of traffic lane. This permits of the rewriting of expression (8) in the form

$$a_i = \overline{W}_i[V] \int_{-1/2\vartheta_i}^{1/2\vartheta_i} h_i(t)\,dt,$$

since the sound power $W_i[V]$, being a function of speed, is thus independent of time.

If we also assume that we have n_1 traffic lanes and n_2 types of vehicles: lorries, passenger cars etc., then the number of vehicle classes is given by $n = n_1 \times n_2$, since a vehicle of any type can move along any traffic lane, maintaining only the speed and travelling direction. Taking this all into consideration we obtain from (6)

$$\langle I(t)\rangle = \sum_{i=1}^{n_1 n_2} n_i \overline{W}_i[V] \int_{-1/2\vartheta_i}^{1/2\vartheta_i} h_i(\tau)\,d\tau + \bar{I}^{(0)}.$$

If we also denote by n_{jk} the traffic flow of vehicles of type j on traffic lane k, and by $W_j(V_k)$ the sound power of the vehicle of type j moving at a speed V_k (thus moving along the traffic lane k), then

$$\langle I(t)\rangle = \sum_{k=1}^{n_1} \int_{-1/2\vartheta_k}^{1/2\vartheta_k} h_k(\tau)\,d\tau \sum_{j=1}^{n_2} n_{jk} \overline{W}_j(V_k) + \bar{I}^{(0)}. \tag{9}$$

The integral $\int_{-1/2\vartheta_k}^{1/2\vartheta_k} h_k(\tau)\,d\tau$ will be calculated for two cases:

(a) the smallest distance of the point A from the axis of the motorway is considerably higher than the width of the motorway: $D \ll d$ (Fig. 3). It is then possible to assume that the travel routes of all vehicles are the same ($k = 1$) and are described by the curve $y(x)$;

(b) the smallest distance of the point A from the motorway is comparable with its width ($d \approx D$), for example when the point A is located on the shoulder of the motorway. In this case it can be assumed that the length of the motorway on which the vehicles, that contribute essentially to the resultant noise intensity, are moving is a straight line.

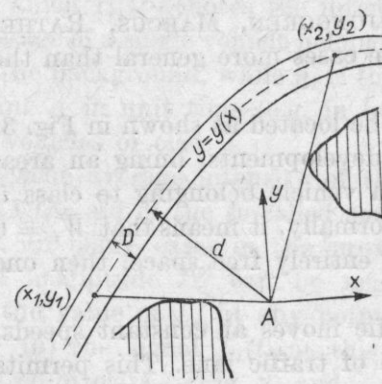

Fig. 3. The highway course $- y(x)$ for the outlying point A.

In this case, in contradistinction to case (a), it is necessary to distinguish the traffic lanes.

By comparing relations (3) and (3a) we get

$$h_k(t) = \frac{1}{[r(t)]^{\varrho}}, \quad \varrho > 2.$$

If the vehicle is moving at a speed V_k, then by virtue of formula

$$ds = \sqrt{1+[y'(x)]^2}\, dx = V_k\, dt$$

and on the basis of Fig. 3, we obtain for case (a)

$$\int_{-1/2\vartheta_k}^{1/2\vartheta_k} h_k(t)\, dt = \frac{1}{V_k}\int \frac{ds}{[r(x,y)]^{\varrho}} = \frac{1}{V_k}\int_{x_1}^{x_2} \frac{\sqrt{1+[y'(x)]^2}}{[x^2+y^2(x)]^{1/2\varrho}}\, dx. \tag{10}$$

By substituting in (9), we obtain the formula for average sound intensity at point A located far from the motorway,

$$\langle I(t)\rangle = \int_{x_1}^{x_2} \frac{\sqrt{1+[y'(x)]^2}}{[x^2+y^2(x)]^{1/2\varrho}}\, dx \sum_{k=1}^{n_1}\sum_{j=1}^{n_2} n_{jk}\frac{\overline{W}_j(V_k)}{V_k} + \bar{I}^{(0)}, \tag{11}$$

and

$$L_{eq} = 10\log\frac{1}{I_0}\langle I(t)\rangle,$$

where $y(x)$ denotes a curve describing the course of the motorway, x_1, x_2 are the abscissae of points (Fig. 3) that limit the length of the motorway from which the noise reaches the point A, n_{jk} is the traffic flow of vehicles of type j moving along traffic lane k, $\overline{W}_j(V_k)$ — the sound power of a type j vehicle moving at a speed V_k, $\bar{I}^{(0)}$ — the average background intensity, n_1 — the number of traffic lanes, and n_2 — the number of vehicle types.

For case (b) it can be seen from Fig. 4 that equation (10) takes the following form:

$$\int_{-1/2\vartheta_k}^{1/2\vartheta_k} h_k(t)\,dt = \frac{1}{V_k}\int \frac{ds}{[r(x,y)]^{\varrho}} = \frac{1}{V_k}\int_{x_1}^{x_2} \frac{dx}{[x^2+d_k^2]^{1/2\varrho}}.$$

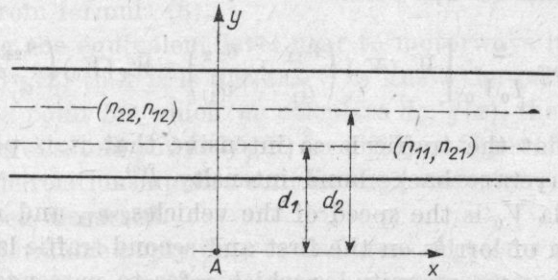

Fig. 4. Two-sided way with the given traffic intensity $- n_{jk}$

Since we assumed that the point A is located very near to the motorway we can accept that $x_{1/2} \to \pm\infty$. For $\varrho > 2$ (this condition is always satisfied) we obtain

$$\int_{-1/2\vartheta_k}^{1/2\vartheta_k} h_k(t)\,dt \int_{-\infty}^{\infty} \frac{dx}{[x^2+d_k^2]^{1/2\varrho}} = \sqrt{\pi}\,\frac{\Gamma(\frac{1}{2}\varrho-\frac{1}{2})}{\Gamma(\frac{1}{2}\varrho)}\,\frac{1}{d_k^{\varrho-1}},$$

where $\Gamma(x)$ denotes the Euler function.

Substituting the expression obtained into formula (9), we obtain for the average sound intensity at a point A near to the motorway the expressions

$$\langle I(t)\rangle = \sqrt{\pi}\,\frac{\Gamma(\frac{1}{2}\varrho-\frac{1}{2})}{\Gamma(\frac{1}{2}\varrho)} \sum_{k=1}^{n_1} \frac{1}{d_k^{\varrho-1}} \sum_{j=1}^{n_2} n_{jk} \frac{\overline{W}_j(V_k)}{V_k} + \bar{I}^{(0)} \qquad (12)$$

and

$$L_{eq} = 10\log\frac{1}{I_0}\langle I\rangle,$$

where d_k denotes the distance of the traffic lane k from the point A, and n_{jk} the traffic flow of vehicles of type j moving along traffic lane k.

The other quantities are the same as in formula (11) describing $\langle I(t)\rangle$ at a point located far from the motorway. The magnitude ϱ, which determines the magnitude of sound attenuation (3), occurs in formulae (11) and (12).

As an example, let us consider a two-lane motorway ($n_1 = 2$) along which two types of vehicles — lorries and passenger cars ($n_2 = 2$) — are moving at a constant speed V_0 in both directions. We wish to determine the equivalent level near to the motorway in case (b) (Fig. 4).

If atmospheric and terrain conditions are such that we can take $\varrho = 3$, then

$$\frac{\Gamma(\frac{1}{2}\varrho - \frac{1}{2})}{\Gamma(\frac{1}{2}\varrho)} = \frac{2}{\sqrt{\pi}}.$$

Substituting into formula (12), we obtain the following expression for the equivalent level near to this road:

$$L_{eq} = 10\log\frac{2}{I_0 V_0}\left[\overline{W}_1(V_0)\left(\frac{n_{11}}{d_1^2}+\frac{n_{12}}{d_2^2}\right)+\overline{W}_2(V_0)\left(\frac{n_{21}}{d_1^2}+\frac{n_{22}}{d_2^2}\right)\right].$$

We assume that the traffic is so intensive that it is possible to neglect the value of the average background intensity $\bar{I}^{(0)}$.

In this formula V_0 is the speed of the vehicles, n_{11} and n_{12} denote the intensity of a stream of lorries on the first and second traffic lanes, respectively, n_{21} and n_{22} are the same magnitudes which refer to passenger cars, d_1 and d_2 are the distances of the traffic lanes from point A, $\overline{W}_1$ and $\overline{W}_2$ are the sound powers of a typical lorry and passenger car, respectively, and I_0 is a reference intensity.

4. Conclusions

The method presented permits of the calculation of the equivalent level L_{eq} of noise generated by means transportation systems in urban situations (formula (7) and Fig. 1), and in open space: near to (formula (12) and Fig. 4) and far from the motorway (formula (11) and Fig. 3).

The method of calculating L_{eq} for the first case is given by formula (7) which requires the knowledge of the traffic flow of vehicles of each class n_i (the notion of the class has been previously given), the intensity level of sound generated by passing typical vehicles of each class $L_i(t)$, and of the average background intensity $\bar{I}^{(0)}$, which can be neglected in the case of intensive traffic.

The measurement of $L_i(t)$ can be done only at night when traffic is small and it is possible to record the noise generated by a single vehicle. In order to describe the proposed method more precisely let us return to the example previously discussed (Fig. 1).

Neglecting the acoustic background and having available six records of the intensity level of sound $L_i(t)$ dB(A) related to the passage of single vehicles of particular classes (remembering that lorries and passenger cars may move on any lane (Fig. 1)), and taking relation (8) we obtain constants $a_1, \ldots, a_6$. Formula (7) then takes the form

$$L_{eq} = 10\log\frac{1}{I_0}(n_1 a_1 + n_2 a_2 + \ldots + n_6 a_6).$$

By this relation we can predict the equivalent level L_{eq} at a point A (Fig. 1) for any values $n_1, n_2, \ldots, n_6$ describing the traffic flow, e.g. for various times of day, various cases of traffic restriction, etc. To this extent formula (7) is of universal application.

In any case, we have to obtain only a few records of the intensity level of sound $L_i(t)$ dB(A) for particular classes of vehicles in order to determine the values of a_i from formula (8).

In calculating the equivalent level near to motorways it can be seen from formulae (11) and (12) that it is necessary to know the parameters describing the location of the point for which we calculate d_k, $y(x)$, the traffic parameters n_{jk}, and the values of sound power $W_j(V)$ for different types of vehicles depending on their speed. The relationship also contain the parameter ϱ, whose significance has previously been discussed.

The derived formulae apply only to flat areas. It seems advisable to extend theoretical works to give consideration to a great many parameters of the environment which are essential from the viewpoint of acoustics, such as e.g. the configuration of the terrain.

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MULTIPLE-LOUDSPEAKER SYSTEMS

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In this paper it is shown that a multiple-loudspeaker system composed of N similar loudspeakers can be replaced by N independent, similar single-loudspeaker systems. This provides the possibility of applying known methods of analysis and synthesis to multiple-loudspeaker systems. The efficiency of the multiple-loudspeaker system is calculated and measured results of the efficiency are given. Standardized transmittance functions for multiple-loudspeaker systems are discussed. The resonance frequencies of multiple-loudspeaker systems are determined.

Notation

B	induction in the gap of a loudspeaker magnet, T,
c	sound velocity in air, in normal conditions ($c = 345$ m/s),
C_{AB}	acoustic compliance of air in enclosure, m^5/N,
C_{AP}	acoustic compliance of the passive membrane suspension, m^5/N,
C_{AS}	acoustic compliance of the loudspeaker membrane suspension, m^5/N,
e_g	open-circuit output voltage of source, V,
$G(s)$	system transfer function,
i_g	current of source, A,
l	length of winding of the voice coil in the magnetic field, m,
M_{AA}	total acoustic mass of the co-vibrating medium, kg/m^4,
M_{AD}	acoustic mass of the membrane and voice coil of the loudspeaker, kg/m^4,
M_{AP}	acoustic mass of the passive membrane or air in the opening, kg/m^4,
M_{AS}	acoustic mass of the membrane and voice coil together with the co-operating mass of medium, kg/m^4,
M_{AT}	acoustic mass of the membrane and voice coil together with the mass of the co-vibrating medium of one loudspeaker in the multiple-loudspeaker system, kg/m^4,
Q_T	resultant magnification at resonance, considering all resistances,
R_{AB}	acoustic resistance of the loss in the opening or the passive membrane in an enclosure, kg/m^{4}s,
R_{AL}	acoustic resistance of loss in the gap, kg/m^{4}s,
R_{AP}	acoustic resistance in the opening or passive membrane, kg/m^{4}s,
R_{AS}	acoustic resistance of the loss of the suspension of loudspeaker membrane, kg/m^{4}s,
R_E	resistance of the voice coil of loudspeaker, Ω,
R_g	output resistance of the source, Ω,

S_D	active surface of the loudspeaker membrane, m^2,
$T_S = 1/\omega_S$	time constant of the resonance circuit, s,
U_B	volume velocity of the membrane in an enclosure, m^3/s,
U_D	volume velocity of the loudspeaker membrane, m^3/s,
U_L	volume veolocity in the loss gap, m^3/s,
U_0	resultant volume velocity, m^3/s,
U_P	volume velocity of the passive membrane or air in the opening, m^3/s,
η_0	reference efficiency,
ϱ_0	air density in normal conditions, kg/m^3,
ω_S	resonance frequency of the system, 1/s.

1. Introduction

A set of loudspeakers together with a suitable network in an enclosure of any type represents a *loudspeaker system*. The influence of the enclosure on properties of the system is significant only within the range of low frequencies. At medium and high frequencies the characteristics of the system do not depend essentially on the type of enclosure, but only on the loudspeakers used [3]. Thus the most important problem encountered during the design of a loudspeaker system is to ensure a proper cooperation of the loudspeaker with its enclosure at the lower frequencies.

Known methods of analysis and synthesis of loudspeaker systems (e.g. [5-8] permit to produce systems of various types which satisfy the design criteria with sufficient practical accuracy. In papers devoted to this problem prominence is given to those systems in which only one loudspeaker is operating in a given frequency range. The purpose of this paper is to show that it is also possible to use the methods mentioned above for the analysis and design of multiple-loudspeaker systems.

A multiple-loudspeaker system is produced by placing in any enclosure two or more loudspeakers destined for operation in a given frequency range. The use of different loudspeakers for joint operation is not advisable: if the loudspeakers differ in their resonance frequencies, the resultant characteristics will be worse than the characteristics of a loudspeaker transmitting a broader band. On the other hand, if the sound powers of the loudspeakers differ considerably, then the sound intensity increase will be small in relation to that produced by a loudspeaker of higher power, and the cost of the system will increase considerably.

2. Equivalent diagrams

Fig. 1 shows an equivalent diagram for a multiple-loudspeaker system with the voice coils connected in parallel. No consideration has been given in this diagram to the negligible [9] inductances of voice coils or to the radiation resistances. It is the diagram of a generalized system [2, 5] since it contains all the mechanical and acoustical elements of closed enclosures with an opening with a passive membrane (the membrane which is used in loudspeakers being suspended from a flexible spring in the enclosure opening), with a loss gap, and also infinitely large acoustic baffles. It is possible to obtain any specific system from it by assigning the suitable values to the elements.

Since the loudspeakers are similar, we have

$$R_{E1} = R_{E2} = \ldots = R_{EN} = R_E, \quad C_{AS1} = C_{AS2} = \ldots = C_{ASN} = C_{AS},$$

$$R_{AS1} = R_{AS2} = \ldots = R_{ASN} = R_{AT}, \quad Bl_1 = Bl_2 = \ldots = Bl_N = Bl,$$

$$M_{AT1} = M_{AT2} = \ldots = M_{ATN} = M_{AT}, \quad S_{D1} = S_{D2} = \ldots = S_{DN} = S_D$$

as well as

$$i_{g1} = i_{g2} = \ldots = i_{gN} = i_g/N, \quad U_{D1} = U_{D2} = \ldots = U_{DN} = U_D/N.$$

It is thus possible to represent the multiple loudspeaker, shown in Fig. 1, in the form of N independent, similar single-loudspeaker systems. Fig. 2 shows the equivalent diagram of a single-loudspeaker system obtained this way.

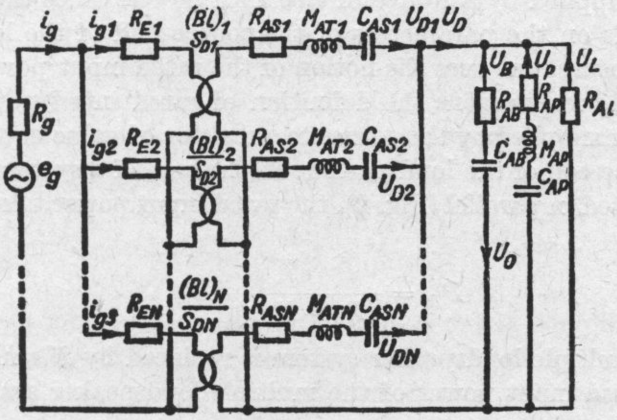

Fig. 1. Equivalent diagram of a generalized loudspeaker system with the loudspeakers connected in parallel

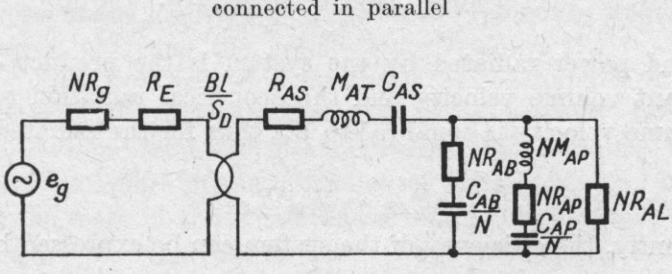

Fig. 2. Equivalent diagram of the single-loudspeaker system obtained from a multiple-loudspeaker system

The acoustic equivalent diagram of a multiple-loudspeaker system reduced to a single-loudspeaker system is shown in Fig. 3.

This reasoning also applies to the analysis of systems with loudspeakers connected in series or in a mixed manner.

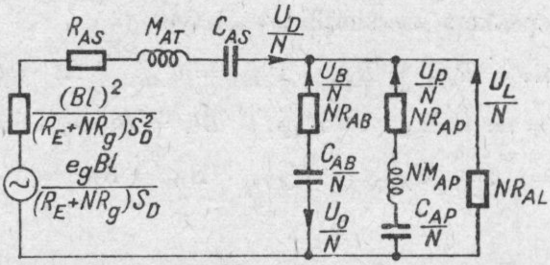

Fig. 3. Acoustic equivalent diagram of a generalized multiple-loudspeaker system with the loudspeakers connected in parallel

3. Efficiency

The *efficiency* of a loudspeaker system is the ratio of the radiated sound power to that supplied by a source of electrical power. Since the electric power supplied depends on the parameters of the source and of the loudspeaker and also on the enclosure, one uses the notion of the rated input power of the loudspeaker. For practical reasons the definition of rated input power accepted is that of the power supplied by the source to a resistor of value equal to the resistance of the voice coil of the loudspeaker. In the case of a system with the loudspeakers connected in parallel (Fig. 2), the rated input power takes the following form:

$$P_R = \frac{e_g^2}{(R_E+NR_g)^2} R_E. \tag{1}$$

Since the multiple-loudspeaker system is replaced by N single-loudspeaker systems, the rated input power of the multiple-loudspeaker system is N times higher:

$$P_{RN} = N\frac{e_g^2}{(R_E+NR_g)^2} R_E. \tag{2}$$

The sound power radiated by the system is the product of the square of the resultant volume velocity and the acoustical radiation resistance. The resultant volume velocity is equal [5] to U_0, thus for the radiated sound power we have

$$P_{AR} = |U_0|^2 R_{AR}. \tag{3}$$

Consequently, the efficiency of the system can be expressed by the relation

$$\eta = \frac{P_{AR}}{P_{RN}} = \frac{R_{AR}(R_E+NR_g)^2}{NR_E}\,\frac{|U_0|^2}{e_g^2}. \tag{4}$$

As has been shown in [1], the acoustic resistance of loudspeaker radiation in the listening room is equal to the resistance of a piston vibrating in an infinite acoustic baffle and it amounts to

$$R_{AR} = \frac{\varrho_0 \omega^2}{2\pi c}. \tag{5}$$

From the equivalent diagram of Fig. 3 we can evaluate U_0/N:

$$\frac{U_0}{N} = \frac{e_g Bl}{(R_E + NR_g) S_D j\omega M_{AT}} G(j\omega). \tag{6}$$

The function $G(j\omega)$, which represents a normalized transmittance for the system, corresponds to the transmittance function of a high-pass filter, of an order and type dependent on the type of enclosure. Equation (6) is at the same time a formula of definition for the function $G(j\omega)$. Substituting equations (5) and (6) into equation (4) we obtain the formula for the frequency efficiency characteristics of the system:

$$\eta = \frac{\varrho_0 (Bl)^2 N}{2\pi c R_E S_D^2 M_{AT}^2} |G(j\omega)|^2. \tag{7}$$

Over the frequency range, for which $|G(j\omega)| = 1$, the efficiency of loudspeaker systems does not depend on the enclosure and represents the reference efficiency:

$$\eta_0 = \frac{\varrho_0 (Bl)^2 N}{2\pi c R_E S_D^2 M_{AT}^2}. \tag{8}$$

This formula does not contain either the acoustic compliance of the suspension C_{AS} or the acoustic loss resistance of the membrane suspension R_{AS} – parameters which show the greatest scatter during measurements of various loudspeakers of the same type. On the other hand, these elements have an effect on the standardized transmittance function in the proximity of the resonance frequencies of the loudspeakers and on the resonance frequencies proper, without, however, changing the limit value $|G(j\omega)| = 1$.

As can be seen from equation (8) the increased number of loudspeakers makes the system more efficient. This increase is, however, not proportional to N, since as the number of loudspeakers increases so also does the mass M_{AT} which appears as a square in the denominator of the formula. The mass M_{AT} is the sum of the mass of the membrane of one loudspeaker M_{AD} and of the mass of the whole co-vibrating medium M'_{AA} referred to one loudspeaker:

$$M_{AT} = M_{AD} + NM'_{AA}. \tag{9}$$

The mass of the co-vibrating medium is inversely proportional to the square root of the surface area of the vibrating piston [10],

$$M_{AA} = \frac{k}{\sqrt{S_D}}, \tag{10}$$

k being the coefficient of proportionality.

Since the multiple-loudspeaker system contains N pistons with a total surface NS_D, the total acoustic mass of the co-vibrating medium M'_{AA} is expressed by the formula

$$M'_{AA} = \frac{k}{\sqrt{NS_D}} = \frac{M_{AA}}{\sqrt{N}}. \tag{11}$$

Introducing (11) into (9) we obtain an expression defining the acoustic mass of a loudspeaker operating in a multiple-loudspeaker system:

$$M_{AT} = M_{AD} + \sqrt{N}\, M_{AA}. \tag{12}$$

If the system contains only one loudspeaker, the acoustic mass is the mass M_{AS} and amounts to

$$M_{AS} = M_{AD} + M_{AA}. \tag{13}$$

Thus the acoustic mass M_{AT} of a multiple loudspeaker, expressed in terms of the acoustic mass of a single-loudspeaker system M_{AS}, is described by the relation

$$M_{AT} = M_{AS} + (\sqrt{N} - 1) M_{AA}. \tag{14}$$

It can be seen that the co-vibrating mass of each loudspeaker in the multiple loudspeaker is higher than the co-vibrating mass of the single-loudspeaker system.

Another limitation on the increase of efficiency of a multiple-loudspeaker system can be accounted for by the fact that increasing the number of loudspeakers necessarily increases the distance between them.

In order to determine the real dependence of the reference efficiency of the multiple loudspeaker on the number of loudspeakers, measurements were carried out. The sound power radiation by a loudspeaker system with one, two and three loudspeakers was measured in an anechoic chamber, the electric power supplied to the loudspeakers being in all cases constant. The power measurement was performed by an intermediate method of measuring the acoustic pressure on a hypothetical spherical surface, at the centre of which the system was located. The measurements indicate that the mean ratio of the efficiency of systems with one, two and three loudspeakers in the frequency range up to 1000 Hz is

$$\eta_1 : \eta_2 : \eta_3 = 1 : \sqrt{2} : 2, \tag{15}$$

the scatter of individual results not exceeding 10%.

It was also found that the mutual arrangement of the loudspeakers on the front plate of an enclosure of dimensions 0.32 m × 0.54 m did not affect the accuracy of the proportion (15).

SATHYANARAYANA [4] has carried out similar investigations by measuring the efficiency of a column speaker composed of loudspeakers with individual enclosures. Although the measurements were made only at the resonance frequency, his results were in agreement with relation (15).

Thus, the expressions describing the efficiency of the loudspeaker system with the number of loudspeakers $N \leqslant 3$ can take the following form:

$$\eta_0 = 2^{(N-1)/2} \frac{\varrho_0 (Bl)^2}{\pi c R_E S_D^2 M_{AS}^2}. \tag{16}$$

It should be noted that in this relation it was possible to replace the heavily determinable M_{AT} by M_{AS}, thus considerably simplifying the measurements.

The efficiency of loudspeaker systems analyzed here pertain to loudspeakers with the voice coils connected in parallel. A similar procedure for different connections of the voice coils leads to an identical relationship for the reference efficiency, but the functions $G(j\omega)$ are different.

4. Standardized transmittance function

As has been shown in section 2, a multiple-loudspeaker system can be reduced to a single-loudspeaker system with corresponding parameters. The reference efficiency of these systems does not depend on the method of connecting the loudspeakers while the transmittance functions, although of a similar type, have different coefficients.

For loudspeakers, placed in an infinite acoustic baffle (Fig. 4), the transmittance function takes the form

$$G(s) = \frac{s^2 T_S^2}{s^2 T_S^2 + \dfrac{s T_S}{Q_T} + 1}, \tag{17}$$

with $T_S = 1/\omega_S$ being the time constant of the resonant circuit.

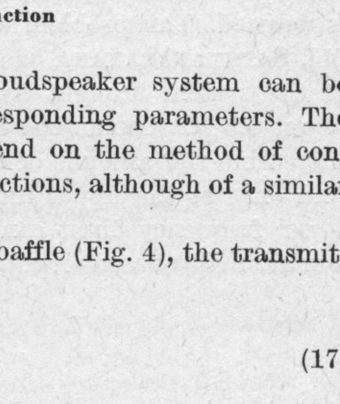

Fig. 4. Equivalent diagram of a multiple-loudspeaker system in an infinite acoustic baffle

The resultant magnification factor Q_T is: for parallel connection

$$Q_{TR} = \omega_S^{-1} C_{AS}^{-1} \left[R_{AS} + \frac{(Bl)^2}{(R_E + NR_g) S_D^2} \right]^{-1}, \tag{18}$$

for series connection

$$Q_{TS} = \omega_S^{-1} C_{AS}^{-1} \left[R_{AS} + \frac{(Bl)^2}{\left(R_E + \frac{R_g}{N} \right) S_D^2} \right]^{-1}. \tag{19}$$

A comparison of formulae (18) and (19) shows that the series connection ensures a better attenuation of the system if $R_g > 0$.

The resonance frequency of the loudspeaker system is expressed by the relation

$$\omega_S' = \frac{1}{\sqrt{C_{AS} M_{AT}}}. \tag{20}$$

Since the mass of the co-vibrating medium increases with the number of cooperating loudspeakers, the resonance frequency of the multiple loudspeaker is reduced. This problem was investigated by Sathyanarayana [4] and Small [7]. Sathyanarayana has shown that the reduction of the resonant frequency of the system brought by the increase of co-vibrating mass did not exceed about 0.5 Hz. Small has calculated that the increase of the mass, caused by increasing the amount of the co-vibrating medium, varies from about 2 to 8 % and only at the upper limit of this increase an insignificant reduction of the lower frequency limit can be observed. Similar results may be obtained from our estimates based on equation (14). Hence, one can assume that $\omega_S' = \omega_S$.

5. Power limitations

At high frequencies the permissible electric power of the system is limited by the ability to dissipate the heat through voice coils of loudspeakers. The permissible power $P_{E(\max)}$ of the whole system represents the sum of the powers $P_{E(\max)}$ of the loudspeakers included in the system. At low frequencies it is necessary to limit the deflections of the membrane of the loudspeaker so that it does not produce excessive nonlinear distortions. If the permissible electric power as governed by the membrane deflection in one loudspeaker is P_{ER}, then a power N times higher can be supplied to a system consisting of N loudspeakers. However, the maximum sound power will be more than N times higher than the maximum sound power of single-loudspeaker system as a result of the efficiency increase.

6. Conclusion

This paper presents an analysis of loudspeaker systems based on the transformation of a system composed of N similar loudspeakers into N independent, similar single loudspeakers and thus utilizes the methods of analysis of single-loudspeaker systems known from the literature. It was found that the efficiency of the multiple loudspeaker increases with an increasing number of loudspeakers. However, this increase is not proportional to N, because, among other reasons, of the increasing air mass which co-vibrates with the loudspeakers in the multiple-loudspeaker system. The relationship determining the increase of the vibrating mass of a loudspeaker, operating in a multiple-loudspeaker system in relation to the vibrating mass of a single loudspeaker, has been described. The coefficients defining the increase of the efficiency for one, two or three loudspeakers have been found empirically. It was found that the decrease of the resonance frequency, caused by the increase of the co-vibrating mass, is negligible.

The method of the approach to the problem of analysis stated in this paper is based on a transformation into equivalent single-loudspeaker systems. It can be used in the process of synthesis of multiple-loudspeaker systems of various types [3].

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THE APPLICATION OF ORGAN SOUND IN REVERBERATION MEASUREMENT IN THE CONCERT HALL OF THE WARSAW ACADEMY OF MUSIC

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Reverberation in the concert hall at the Academy of Music, Warsaw, was determined in the range from 50 to 8000 Hz using the organ as a sound source. The results obtained for 100-4000 Hz do not differ significantly from those obtained from conventional method.

To explain the differences in the range of lowest frequencies, the control measurement of the sound decay in low register organ pipes was performed. It was found that the time of sound decay in these pipes may actually be comparable to the reverberation time of the hall.

In the range of highest frequencies the measurement was carried out using a specially selected organ chord to get the sufficient energy in the upper part of the spectrum. The results are in agreement with those obtained by conventional measurements.

1. Introduction

For some time the increased interest in possible application of musical instruments, as natural sound sources in measurements of the reverberation time in concert halls, has been observed. These measurements are based on recording of a music piece, ending with a strong, suddenly interrupted chord, or on recording of a single chord with special harmonic structure and instrumentation. The reverberation time is evaluated at various frequencies with the aid of band-pass filter.

Reverberation measurement using natural sound source has several advantages. Firstly, it may be expected that this method preserves the natural conditions of sound radiation in a hall. The kind of a sound source, the placement of instruments on the stage, their spectral and directional characteristics etc. are adequate to those usually occurring during the concert. Secondly, the use of natural sound sources enables the measurements with the audience being present in the hall, i.e. in the condition when routine measurements are not possible. The recording made for measuring purposes must satisfy some special conditions of a microphone placement. Also the microphones used should not have directional characteristics. The artistic phonographic recording, often distorting the natural reverberation, is therefore not advisable.

Apart from the advantages mentioned above the measurement with natural sound sources presents also some disadvantages. One of the most serious is the relatively low acoustic power of musical instruments and difficulties in achieving immediate damping of their radiation which is indispensable for evaluating the reverberation correctly. The lack of power, necessary for building up the appropriate sound intensity in the hall, is most evident at the extremes of the auditory spectrum, thus making the evaluation of reverberation at very high and very low frequencies difficult or even impossible.

The distortion of measurement, caused by the low rate of sound decay at the source, may be of two kinds. First, it may result from inefficient additional damping or non-simultaneously applied disconnection of power supply in vibrators of musical instruments. This may concern either a single instrument (e.g. non-simultaneous release of keys) or a group of musicians or singers. Secondly, the distortion of measurement may be caused by the insufficient rate of sound decay in such cases where additional damping cannot be introduced. This factor eliminates some instruments from their possible application as natural sound sources in reverberation measurements [4].

Among many musical instruments that might be considered as suitable in reverberation time measurements, the organ seems to be particularly useful. This is because of several reasons. Firstly, the organ may produce high level of spectral energy in broad frequency range. Secondly, the vibrators of organ pipes have small inertia and their vibrations quickly disappear (this claim does not fully pertain to the lowest register which will be discussed later on). Thirdly, the organ is operated by one musician, which is very convenient for technical reasons in reverberation measurements. Another favourable fact in measuring reverberation using organ as a sound source is that this instrument can be found in most concert halls and churches. LOTTERMOSER [2] has described an interesting case of using the recording of the organ sound for measuring reverberation. It comprised the evaluation of acoustic conditions of the church "Frauenkirche" (Dresden), which was completely destroyed during the World War II, on the basis of old recordings of the organ. Thus additional clues to the reconstruction of the interior were obtained.

2. Reverberation measurement over full frequency range using single organ chord

The organ at Warsaw Academy of Music is equipped with four manuals, pedals and 60 flue and reed stops. Electropneumatic action is combined here with a mixed, slider- and ventil-wind-chest action (Great). The instrument is spaced at the two opposite side walls of the concert hall at the stage level. Total volume of the hall is 5.500 m^3.

For reverberation measurements two following chords were recorded:

– the final chord of J. S. Bach's organ toccata in D minor,

– the final chord of J. S. Bach's organ prelude in E flat major,

– a chord obtained by depressing all the twelve keys in lowest octave of the Great (tutti combination).

Sound intensity levels, measured in the middle of the 8th row of the orchestra seats, were 92 dB for D minor chord, 93 dB for E flat major chord and 95 dB for the twelve-tone chord. The chords are presented in Fig. 1. The results of the spectral analysis of these chords, obtained with the use of B-K 1614 1/3 octave band analyzer, are presented in Fig. 2.

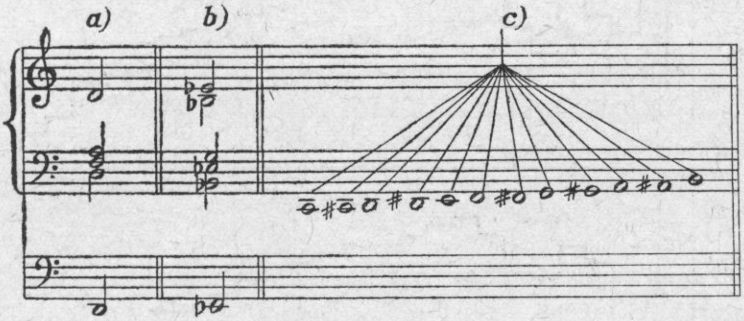

Fig. 1. Musical notation of three organ chords used in reverberation measurements a) D minor chord, b) E flat major chord, c) chromatic twelve-tone chord

The frequency analysis of D minor and E flat major chords displays the existence of partials in a considerably wide range. High intensity levels within the frequency bands, centered at 40 and 50 Hz, are produced in pedal register. It should be noted here that acoustic power distribution in the sound spectrum of organ chords may significantly change, depending on the specimen of the instrument. Differences are most significant for the lowest and highest frequencies [3].

The analysis of the twelve-tone chord as compared with the analyses of other chords shows the lack of the lowest partials in frequency bands 40 and 50 Hz. This is due to the fact that manual keyboards only were used in this case, connected by means of coupling mechanism. Manual keyboards of the organ in Warsaw Academy of Music have only two 16-foot stops, with the frequency of the lowest tone C – 32.7 Hz. These registers, Quintadena and Rankett, are characterized by weak fundamental tones. To secure the presence of strong low partials in the spectrum, it should be advisable to use the pedal keyboard. It appeared, however, impossible because of the difficulties in simultaneous release of 12 consecutive pedal keys without a special device.

Spectral analysis of the chords used leads to the preliminary conclusion that the measurement of reverberation by means of the organ sound may be

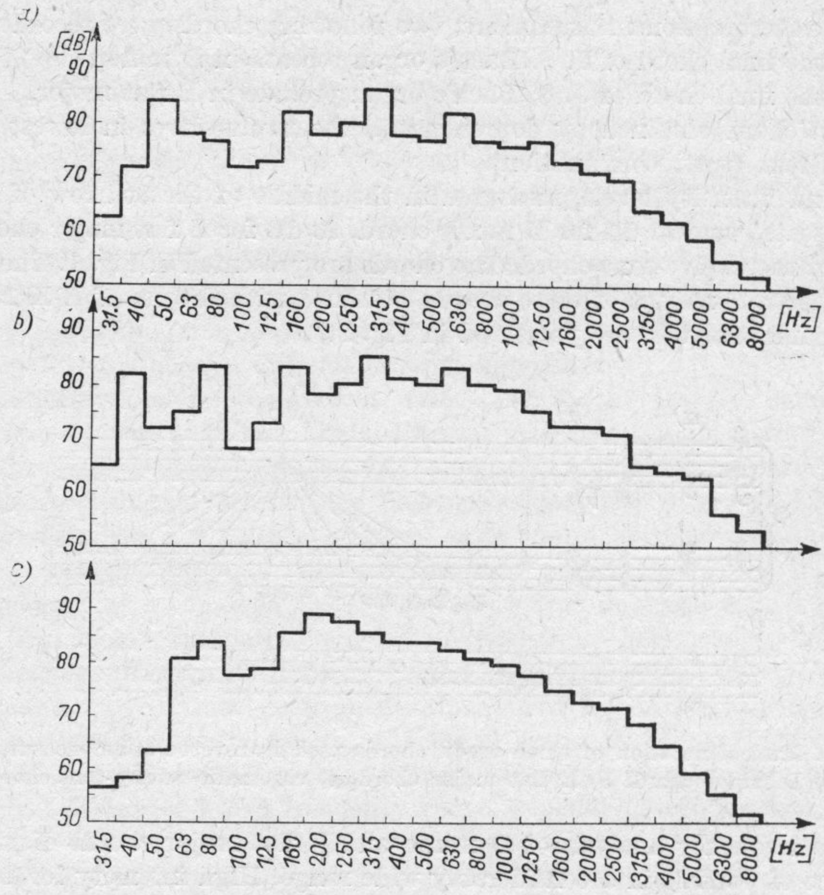

Fig. 2. The analysis of triadic organ chords used in reverberation measurements
a) D minor chord, b) E flat major chord, c) chromatic twelve-tone chord. On the abscissa the mean values of 1/3 octave bands are given in Hz

difficult to accomplish at very low and very high frequencies because of the lack of sufficiently strong spectral components. This conclusion has been confirmed in further investigations.

The recording of the organ sound (stationary sound and the decay) was made by means of 4 condenser microphones with nondirectional characteristics and a 4-track tape recorder.

The placement of microphones was following:

1. in the middle of the 8th row of seats (height 1.5 m),
2. in the middle of the first row of the circle (height 1.5 m),
3. over the conductor's stand (height 3 m),
4. over the 8th row of seats, height 8 m (the so-called reverberation microphone)

A signal from each of the microphones was recorded separately.

Each of the four recordings of the three chords used were analysed by successive filtering through 23 1/3 octave filters with central frequencies from 50 Hz to 8 kHz. Values of the reverberation time were found from the rate of level decay in consecutive frequency bands in which signal to noise ratio was high enough. The reverberation time values, obtained in this way from four different microphone recordings, have been averaged for each frequency band. The characteristics of reverberation time, obtained by using three different organ chords and averaging microphone positions, are presented in Fig. 3 (a, b, c).

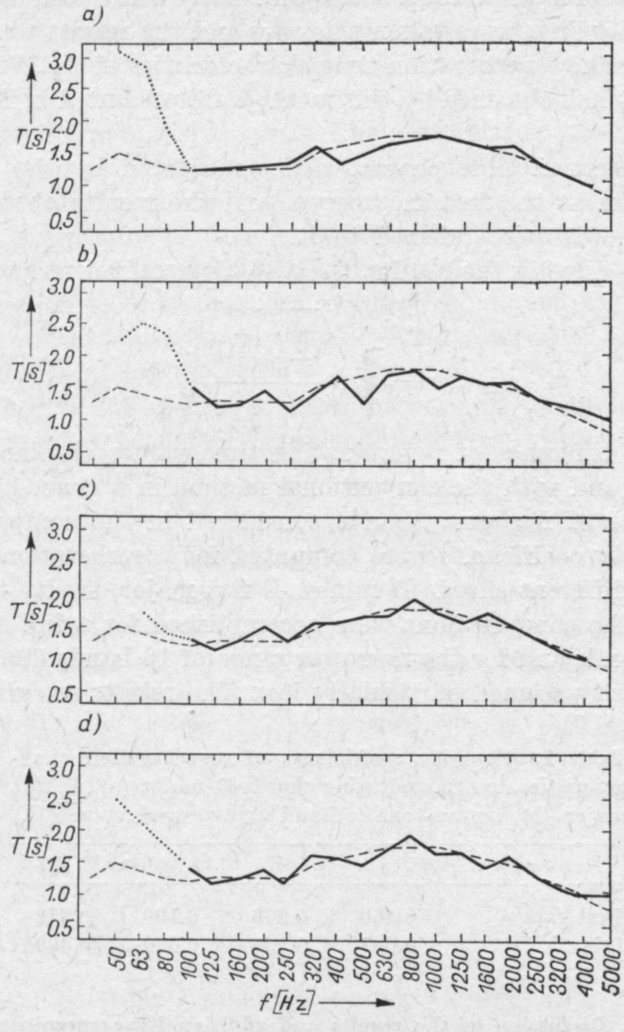

Fig. 3. The frequency characteristics of reverberation at the Concert Hall of Warsaw Academy of Music, taken with the use of the organ as a sound source. Broken line represents the reference characteristic obtained by conventional method (data of the Film Research Centre). Dotted line – results distorted due to long extinction time of the pipes
a) D minor chord, b) E flat major chord, c) twelve-tone chord, d) mean value

The curve in Fig. 3d presents the result of averaging the previously showed values.

All curves representing measured (a, b, c), Fig. 3, and averaged (d) values of reverberation time were compared with the reverberation characteristics obtained by the Acoustical Laboratory of the Film Research Center in Warsaw using conventional method. In these comparative measurements frequency modulated tones and bands of noise were used for each frequency 1/3 octave apart. Every measurement was taken with 10 different placements of the microphone and 2 placements of loudspeakers. The sound decay was traced by level recorder. 920 separate recordings were taken this way and the results were averaged for each frequency. The reverberation time characteristics of the Warsaw Academy of Music concert hall obtained by this method are presented in Fig. 3 as broken lines.

The reverberation time characteristics measured by the Film Research Center was taken as the freerence curve, and the results obtained by means of the organ chords were referred to it.

To compare directly the results, the disparity coefficient r was used,

$$r = \sqrt{\frac{\sum_{i=1}^{n}(\Delta T_i)^2}{n}},$$

where ΔT_i is a difference of reverberation time values obtained with the use of organ chords and with the conventional method in a given 1/3 octave band, and n – number of frequency bands considered in the comparison.

The disparity coefficients were computed for reverberation characteristics obtained using different chords D minor, E flat major, twelve-tone, and mean characteristic. The computations were accomplished for a full accessible range of 1/3 octave bands and for the narrower range of 16 bands (100 Hz to 4 kHz), where the disparity seemed particularly low. The results are given in Table 1.

Table 1. Disparity coefficients of reverberation characteristics determined using chords D minor (a), E flat major (b), twelve-tone (c), and at averaged values (d)

r	a	b	c	d
Full range	0.460	0.366	0.190	0.312
0.1 - 4 kHz	0.087	0.165	0.079	0.099

3. Discussion of the results and additional measurements

On the basis of the results presented, it may be claimed that reverberation measurements with the organ as a sound source led to the results very close to those obtained from conventional measurements. The disparity coefficient was lowest in measurements with the use of the twelve-tone chord. As may be seen from Fig. 3, a particularly good agreement with the results from the conventional measurements was obtained in the range 100 - 4000 Hz. Also in this range the disparity coefficient obtained with the twelve-tone chord was the lowest. At frequencies above 4 kHz the spectral density was in most cases not sufficient and the signal to noise ratio was too small for quantitative assessments of the decay rate.

At frequencies below 100 Hz the reverberation values obtained were systematically higher than the ones measured by means of a loudspeaker. This could suggest that apart from the lower accuracy of measurements, due to insufficient acoustic energy, there occurred a systematic distortion caused by the high rate of sound decay in the organ itself.

To verify this hypothesis, tones of single pipes from the lowest registers were recorded, using a microphone with unidirectional characteristics, placed 10 cm from the mouth of the pipe. The time of sound decay of the pipe was estimated from the records as a hypothetical time necessary for the decrease of the acoustic pressure level by 60 dB. Per analogy to the conditions used in the determination of the reverberation time in rooms, the time of decay of vibrations in musical instruments thus defined was called *self reverberation time* [4] or *normalized decay time*.

Table 2 presents the results of measurements of normalized decay time for flue and reed pipes belonging to various organ stops in the Concert Hall of Warsaw Academy of Music.

Table 2. Normalized sound decay time of some organ pipes in the Concert Hall of the Academy of Music, Warsaw

Stop	Pipe	Pitch	T [s]
Diapason 16′ Ped.	open flue	C_1	3.25
Diapason 8′ II Man.	open flue	C_2	0.80
Octave 4′ II Man.	open flue	C_3	0.41
Diapason 8′ II Man.	open flue	C_3	0.44
Octave 4′ II Man.	open flue	C_4	0.42
Subbas 16′ Ped.	stopped flue	C_1	2.60
Trombone 16′ Ped.	reed	C_1	1.00
Trumpet 8′ II Man.	reed	C_2	0.50
Trumpet 8′ II Man.	reed	C_3	0.40
Trumpet 8′ II Man.	reed	C_4	0.42
Rankett 16′ IV Man.	reed	C_1	0.47

It seems that the decay time of the organ sound is directly connected with the size of a resonator. This is confirmed by the fact that the pipe C_1 Rankett 16′ with a short resonator has a relatively short time of decay (0.47 s). Besides, flue pipes have longer time of decay than reed pipes of the same length. Thus one may conclude that reed-stop combinations are most appropriate in measuring reverberation time because of the short time of decay and rich spectrum.

Evaluating the sound decay time of organ pipes is not very reliable due to the possible influence of the reverberation of the hall. To decrease this effect, the microphone was placed very close to the pipe under investigation. The positive factor influencing hypothetical accuracy of the measurement is that the reverberation time of the Concert Hall of Warsaw Academy of Music in the range from 50 do 100 Hz, measured by conventional method, is 1.50 to 1.35 s, respectively.

The evaluation of the sound decay time of organ pipes leads to the conclusion that at frequencies smaller than 100 Hz this time is comparable to the reverberation time of the hall itself. Application of the organ as a sound source in measuring reverberation at low frequencies is not possible in halls like the Concert Hall in Warsaw Academy of Music with short reverberation time in low frequency range. In halls, where reverberation time at frequencies below 100 Hz is much longer (as it usually happens in churches), the measurements of reverberation with the use of organ sound may be accomplished also at low frequencies. This possibility allowed LOTTERMOSER [2], as already mentioned, to reconstruct the reverberation conditions of the Frauenkirche church in Dresden, where reverberation at lowest frequencies exceeded 4 s.

Above 4 kHz reverberation measurements using previously described organ chords were practically impossible, due to insufficient acoustic energy of the organ chords used.

To complete the results, an additional measurement of reverberation time at frequencies 4 kHz, 5 kHz, 6.3 kHz and 8 kHz was accomplished, applying identical measuring technique as in the main experiment. The sound source in this case was a group of pipes with greater acoustic energy at high frequencies. The twelve-tone chord C_6-B_6 natural was used, with all stops drawn. The value of reverberation obtained in 1/3 octave bands at medium frequencies 4 kHz, 5 kHz, 6.3 kHz and 8 kHz were: 1.05, 0.80, 0.65 and 0.60 s, respectively. The respective values obtained with conventional method are: 1.00, 0.80, 0.70 and 0.50 s, so there is a high coincidence of the results also in this case.

4. Conclusions

The measurements of the reverberation time at the Concert Hall of Warsaw Academy of Music lead to the conclusion that the application of the organ in reverberation measurements gave the results very close (within definite frequency range) to the ones obtained by a conventional method. This agreement might have resulted partly from the fact that the organ pipes at the Concert Hall of the Academy are situated over two opposite walls and the sound produced is well diffused. To measure the reverberation in the frequency range from 100 to 4000 Hz, a recording of the chords ending a musical phrase can be used. The measurement may be accomplished on the conventional recording of a concert, provided an adequate microphone technique was used, as specified earlier. Evaluating reverberation with the use of an organ chord at frequencies below 100 Hz is possible only in such halls in which the reverberation at these frequencies is considerably long, e.g. in some churches. The measurement at frequencies higher than 4 kHz is possible by means of specially matched organ chords with appropriate level of acoustic energy within this frequency range.

In the above-described investigations two tradic chords, D minor and E flat major, in "tutti" registration have been used. These chords were endings of well-known music pieces. The agreement of the results obtained with conventional measurements has not been much worse than in the case of a specially matched twelve-tone chord. Surely, a similar result may be obtained applying other chords completing phrases of various music pieces, provided the appropriate registration has been secured.

The registration of the organ chords, applicable to reverberation measurements, should give possibly high intensity level and equal distribution of energy in wide frequency range. Suggested here is the "tutti" combination, including reed stops rich in higher partials, or "pleno" registration with the usual choice of foundation and mixture stops. Highly unsuitable are chords played on flute stops which lack full overtone series.

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THE BOWED STRING AS THE TWO-TERMINAL OSCILLATOR

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The paper presents a method of the evaluation of the shape of bowed string oscillations. The method employs the analogy between the bowed string and the electrical, two-terminal oscillator. The velocity-dependent friction force between string and bow, investigated experimentally many years ago, was applied as a stimulating function of the oscillator. The evaluated shapes and amplitudes are in good agreement with experimental results.

1. Introduction

The analysis of the generating process of bow-sustained string vibrations is a classical, but still valid subject in the field of both musical acoustics and the theory of oscillations. The reality of this problem results from the fact that the successive investigations yield merely approximate solutions because the string vibration maintenance mechanism is highly complicated.

It should be remembered that the bow simultaneously generates transverse, longitudinal and torsional vibrations of the string as well as of its fixed points, with the consequent so-called octave vibrations [11]. All these vibrations interact by superposition at the incommensurate frequencies of their individual modes [8].

It is to be noticed, moreover, that the string corresponds to a distributed-constant system where propagation conditions affect the shape of vibrations. Moreover, string vibrations are highly nonlinear due to nonlinear relation between transverse, longitudinal and torsional elastic forces and their corresponding displacements. Also the bow, vibrating jointly with the string, brings other nonlinear effects into string motion [7]. Thus, many papers devoted to the vibration theory of strings, deepening the analysis of various types of nonlinearities, disregard the nonlinear problems which arise from the excitation mechanism, i.e. from the bow-string excitation [14, 15].

The complexity and difficulty of the bowed string vibration maintenance analysis justifies the interest which accompanies subsequent trials of finding approximate solutions of the problem.

Although the nature of bowed string oscillations has been thoroughly investigated already in the XIXth century by HELMHOLTZ [13] the mechanism of its excitation into self-sustained oscillations remained an ambiguous problem for a long time. Successful trials of solving it up began about sixty years ago with CHARRON'S thesis [3].

The author investigated with a special attention the dependence of the friction force between bow and string upon their mutual velocity acting as the so-called stimulating function in the process of vibrations generation [5].

Giving up to present here the results of numerous contributions to the problem of the self-oscillating bowed string, it should be mentioned that relatively frequent misinterpretations of observed phenomena may be found in the literature of the subject. An interpretation of a typical oscillogram of the string velocity versus time, where parts of the vibration period corresponding to intervals of the steady and of the varying velocity are obviously interchanged, may serve as an example [6]. This interpretation, given by authors of many outstanding papers on the vibration theory of the violin, proves that some important problems here are still to be studied and explained.

A friction force versus velocity diagram, presented in a JASA paper [12], provides another example. A continuous curve showing zero friction force at zero relative velocity is given there, which is discrepant from all typical diagrams of this kind and from measured characteristics of string-bow systems [3, 6]. Friction characteristics at zero velocity have necessarily a discontinuity called *dead zone*. It plays an essential role in vibrations maintenance, providing proper conditions for a stimulating function exist.

So, it seemed reasonable to examine once more the problem of string excitation by a bow, even in a simplified way.

Attempting an analytical formulation of the problem the following assumptions have been accepted:

– only transverse vibrations are considered, as energetically prevailing,

– lumped-constant mechanical system is assumed as a string model,

– the damping of the free vibrating string is provided to be negligible in comparison with the bowed string one.

2. Bow-string system as a two-terminal oscillator

String vibrations maintained by permanent bow motion are typical self-sustained oscillations. Therefore, a bow-string system may be analysed as a mechanical oscillator. Although description methods of mechanical systems, based on the theory of electrical oscillators, are not frequently applied, such a procedure may be easily adopted by the use of electromechanical analogies.

The considered case of a string-bow oscillator may be presented as the electrical circuit shown in Fig. 1. This circuit permits the formulation of mechanical string motion equations (2) as an equivalent of the electrical voltage-

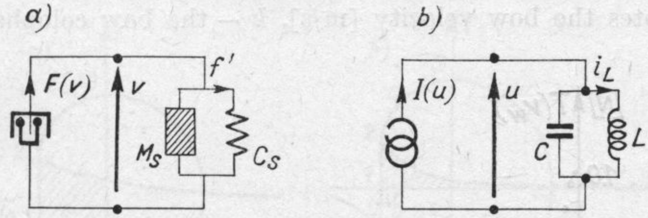

Fig. 1. Two-terminal oscillating system: a) mechanical system, b) its electrical motional analogue

-current equations (1). The corresponding quantities of (1) and (2) are coupled by the motional analogy:

$$\frac{du}{d\tau} = -\sqrt{\frac{L}{C}}\, I(u) - i, \quad \frac{di}{d\tau} = u, \tag{1}$$

where

$$\tau = t/\sqrt{LC}, \quad i = i_L\sqrt{L/C} = \Phi/\sqrt{LC},$$

i – variable quantity proportional to coil current i_L, Φ – magnetic flux in the coil;

$$\frac{dv}{d\tau} = -\sqrt{\frac{C_s}{M_s}}\, F(v) - f, \quad \frac{df}{d\tau} = v, \tag{2}$$

where

$$\tau = t/\sqrt{C_s M_s}, \quad f = f'\sqrt{C_s/M_s} = x/\sqrt{C_s M_s},$$

f – variable quantity proportional to force f' applied to string compliance (see Fig. 1), x – string displacement.

The nonlinear function $F(v)$ of equation (2) represents the friction force between the bow and the string as a function of the relative string velocity. The function values found empirically by CHARRON [3] are presented in Fig. 2 given in their original form. It is obvious that this function does not depend on the sense of the bow movement, hence the curve $T(v_w)$ can be completed antisymmetrically to the left.

Assuming the string absolute velocity as an independant variable, i.e. transferring the curve $T(v_w)$ by the value of bow velocity V_0, we obtain formula (3). It performs the role of a stimulating function (see Fig. 3), like similar negative resistance functions do in typical two-terminal electrical oscillators [1],

$$F(v) = \frac{T_0 \operatorname{sign}(v+V_0)}{1+k|v+V_0|}, \tag{3}$$

where V_0 denotes the bow velocity [m/s], k — the bow colophanying factor.

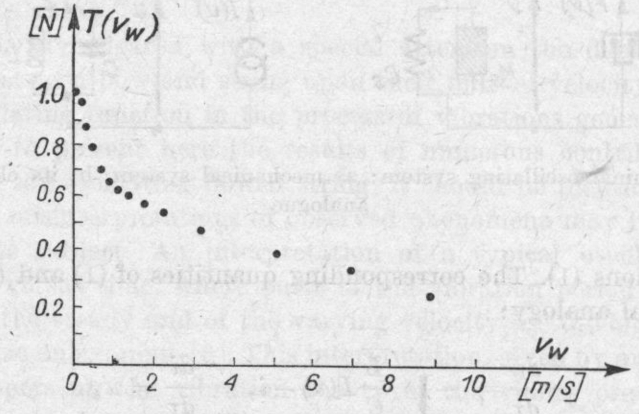

Fig. 2. Friction force between bow and string as a function of its relative velocity [3]. The dependence is given by the formula $t(v_W) = T_0/(1+kv_W)$, where T_0 — static friction force [N], k — bow colophanying factor

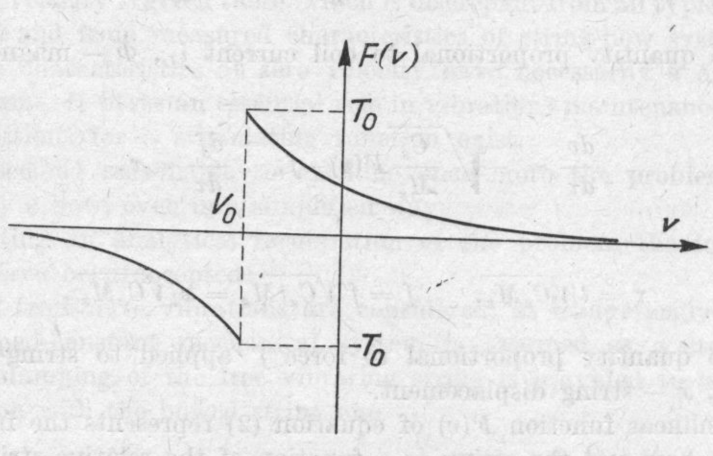

Fig. 3. Bow acting as the string stimulating function

3. Solution of the string motion equation

Equation (1), with nonlinearity given by formula (3), describes self-sustained oscillations. The equation variables $f(\tau)$ multiplied by a constant factor $(C_s M_s)^{1/2}$ and $v(\tau)$ represent the string displacement and the absolute string velocity. Since analytical solutions of this equation type are not available, a numerical method has been used. The method enables to calculate [10] steady-state values of $v(\tau)$ and $f(\tau)$. It is also possible to find limit cycles of the equation by means of the widely known Lienard's graphical construction, as shown in Fig. 4.

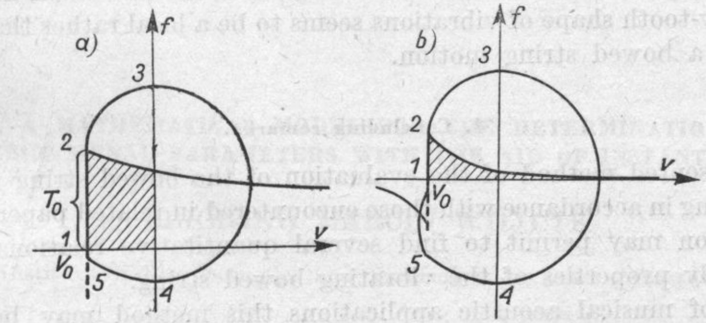

Fig. 4. Limit cycle of string vibrations. Exciting in intervals 1-2, 2-3, 4-5; damping in intervals 3-4, 5-1

String velocity: a) does not exceed bow velocity, b) exceeds bow velocity

Both numerical calculations and graphical method confirm the possibility of an increase of the string velocity above the bow velocity [6]. Really, at appropriately chosen movement parameters, one can observe an apparently paradoxical phenomenon of the string preceding the bow during a part of a period (Fig. 4b, 5a).

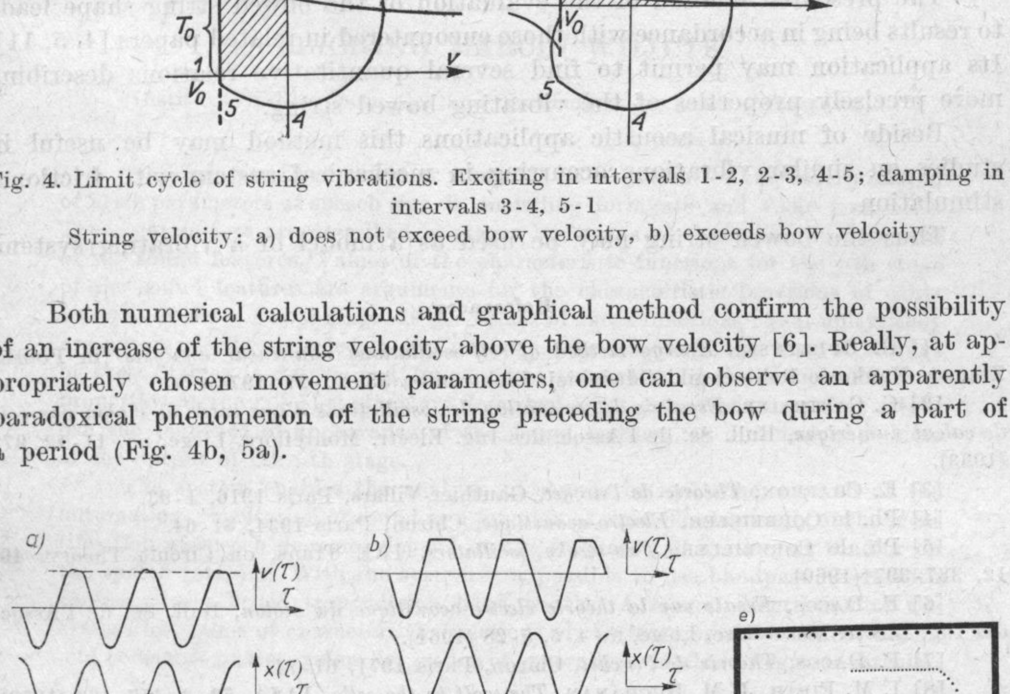

Fig. 5. Shapes of solutions of string motion equations. The following values of the parameters V_0, T_0, k of equation (3) are taken: a) 0.8; 0.1; 0.9; b) 0.4; 0.53; 0.415 (empirical values [2, 3]); c) 0.2; 0.9; 0.9; d) 0.15; 1.8; 1.8; e) computer printout from which the function $v(\tau)$ shown in Fig. 5c has been derived. Other functions have been formed in a similar way

Results of numerical calculations (see Fig. 5) show that the saw-tooth shape of a string displacement as a function of time is not only one possible, although it was just so presented in papers [8, 13].

Vibrations of the saw-tooth type can be produced only in a relatively narrow range of parameter values which condition the oscillation maintenance. Then the saw-tooth shape of vibrations seems to be a local rather than a general property of a bowed string motion.

4. Concluding remarks

The presented method of the evaluation of the bowed string shape leads to results being in accordance with those encountered in related papers [4, 5, 11]. Its application may permit to find several quantitative relations describing more precisely properties of the vibrating bowed string.

Beside of musical acoustic applications this method may be useful in studies on similar vibrations occurring in mechanical system with frictional stimulation.

Thus the bowed string may be used as a model of a vibrating system.

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A MATHEMATICAL MODEL FOR THE DETERMINATION OF SPEECH SIGNAL PARAMETERS WITH THE AID OF INSTANT MEMORY

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In this paper the concept is presented of a system for the determination of basic parameters of speech signals, including formants and their transients. These parameters are described in stages by means of characteristic functions of the sound features. Values of the characteristic functions for the j-th stage of the sound features are arguments for the characteristic functions of other features of the $(j+1)$-st stage. If the values of these functions equal unity, they are automatically recorded in the memory. The length of the time of storage of these values is determined by physical properties of the sound. Physical properties at the $(j+1)$-st stage are described by means of relations that establish the sequence of appearance of the sound features as a function of time at the output of the j-th stage.

The system enables the analysis of sounds in real time, as well as the automatic segmentation of sound as a function of time. The accuracy of the identification of speech does not depend on the frequency of the larynx tone or on the speech rate, etc. With the system it is possible to use bandpass filters with comparatively broad transmission band widths. A formal description of the system by means of characteristic functions of the features allows direct design and construct of the system by means of generally available integrated circuits.

1. Introduction

Present methods for the identification of speech sounds take advantage of the formant theory of the speech signal. The sounds developed as a result of inducing vibration of the vocal chords (larynx tone), and by air passing through the contracting oral cavity and larynx cavity (noise), are modulated and filtered through the larynx cavity, the oral cavity separated by the tongue, and nasal canal, and have a formant structure. That is to say that one can distinguish frequency ranges in which the spectral energy density assumes locally maximum values, and also frequency ranges in which the spectral energy density decreases strongly to minimum values (antiformants). The changes of geometrical dimensions of the cavities (especially of the oral cavity), observed during speaking, cause the formation of transients, i.e. changes of the frequencies of the formants with time. Formants and their transients are the basic acoustic-phonetic parameters used to describe speech signals and, especially, voiced sounds.

From point of view of essence transmission a huge excess of information in a speech signal occurs. In the system presented in this paper the reduction of this excess is connected with the measurement of basic parameters: formants and their transients. The measurement is effected automatically by the spectral analysis of a speech signal.

2. Assumptions, conceptions and working range

The direction of this paper has been provided by the need to build a system for the automatic identification of speech in real time, which could be constructed with generally obtainable electronic subassemblies, i.e. integrated circuits and medium-size computers.

Proposed and constructed sound identification systems face various difficulties that result from the physical nature of a speech signal, from imperfections of the individual subsystems and from simplifications introduced as assumptions in the solution of technical problems.

In view of the high operating costs of computers with very large memories, it is more advantageous to use a system permitting input of a speech signal into the computer (Fig. 1) with the basic parameters already determined. Such

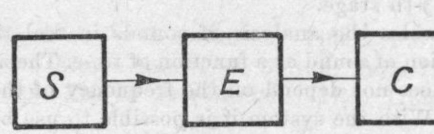

Fig. 1. General block diagram of the system for speech identification
S – source of speech signal, E – system of singling out the basic parameters of the speech sound, C – universal digital machine

an approach would permit the construction of useful systems within which the computer could be utilized for performing various tasks, its operation being controlled by means of a speech signal. This paper presents the concept of a system intended to determine the basic parameters of a speech signal (block E in Fig. 1).

To maintain the general character of these considerations no specific technical parameters of individual subsystems have been stated. In the description the possibility has been accepted of the construction of a set of bandpass filters with parameters to enable them to locate (as functions of time and frequency) the formant parameters of a speech sound with sufficient accuracy. These assumptions are substantiated by the results of works cited by SAPOZHKOV [5] and, especially, the results of works obtained by KUBZDELA [4]. The contemporary literature on the subject of the automatic identification of speech points to the necessity of also considering in the analysis the dynamics of changes in the measured parameters of a speech signal. This paper is a proposal to associate the already known parameters with their dynamics and changes in the speech signal (e.g. the definition of a formant is – according to the conception of their detectability stated in items 9 and 12 – very similar to classical definitions). The criteria of the determination of basic parameters of speech signal, accepted simultaneously, are: the magnitude of relative changes of the signals at the output of band filters (i.e. positive and negative amplitude increments), the rate of these changes, and the time intervals between individual and predetermined changes of speech signals. The value of each of determined parameter is stored in the system (in the memory) during a short period of time (the so-called *instant memory time*).

The concept of instant memory and its role in the process of determing the parameters of the speech signal are presented in the following sections. The final verification of the conception presented can unfortunately only be effected experimentally.

The anticipated effects of the conception are stated in the concluding section of the paper.

3. The organization of instant memory

In the proposed system the speech signal is described in stages by means of characteristic functions of the sound features. Each of these functions describes a specific property (feature) of the speech signal. The function assumes a value equal to unity when the shape of the speech signal conforms to the assumed description. Only those properties of the speech signal are described which are of essential significance for the determination of basic speech parameters.

The system is built hierarchically: the values of characteristic functions of the features of the speech sounds of the j-th stage are arguments for the characteristic functions of features of these sounds at the $(j+1)$-st stage of processing. When the characteristic functions assume values equal to "1", an automatic recording of these values into the memory is made.

The length of time t_{pj} of the storage of the value "1" of characteristic functions of the j-th stage of the system are described in the following manner:

$$t_{p(j-1)} \leqslant t_{pj} \approx \tau_{pj}, \tag{1}$$

where $j = 2, 3, \ldots, \tau_{pj}$ is the experimentally determined time of the occurrence of properties of the speech sound, as described at the j-th stage of the system. Thus the duration of time t_{pj} is determined by physical properties of sounds.

It is assumed that the access to the memory (i.e. to the values of the arguments of characteristic functions at each stage of the system) is automatic. The times of reading of the individual values "1" of characteristic functions from the memory, at the transition from the j-th stage to the $(j+1)$-st stage, correspond to the times of their storage at the j-th stage.

Physical properties of sounds at the j-th stage are described by means of relations of partial order in time for the value "1" of the characteristic functions of features of the $(j-1)$-st stage.

Let $D = \{d_1, d_2, \ldots, d_g, \ldots, d_h, \ldots, d_G\}$ be a set of various properties of the sound signals determined at the j-th stage. Any point on the time axis $t_a \in T = \{t_1, t_2, \ldots, t_S\}$ is the time of occurrence of the property d_g of the sound if the value of the characteristic function $Cd_g(t')$ of the feature d_g of the sound satisfies the condition

$$Cd_g(t') = \begin{cases} 0 & \text{for } t' < t_a, \\ 1 & \text{for } t' > t_a. \end{cases} \tag{2}$$

Thus the individual times $\{t_1, t_2, \ldots, t_S\}$ are not known before the analysis of the signal of a speech sound. These times are determined automatically in the course of the analysis as the times at which the characteristic functions have the value "1".

The process of determining the values of the times of the set T can be regarded as a process of an automatic segmentation of the speech signal as a time function.

The function $Cd_g(t')$, which satisfies condition (2), will be denoted by the symbol $Cd_g(t_a)$ or $Cd_g(t)$. The apparent argument t or t_a of the characteristic function will denote the first moment at which this function has the value "1". Let us form the set of functions $\{Cd_g(t_a)\}^{D\times T}$. This is a set of all features determined at the j-th stage during the entire period of analysis (these features may repeat).

Let us select from the set $\{Cd_g(t_a)\}^{D\times T}$ each pair of elements $Cd_g(t_a)$, $Cd_h(t_b)$ $(d_g, d_h \in D; t_a, t_b \in T)$ for which the relation

$$0 < |t_a - t_b| < t_{pj} \tag{3}$$

is satisfied.

For each pair of elements of the set $\{Cd_g(t_a)\}^{D\times T}$, for which relation (3) is satisfied, there is a relation $\in\!-$ of partial order in time defined in the following manner:

$$Cd_g(t_a) \in\!- Cd_h(t_b) \Leftrightarrow t_a \geqslant t_b. \tag{4}$$

The relation $\in\!-$ establishes the sequence of the occurrence of two features d_g and d_h of sounds as a time function; the time between the occurrence of these features is shorter than t_{pj}.

Pairs of elements of the set $\{Cd_g(t_a)^{D\times T}\}$, for which relation (3) is not satisfied, are not arguments for characteristic functions of the speech sound of the $(j+1)$-st stage.

The input data for each stage of the identification are selected in this manner. The reduction of redundant information, and also the extraction of useful information from disturbances, is thus effected at each stage of the operation.

By means of pulses of duration t_{pj} it is possible to detect, with technical facilities available, the features to which these pulses correspond. The duration of the value "1" of the function from the j-th stage is equal to the times of coincidence of the value "1" of the function from the $(j-1)$-st stage, increased by the memory times of the value "1" for the function in the j-th stage.

4. Input system

Investigations of the properties of human hearing show that «with regard to selective properties, the hearing is similar to bandpass filters with a critical band» [5].

In the proposed system the speech sound from the microphone output is transmitted to a set of bandpass filters. At the filter outputs, the signals are rectified and then integrated (Fig. 2). The value of the time constant of the

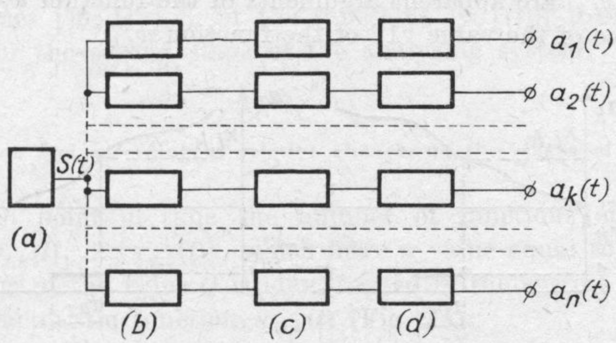

Fig. 2. Diagram of input systems for the analysis of sounds: a) microphone, b) set of bandpass filters, c) rectifying systems, d) integrating systems

integrating system for each filter should not exceed the reciprocal of the mid-frequency of this filter. In this manner n signals are obtained,

$$\{a_k(t)\}_{k\in N}, \qquad N=\{1,2,\ldots,n\},$$

where k is the number of filters. The signals $\{a_k(t)\}_{k\in N}$ are the arguments for the characteristic functions of the features of sound for the first stage of the system's operation.

5. First stage of system: the determination of the value of characteristic functions of the increase and decrease of signals $\{a_k(t)\}_{k \in N}$

The following properties of the signals output from the analyzer are determined at the first stage of the system: the increase and (independently) the decrease of the signals through set threshold values. It is assumed that for each filter there exists a corresponding set of threshold values $\{x_{1,k}, \ldots$ $\ldots, x_{i,k}, \ldots, x_{q,k}\}$, where k is an index of a band filter and i is an index of the threshold value, $k \in N = \{1, 2, \ldots, n\}$, $i \in E = \{1, 2, \ldots, q\}$.

Each of the elements of the set $\{x_{i,k}\}_{k \in N}$ is denoted by the symbol $x_{i,k}$. Let

$$\{w_{i,k}(t)\}_{k \in N} = \{w_{1,k}(t),\ w_{2,k}(t), \ldots, w_{g,k}(t)\}_{k \in N}$$

and

$$\{m_{i,k}(t)\}_{k \in N} = \{m_{1,k}(t), m_{2,k}(t), \ldots, m_{q,k}(t)\}_{k \in N}$$

be families of sets of characteristic functions which transform signals $\{a_k(t)\}_{k \in N}$ and families of sets of threshold values $\{x_{i,k}\}_{k \in N}$ into sets of values $\big\{\{0, 1\}^E\big\}_{k \in N}$.

Let us consider the function $w_{i,k}(t) \in \{w_{i,k}(t)\}_{k \in N}$. The function $w_{i,k}(t)$ is a characteristic function of the increase of the signal $a_k(t)$ in relation to the preset threshold value $x_{i,k}$ (Fig. 3a),

$$w_{i,k}\big(t, a_k(t), x_{i,k}\big) = \begin{cases} 1 & \text{if } a_k(t) \leqslant x_{i,k} \wedge a_k(t+\Delta t) > x_{i,k}, \\ 0 & \text{otherwise,} \end{cases} \tag{5}$$

where $\big(t, a_k(t), x_{i,k}\big)$ are apparent arguments of the function $w_{i,k}$ and moments of the occurrence of the value "1" of the function $w_{i,k}$.

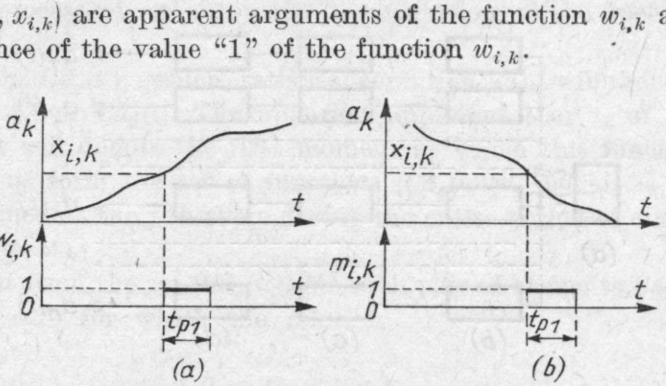

Fig. 3. a) Graphic interpretation of the value "1" of the characteristic function $w_{i,k}(t)$ of the signal increasing $a_k(t)$ through the threshold value $x_{i,k}$, b) physical interpretation of the value "1" of the characteristic function $m_{i,k}(t)$ of the signal decreasing $a_k(t)$ through the threshold value $x_{i,k}$

Let as assume that $m_{i,k}(t) \in \{m_{i,k}(t)\}_{k \in N}$, where $m_{i,k}(t)$ is a characteristic function of the decrease of the signal $a_k(t)$ in relation to the preset threshold value $x_{i,k}$ (Fig. 3b):

$$m_{i,k}[t, a_k(t), x_{i,k}] = \begin{cases} 1 & \text{if } a_k(t) \geqslant x_{i,k} \wedge a_k(t+\Delta t) < x_{i,k}, \\ 0 & \text{otherwise.} \end{cases} \tag{6}$$

Since the human hearing reacts basically to relative change of sound intensities, the values of the thresholds $\{x_{i,k}\}_{k\in N}$ can be distributed in the intervals $\{[x_1, x_q]_k\}_{k\in N}$ according to the logarithmic scale. Such a distribution is justified also by the economics of the system construction (using a smaller number of threshold values).

The values $\{x_{q,1}, x_{q,2}, \ldots, x_{q,k}, \ldots, x_{q,n}\}$ are distributed over a range of corresponding frequencies $\{f_1, f_2, \ldots, f_k, \ldots, f_n\}$, according to the value of the mean speech spectrum.

An essential property of the system is the memorizing of each of the value "1" of the characteristic functions $\{w_{i,k}(t)\}_{k\in N}$ and $\{m_{i,k}(t)\}_{k\in N}$ over short periods of time t_{p1} (Fig. 3), which satisfies the condition

$$t_{p1} \leqslant \tau_{p1}, \tag{7}$$

where τ_{p1} is the smallest duration of an extreme of the sound spectral envelope.

The duration of the maximum of the sound spectral envelope at a frequency f_k is understood as a period during which the output of k-th filter signal exceeds the signals at the $k-1$ and $k+1$ outputs of the filter. More precise determinations of extremes are stated in section 8. The time t_{p1} can be found only by experiment.

The use of the functions $\{w_{i,k}(t)\}_{k\in N}$ and $\{m_{i,k}(t)\}_{k\in N}$ permits further consideration to include only relative changes in the filter output signal stages which satisfy condition (7).

The values $\left\{\{0, 1\}^E\right\}_{k\in N}$ of the functions $\{w_{i,k}(t)\}_{k\in N}$ and $\{m_{i,k}(t)\}_{k\in N}$ are arguments for the second stage of the analyzing system.

6. Detection and recording of relative changes in the stages of analyzed signals

For each point in time the number of functions of the set $\{w_{i,k}(t)\} = \{w_{1,k}(t), w_{2,k}(t), \ldots, w_{q,k}(t)\}$, which have a value equal to unity, is totalled. The exceeding of the value Q is identificated in the system by a value of "1" for the characteristic function $w_{Q,k}(t)$ (Fig. 4a):

$$w_{Q,k}[t, \{w_{i,k}(t)\}, Q] = \begin{cases} 1 & \text{if } \left(\text{card}\left\{w_{i,k}(t) \in \{w_{i,k}(t)\} : w_{i,k}(t) = 1\right\}\right) > Q, \\ 0 & \text{otherwise,} \end{cases} \tag{8}$$

where card $\{\ldots\}$ is the number of elements of the set $\{w_{i,k}(t)\}$, which satisfy the condition $w_{i,k}(t) = 1$.

Similar operations are carried out for the set of functions $\{m_{i,k}(t)\} = \{m_{1,k}(t), m_{2,k}(t), \ldots, m_{q,k}(t)\}$:

$$m_{Q,k}[t, \{m_{i,k}(t)\}, Q] = \begin{cases} 1 & \text{if } \left(\text{card}\left\{m_{i,k}(t) \in \{m_{i,k}(t)\} : m_{i,k}(t) = 1\right\}\right) > Q, \\ 0 & \text{otherwise.} \end{cases} \tag{9}$$

A value of "1" for the function $w_{Q,k}(t)$ expresses an increase of the signal $a_k(t)$ in the time interval $\Delta t \leqslant t_{p1}$ by the successive threshold values, with the number of surpassed thresholds being higher than Q (Fig. 4). In a similar man-

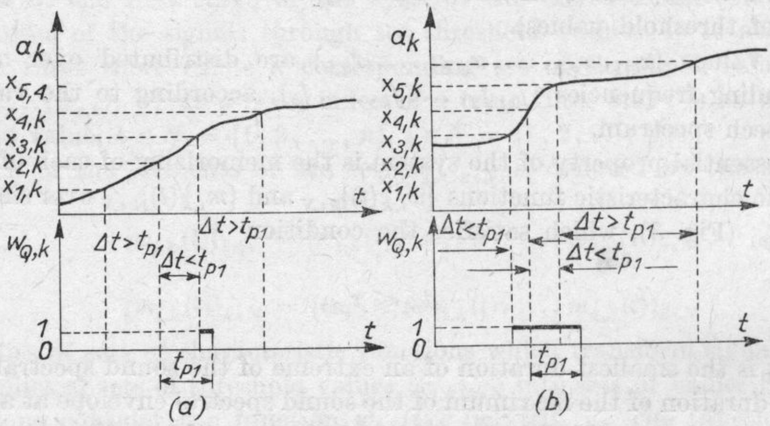

Fig. 4. Physical interpretation of the value "1" of the function $w_{Q,k}(t)$: signals (a) and $(b$ for $Q = 1$ are identified

ner, a value of "1" for the function $m_{Q,k}(t)$ describes the decrease of the signal $a_k(t)$ in the interval Δt, $\Delta t \leqslant t_{p1}$ by more than Q threshold values. The representation of the set of functions $\{w_{i,k}(t)\}$ gives in effect a large reduction of the input information. The use of the threshold value Q permits a determination of the parameters of the speech signal, which is independent of its stage. The subsequent analysis utilizes only relative changes of the stage of signals output by filters.

The longer the time t_{p1} (formula (7)), the smaller the stage changes of the signals $\{a_k(t)\}_{k\in N}$ will be for the values of the functions $\{w_{Q,k}(t)\}_{k\in N}$ and $\{m_{Q,k}(t)\}_{k\in N}$ to assume the value $\{1\}_{k\in N}$. The higher the threshold value Q, the higher the relative increments of signals $\{a_k(t)\}_{k\in N}$ as well as $\{w_{Q,k}(t)\}_{k\in N}$ and $\{m_{Q,k}(t)\}_{k\in N}$ should assume the value $\{1\}_{k\in N}$.

The values $\{0, 1\}_{k\in N}$ of the functions $\{w_{Q,k}(t)\}_{k\in N}$ and $\{m_{Q,k}(t)\}_{k\in N}$ are arguments for the "sharpening" stage of the extremes of the frequency-time envelope of the sound.

7. „Sharpening" of extremes of the frequency-time envelope of the sounds

In order to obtain sufficiently small time constants for the input system it is possible to use bandpass filters with broad, overlapping transmission bands. The entire frequency range to be analyzed can then be covered with a comparatively small number of filters. A single extreme of the sound spectral envelope will then occur at the output of several neighbouring filters.

At this point we will discuss the process of sharpening a single local extreme, independently of whether a formant and its transient will be discovered in the next stages of analysis.

Our system determines automatically at each moment the numbers of filters at the outputs of which the extremes have appeared. The earlier of each two directly adjoining filters is taken. In effect, the local extremes of the spectral envelope are determined.

If an extreme is registered at the output of the k-th filter, followed by a local extreme at the output of the $(k-1)$-st or $(k+1)$-st filter, but not more than a time t_{p3} later (see formula (18)), then the change in the frequency of this extreme will be registered (see item 12). However, if the local extrema occur at directly adjoining (neighbouring) filters after a time interval longer than t_{p3}, these extrema are analyzed separately.

Let us consider the sequence of directly neighbouring filters $k_1, k_2, \ldots, k_m$ at the outputs of which one extreme occurred. In our system the quickest to determine is the value "1", obtained for the function $w_{Q,k}(t)$ amongst all values $\{1\}^M$ of the function $\{w_{Q,k}(t)\}_{k\in M}$, $M = \{k_1, k_2, \ldots, k_m\} \in N$, where $k_1, k_2, \ldots, k_m$ are indices of successive filters at the outputs of which values $\{1\}^M$ of the function $\{w_{Q,k}(t)\}_{k\in M}$ are obtained. The function $w_{Q,k}(t)$, determined in this manner, will be denoted by $w_{r,k}(t)$.

In a similar manner the most quickly determined is value "1", obtained for the function $m_{Q,k}(t)$ amongst all values $\{1\}^M$ of the function $\{m_{Q,k}(t)\}_{k\in M}$. In this case the function $m_{Q,k}(t)$ will be denoted by $m_{r,k}(t)$. Let $g, h \in M, g = h$:

$$w_{Q,g}(t_a), w_{Q,h}(t_b) \in \{C_k(t)\}_{k\in M} \Leftrightarrow 0 < |t_a - t_b| < t_{p2}, \tag{10}$$

$$m_{Q,g}(t_a), m_{Q,h}(t_b) \in \{C_k(t)\}_{k\in M} \Leftrightarrow 0 < |t_a - t_b| < t_{p2}. \tag{11}$$

In the set $\{C_k(t)\}_{k\in M}$ the relation of partial order

$$C_g(t_a) \in\!\!- C_h(t_b) \Leftrightarrow t_a \geqslant t_b \tag{12}$$

is determined.

The relation $\in\!\!-$ arranges the values $\{1\}^M$ of the function $\{w_{Q,k}(t)\}_{k\in M}$ as well as of $\{m_{Q,k}(t)\}_{k\in M}$, that describe one extreme of the frequency-time envelope of the analyzed sequence.

A characteristic property of the system is the memorizing each of the values "1" of the function $\{w_{r,k}(t)\} = \{w_{r,1}(t), w_{r,2}(t), \ldots, w_{r,n}(t)\}$ throughout a short length of time t_{p2}. The length of time t_{p2} is described by the inequality

$$t_{p1} < t_{p2} \leqslant \tau_{p2}, \tag{13}$$

where τ_{p2} is the longest duration of the maxima of the frequency-time envelope of the speech sound (see also the definition of the time τ_{p1}, formula (7)). In analyzing voiced sounds, we obtain the value

$$\tau_{p2} \approx \frac{1}{2\pi f_t}, \tag{14}$$

where f_t corresponds to the lowest frequency of the larynx tone.

If f_t denotes the average value of the frequency of the larynx tone, then an automatic analysis of speech pronounced with a larynx tone frequency smaller than the average will not determine the extremes. Condition (13) results from expression (1).

The duration of the value "1" of the function $w_{r,k}(t)$ is equal to

$$t_{r2} = t_{p2} + \tau_Q, \tag{15}$$

where τ_Q is the time of coincidence of the value "1" of the function $w_{Q,k}(t)$ or $m_{Q,k}(t)$ (see expressions (8) and (5)).

The times of memorizing are not added to the durations of the value "1" of the function

$$\{m_{r,k}(t, m_{Q,k}(t), Q)\} = \{m_{r,1}(t), m_{r,2}(t), \ldots, m_{r,n}(t)\}$$

which are equal to τ_Q.

8. Detection of maxima of the frequency-time envelope of a speech signal

In this paper it is assumed that at the output of the analyzer or, more strictly, at the output of each channel, a rectified signal is obtained with an instant amplitude which changes periodically, according to the frequency of larynx tone (the pulsations with a frequency of the bandpass filter are smoothed out by means of integrating systems with suitable time constants). If in a given channel a formant occurs, then this brings about an increase in the relative changes of the output signal. Generally speaking, the extremes of the signal spectral envelope are determined for each time interval t_{p2} in such a manner that the values of the characteristic functions of the increase $(\{w_{i,k}(t)\})$ and decrease $(\{m_{i,k}(t)\})$ of the signal are determined. The statement that local extremes of the spectral envelope occurred in the k-th channel means that relative changes (increase and decrease) of the signal in this channel were greater and quicker than the corresponding changes in the $(k-1)$-st or $(k+1)$-st channel. However, this assumption makes it possible to overlook the earlier "sharpening" stage of the function $\{m_{r,k}(t)\}$. In general, the characteristic function of the formant of the analyzed sequence is determined in the following manner:

$$wm_k[t, w_{r,k}(t_1), m_{r,k}(t_2)] = \begin{cases} 1 & \text{if } w_{r,k}(t_1) \multimap m_{r,k}(t_2), \\ 0 & \text{otherwise}. \end{cases} \tag{16}$$

An essential property of the system is the storage of each value "1" of the function $\{wm_k(t)\}$ for an instant of time (Fig. 5). The length of time t_{p3} is determined by the inequality

$$t_{p2} < t_{p3} \leqslant \tau_{p3}, \tag{17}$$

where τ_{p3} is somewhat shorter than the duration of the shortest voiced sound.

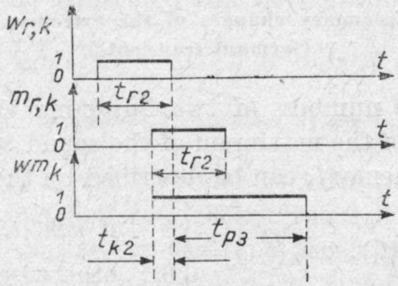

Fig. 5. Graphical interpretation of the function described by formula (16)

The duration of the value "1" of the function $wm_k(t)$ is

$$t_{r3} = t_{p3} + t_{k2}, \tag{18}$$

where t_{k2} is the time of coincidence of the value "1" of the function $w_{r,k}(t)$ (Fig. 5).

9. Determination of the position in the spectrum of formants of voiced sounds

The function of the voiced formant $\varphi_k(t)$ is described by formula

$$\varphi_k[t, wm_k(t), t_{r3}, c, t_{p3}] = \begin{cases} 1 & \text{if } t_{r3} > ct_{p3}, \\ 0 & \text{otherwise}, \end{cases} \tag{19}$$

where c is a constant, chosen experimentally. The value of the constant c should vary within a range from 1 to 3.

From equation (15) it results that in order to detect a formant of frequency f_k, the maxima of the spectral envelope have to occur in the k-th filter several times with the frequency of larynx tone (see expressions (17), (13) and (1)).

The values "1" of the function $\{\varphi_k(t)\} = \{\varphi_1(t), \varphi_2(t), \ldots, \varphi_n(t)\}$ are memorized in the system over period of time t_{p4}.

The value of t_{p4} can be described by the inequality

$$t_{p3} < t_{p4} \leqslant \tau_{p4}, \tag{20}$$

where τ_{p4} equals at least twice the duration of an average sound. Such an assumption is accepted in view of the need to ensure the coincidence of pulses when measuring formant transients (see item 11).

The duration of the value "1" of the function $\varphi_k(t)$ or of the function $a_k(t)$ is

$$t_{r4} = t_{p4} + t_{k3}, \tag{21}$$

where t_{k3} is the time of coincidence of the value "1" of the function $wm_k(t)$ or of the function $mw_k(t)$.

10. Measurements of frequency changes of the extremes of the sound envelope (formant transients)

Let g and h be the numbers of two adjoining channels of the system; $g, h \in N$. The transition of the maximum of the sound signal envelope from the frequency f_g to the frequency f_h can be described by the expression

$$M_{g,h}[t, wm_g(t_1), wm_h(t_2)] = \begin{cases} 1 & \text{if } wm_g(t_1) \in wm_h(t_2), \\ 0 & \text{otherwise}. \end{cases} \tag{22}$$

According to expression (22), the characteristic function $M_{g,h}(t)$ takes the value "1" when a maximum of the envelope in the channel g, and then a maximum of the envelope in the neighbouring channel h, there are identified in succession: in the time interval t_{p3}.

We notice that conditions for the increase and decrease of signals are satisfied even for quick changes of the frequency of formants. However, only the changes slower than the time constants of bandpass filters (see item 4) are detected, i.e. we must have

$$|t_1 - t_2| > \tau_k, \tag{23}$$

where τ_k is the time constant of the filter of the input system.

To satisfy inequality (23) it is necessary to use filters with suitably broad bands.

11. Detection of formant transients of voiced sounds

Let g and h be the voiced numbers of two neighbouring channels of the system $g, h \in N$. The transient formant beginning with a frequency f_g and ending up with a frequency f_h can be detected by means of the characteristic function $Y_{g,h}(t)$:

$$Y_{g,h}[t, \varphi_h(t_2), wm_g(t_1)] = \begin{cases} 1 & \text{if } \varphi_h(t_2) \in wm_g(t_1), \\ 0 & \text{otherwise}. \end{cases} \tag{24}$$

The formant transient, beginning with a frequency f_h and ending up with a frequency f_g, is described by the characteristic function $T_{g,h}(t)$:

$$T_{g,h}[t, wm_g(t_2), \varphi_h(t_1)] = \begin{cases} 1 & \text{if } wm_q(t_2) \in \varphi_h(t_1), \\ 0 & \text{otherwise}. \end{cases} \tag{25}$$

The value "1" of the characteristic functions

$$\{Y_{g,h}(t)\} = \{Y_{1,2}(t), Y_{2,3}(t), \ldots, Y_{n-1,n}(t_1)\}$$

and

$$\{T_{g,h}(t)\} = \{T_{1,2}(t), T_{2,3}(t), \ldots, T_{n-1,n}(t_1)\}$$

are memorized in the system for a short period of time t_{p4} (see inequality (20)). Current frequencies of the transients can be determined from relations (30) and (31).

12. Determination of current frequencies of the formant transients

The considerations of sections 12 and 13 refer to the case where, in the frequency range of two neighbouring filters, the k-th and $(k+1)$-st, there is at most one maximum of the spectral envelope, the energy of which considerably exceeds the energy of other maxima within the range of these two filters. We assume that the bands of the k-th and $(k+1)$-st filters overlap.

The solution presented hitherto permits determination of positions of the transients with an accuracy given by the interval $f_{k+1}-f_k$. A method of determinating current frequencies of the transients with an accuracy of $(f_{k+1}-f_k)/q$ (with the assumption of a linear distribution of threshold values $\{x_{1,k}, x_{2,k}, \ldots, x_{q,k}\}$ in the interval $[x_{1,k}, x_{q,k}]$) will now be presented.

We denote the known coefficients of the attenuation of the bandpass filter of channels k and $k+1$ by $B_k(f)$ and $B_{k+1}(f)$, respectively. For values of the signal $A(t,f)$ with a frequency f, we obtain the following relations:

$$a_k(t) = B_k(f)A(t,f), \tag{26}$$

$$a_{k+1}(t) = B_{k+1}(f)A(t,f). \tag{27}$$

Let us divide the members of equation (26) by equation (27):

$$\frac{a_k(t)}{a_{k+1}(t)} = \frac{B_k(f)}{B_{k+1}(f)} = B(f). \tag{28}$$

For the current frequency of the signal $A(t,f)$ we obtain the following expression:

$$f = B^{-1}\left[\frac{a_k(t)}{a_{k+1}(t)}\right]. \tag{29}$$

Values of the signals $\{a_k(t)\}_{k\in N}$ are also represented in the system by values of elements of the sets

$$\big\{w_{i,k}(t) \in \{w_{i,k}(t)\} : w_{i,k}(t) = 1\big\}_{k\in N}$$

and

$$\big\{m_{i,k}(t) \in \{m_{i,k}(t)\} : m_{i,k}(t) = 1\big\}_{k\in N},$$

while values of the signals $\{a_{k+1}(t)\}_{k\in N}$ are represented by values of elements of the sets

$$\{w_{i,k+1}(t) \in \{w_{i,k+1}(t)\} : w_{i,k+1}(t) = 1\}_{k\in N}$$

and

$$\{m_{i,k+1}(t) \in \{m_{i,k+1}(t)\} : m_{i,k+1}(t) = 1\}_{k\in N}.$$

The current frequency of the maximum that is shifting from the frequency f_k to the frequency f_{k+1} is determined by the following expression:

$$w_f = B^{-1}\left(\frac{\operatorname{card}\{m_{i,k}(t) \in \{m_{i,k}(t)\} : m_{i,k}(t) = 1\}}{\operatorname{card}\{w_{i,k+1}(t) \in \{w_{i,k+1}(t)\} : w_{i,k+1}(t) = 1\}}\right), \tag{30}$$

while the current frequency of the maximum that is shifted from the frequency f_{k+1} to the frequency f_k is determined by the relation

$$m_f = B^{-1}\left(\frac{\operatorname{card}\{w_{i,k}(t) \in \{w_{i,k}(t)\} : w_{i,k}(t) = 1\}}{\operatorname{card}\{m_{i,k+1}(t) \in \{m_{i,k+1}(t)\} : m_{i,k+1}(t) = 1\}}\right). \tag{31}$$

13. Determination of initial frequencies and of final transients

The frequency at which the transient begins to shift in the direction of smaller frequencies is evaluated by means of the expression

$$LM \approx B^{-1}\left(\frac{\min\limits_i \{w_{i,k}(t)\}}{\max\limits_i \{m_{i,k+1}(t)\}}\right), \tag{32a}$$

where

$$\min_i \{w_{i-1,k}(t)\} = \sim w_{i,k}(t_1) \mathrel{\reflectbox{$\exists$}} w_{i,k}(t_2) \mathrel{\reflectbox{$\exists$}} T_{k+1,k}(t_3), \tag{32b}$$

$$\max_i \{m_{i,k+1}(t)\} = \sim m_{i+1,k+1}(t_1) \mathrel{\reflectbox{$\exists$}} m_{i,k+1}(t_2) \mathrel{\reflectbox{$\exists$}} T_{k+1,k}(t). \tag{32c}$$

From (32b) it results that the threshold value $x_{i,k}$ has been exceeded by the signal a_k at the moment t_2 and the threshold value $x_{i-1,k}$ has not been exceeded by the signal a_k in the time interval $[t_2 - t_{pi}, t_2]$, and the formant transient that shifts from the filter $k+1$ towards the filter k is recorded in the time interval $[t_2, t_2 + t_{pi}]$.

The initial frequency of the transient which shifts towards higher frequencies can be determined from the relation

$$LW \approx B^{-1}\left(\frac{\max\limits_i \{m_{i,k}(t)\}}{\min\limits_i \{w_{i,k+1}(t)\}}\right), \tag{33a}$$

where

$$\max_i \{m_{i,k}(t)\} = \sim m_{i+1,k}(t_1) \ni m_{i,k}(t_2) \ni T_{k,k+1}(t_3), \tag{33b}$$

$$\min_i \{w_{i,k+1}(t)\} = \sim w_{i-1,k+1}(t_1) \ni w_{i,k+1}(t_2) \ni T_{k,k+1}(t_3). \tag{33c}$$

The frequency at which the transient shifting towards the higher frequencies finishes is determined by means of the expression

$$WL \approx B^{-1}\left(\frac{\min_i \{m_{i,k}(t)\}}{\max_i \{w_{i,k+1}(t)\}}\right), \tag{34a}$$

where

$$\min_i \{m_{i,k}(t)\} = \sim m_{i-1,k}(t_3) \in m_{i,k}(t_2) \in Y_{k,k+1}(t_1), \tag{34b}$$

$$\max_i \{w_{i,k+1}(t)\} = \sim w_{i+1,k+1}(t_3) \in w_{i,k+1}(t_2) \in Y_{k,k+1}(t_1). \tag{34c}$$

The frequency at which the transient shifting towards decreasing frequencies finishes can be calculated from the relation

$$ML \approx B^{-1}\left(\frac{\max_i \{w_{i,k}(t)\}}{\min_i \{m_{i,k+1}(t)\}}\right), \tag{35a}$$

where

$$\max_i \{w_{i,k}(t)\} = \sim w_{i+1,k}(t_3) \in w_{i,k}(t_2) \in Y_{k+1,k}(t_1), \tag{35b}$$

$$\min_i \{m_{i,k+1}(t)\} = \sim m_{i-1,k+1}(t_3) \in m_{i,k+1}(t_2) \in Y_{k+1,k}(t_1). \tag{35c}$$

14. Modifications of times of the instant memory

For the correct operation of the system, the constancy of times t_{pj} at the j-th stage of the identification for all channels is very important:

$$\forall_j \forall_k (t_{pj} = \text{const}). \tag{36}$$

In order to give consideration to the technical problems connected with the satisfaction of condition (36) and also to the construction of band filters with identical time constants, it is possible to modify inequality (3) to the condition

$$t_{p0} < |t_a - t_b| < t_{pj}, \tag{37}$$

where t_{p0}, chosen experimentally, permits consideration to be given to the above-mentioned difficulties.

If condition (36) is unsatisfied, then it causes that the frequency changes of the extremes are detected despite their absence in the real sound signal.

15. Advantages of the system

The use of instant memory at each stage of the identification permits analysis of sounds in the real time, and also permits automatic segmentation of sounds in the time domain. This segmentation consists in the division of the sound sequences analyzed into uneven lengths of time, with lengths not shorter than the times $\{t_{pj}\}$. The normalization of lengths of the time $\{t_{pj}\}$ up to magnitudes that correspond to the extreme values of the occurrence of certain features in the analyzed sequences permits the results of the identification of the speech to be made independent of the speeking speed, the frequency of the larynx tone, etc. The use of suitable times $\{t_{pj}\}$ at various stages of the system allows for a careful consideration of physical properties of speech sounds and the determination of basic parameters of speech signals amongst external disturbances and excess information. This selection takes place at each of the $\{j\}$ stages of the system.

In the system it is possible to use bandpass filters with comparatively broad and overlapping transmission bands. Owing to this it is possible to obtain filters with sufficiently small time constants.

The system proposed permits the determination, at any moment, of local times of the extremes of the speech signal (section 7). Their number can theoretically reach the value $n/2+1$.

The system permits a determination of parameters of the speech signal independent of the absolute stage of the speech signal (section 6), e.g. an increase of the signals $\{a_k(t)\}_{t\in N}$ from various initial values, and a decrease of these signals to various final values.

The pulsations of signals with the frequency of the larynx tone at the output of a set of filters are utilized in the system to determine the feature of the voiced ability of the analyzed signals (sections 9 and 11). The effects of phase incompatibility at the outputs of the filters are eliminated by the application of an instant memory (memory times $\{t_{pj}\}$).

Initial and final frequencies of the transients (section 13) should provide the basis for hypotheses about the position of locuses through the system for the identification of sounds (block (*c*) in Fig. 1).

The description of the system by means of characteristic functions of the features of sounds permits the system to be designed and built directly with the aid of integrated circuits [6].

The individual definition of the function of increase and decrease permits the automatic determination of directions of the course of transients in the frequency domain, the initial or final routes of transients, the detection of periodicity (soundability) and also a description of duration of these features of speech sounds.

In the case of an absence of periodicity the values of the function $\varphi_k(t)$ are equal to zero for each channel.

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ABSORPTION OF ULTRASONIC WAVES IN $ZnCl_2$ SOLUTIONS IN METHANOL

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The investigation of the absorption of ultrasonic waves in $ZnCl_2$ solutions in methanol has shown the occurrence of a relaxation process which is considered, in this paper, to be the disintegration and the formation of ion pairs. The activation energy has been determined on the basis of the temperature dependence of the relaxation time of this process, while reaction rate constants have been established from the concentration dependence of the relaxation time. In the solutions investigated a concentration of 2.0 mol/l is a characteristic concentration, above which the structure of the solution becomes considerably stabilized.

1. Introduction

Zinc chloride solutions in methanol has a number of interesting physical properties which distinguish them from other electrolyte solutions. Results obtained hitherto indicate that zinc chloride occurs in the methanol solution in configurations determined by a series of equilibriums:

$$Zn^{2+} + 2\,Cl^- \rightleftharpoons ZnCl_2, \tag{1}$$

$$2\,ZnCl_2 \rightleftharpoons Zn^{2+} + ZnCl_4^{2-}, \tag{2}$$

$$n\,ZnCl_4^{2-} + n\,Zn^{2+} \rightleftharpoons 2n\,ZnCl_2 \rightleftharpoons (ZnCl_2)_{2n}. \tag{3}$$

$ZnCl_2$ can also occur in the form of tetra and octahedral complexes which coordinate methanol molecules as ligands. The equilibrium described by equation (1) occurs in solutions with very small concentrations ($k < 0.1$ mol/l). In solutions with higher concentrations the predominate configurations are described by equations (2) and (3). Investigations of Raman spectra have shown that zinc chloride molecules in a methanol solution combine into "polymer" chains in which the number of molecules varies within broad limits. However, Zn^{2+} ions and $ZnCl_4^{2-}$ complex ions cannot exist only in the form of free ions since the Bjerrums critical distance for the formation of ion pairs is exceeded throughout the whole volume of the solution at concentrations above 0.11 mol/l at a temperature of 25°C. Thus in addition to equations (1) to (3), the equation describing the equilibrium of the decomposition and formation of ion pairs should also be considered:

$$Zn^{2+} \times ZnCl_4^{2-} \rightleftharpoons Zn^{2+} + ZnCl_4^{2-}. \tag{4}$$

The investigation of the absorption of ultrasonic waves in solutions can supply a valuable information on the kinetics of the reaction and activation energy of processes encountered.

It follows from classical STOKES-KIRCHHOFF theory that the absorption coefficient of ultrasonic waves in a liquid is proportional to the square of the wave frequency, the medium viscosity and the thermal conductivity. Hence, the important conclusion can be drawn that in a liquid in which relaxation processes do not occur, the ratio of the coefficient of sound absorption to the square of the frequency is constant, i.e.

$$\frac{\alpha_{\text{classical}}}{f^2} = \text{const}. \tag{5}$$

When a relaxation process occurs in a liquid, relation (5) is not satisfied and must take the form

$$\frac{\alpha}{f^2} = \frac{A}{1+(f/f_r)^2} + B, \tag{6}$$

where f_r denotes a relaxation frequency, and A and B are constants.

The constant A involves an information about the contribution of the relaxation process to the absorption, whereas the constant B describes the

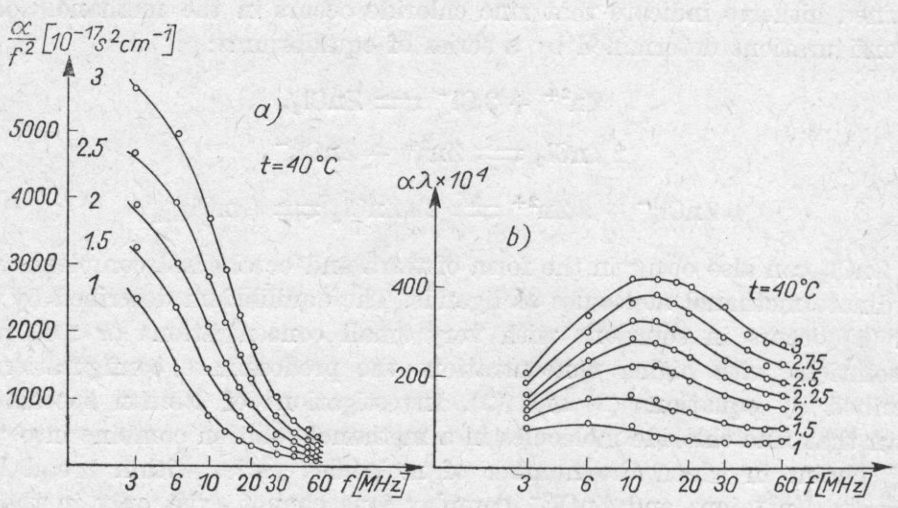

Fig. 1. α/f^2 and $\alpha\lambda$ as functions of frequency in solution of $ZnCl_2$ in methanol; $t = 40°C$

absorption of ultrasonic waves caused by the viscosity, heat conductivity, and relaxation processes whose relaxation times are considerably shorter than the relaxation time of the process under consideration.

It is a good practice to consider the measurement results of the absorption of ultrasonic waves in the forms of the relation α/f^2 versus frequency (for $f = f_r$ there is a turning point in the graph) and the relationship $\alpha\lambda$ versus frequency (for $f = f_r$ the graph has a maximum).

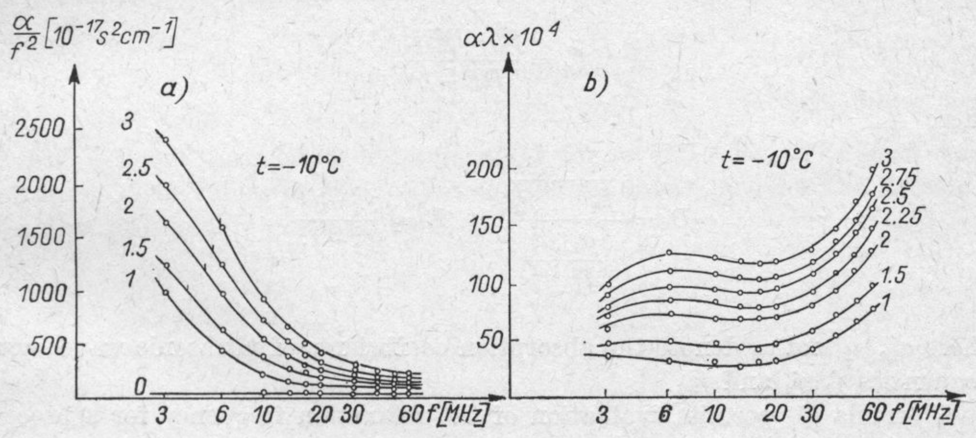

Fig. 2. α/f^2 and $\alpha\lambda$ as functions of frequency in solution of $ZnCl_2$ in methanol; $t = -10°C$

2. Method and experimental results

The solutions investigated were prepared with the concentration ranging from 0.1 to 3.0 mol/l using carefully dehydrated methanol and zinc chloride. The maximum error of the concentration determination was 75×10^{-4} mol/l.

The measurements of the absorption coefficient of ultrasonic waves in the solutions were made using an ultrasonic pulse-phase interferometer UI-13 and a high-frequency ultrasonic unit US-4 (maker: IPPT PAN, Warsaw) in the frequency range from 3 MHz to 60 MHz. The relative error of the determination of the absorption coefficient of ultrasonic waves varies from 6 % for $f = 3$ MHz to 1 % for $f = 60$ MHz.

The systems used for thermostatic control and temperature measurement permit the determination of the temperature of the investigated solution to an accuracy of ± 0.1 deg.

The measuring vessels together with probes were carefully sealed in order to prevent evaporation of the solvent and consequent changes in the concentration of the investigated solutions.

The results of measurements of the absorption coefficient of ultrasonic waves were analyzed as α/f^2 and $\alpha\lambda$ as functions of frequency. From this analysis it can be concluded that within the range of ultrasonic waves, concentrations and temperatures used molecular relaxation processes are seen to contribute to the sound absorption.

To determine exactly the relaxation frequency, the advantage has been taken of the following method. The relation α/f^2 versus frequency can be presented by formula (6).

Since an exact determination of the constants A and B has been impossible within the applied frequency range, equation (6) is transformed to take the form

$$f_r^2 = \frac{f_3^2 - DEf_2^2}{DE - 1}, \tag{7}$$

where

$$D = \frac{\dfrac{\alpha_1}{f_1^2} - \dfrac{\alpha_2}{f_2^2}}{\dfrac{\alpha_1}{f_1^2}\dfrac{\alpha_3^2}{f_3^2}}, \quad E = \frac{f_3^2 - f_1^2}{f_2^2 - f_1^2},$$

where α_1, α_2 and α_3 denote the absorption coefficients of ultrasonic waves with frequencies f_1, f_2 and f_3.

Formula (7) permits evaluation of the relaxation frequency for a process with a discrete time of relaxation on the basis of the three values of α/f^2 determined experimentally.

For the temperatures and concentrations established ten measurements of α/f^2 at ten frequencies were made. The number of combinations without the repetition of 10 elements – 3 elements each – equals 120. The total number of 120 three-element combinations has been reduced to 56 by the elimination of combinations containing the measurement points for neighbouring measurement frequencies, i.e. only the threes of non-neighbouring measurement points have been considered. The relaxation frequency for each of three-element combinations of values of α/f^2 was evaluated, and the obtained results averaged to obtain the value of the relaxation frequency $\bar{f}_r$. For this value of $\bar{f}_r$, relations using the measurement points were plotted againts frequency. The results of this analysis are presented for several selected temperatures and solution concentrations in Figs. 1 and 2 in the forms of α/f^2 and $\alpha\lambda$ against frequency.

3. Analysis of experimental results

The relaxation frequencies determined in the investigations increase with increasing temperature and solution concentration. One such relation indicates that the detected relaxation process is a thermally activated process of chemical relaxation.

The time of relaxation of a thermally activated process obeys Arrhenius' law

$$\tau = \tau_0 \exp \frac{E}{RT}, \tag{8}$$

where E denotes the activation energy of the process, R is the gas constant, T – the absolute temperature, and τ_0 – a constant.

The activation energy of the process can be determined from the slope of the straight line representing the relation $\ln \tau$ versus $1/T$, resulting from the relation

$$\ln \tau = \ln \tau_0 + \frac{E}{RT} = \text{const} + \frac{E}{R} \frac{1}{T}. \tag{9}$$

Fig. 3 shows the relation $\ln \tau$ versus $1/T$ for several selected solution concentrations. The values of the activation energy, determined for the detected relaxation process, are given in Table 1.

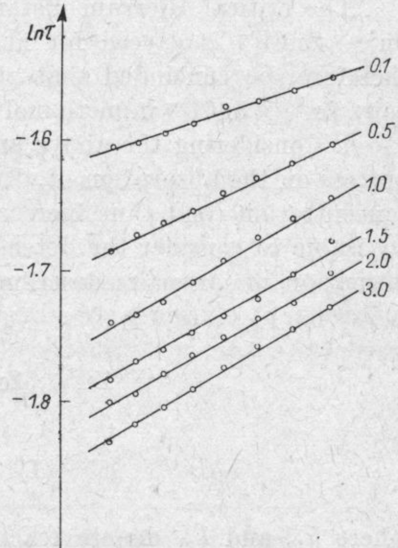

Fig. 3. $\ln \tau$ versus T^{-1} for several selected solution concentrations of $ZnCl_2$ in methanol

Previous investigations of solutions of zinc chloride in methanol [3, 7-10] indicate that zinc chloride occurs in the solution primarily in the form of complex ions $ZnCl_4^{2-}$ and zinc ions Zn^{2+}. The zinc ion is distinguished by a very large surface density of electric charge and a small radius ($r = 0.74$ Å). These imply its strong interaction with the molecules of the solvent and other ions in the solution. $ZnCl_4^{2-}$ complexes are large and their surface charge density is considerably smaller than that of Zn^{2+} ions and they exhibit very great longevity.

Table 1. Values of the activation energy E and its dependence on the solution concentration

$k\left[\frac{\text{mol}}{\text{l}}\right]$	0.10	0.20	0.30	0.40	0.50	0.60
$E\left[\frac{\text{kcal}}{\text{mol}}\right]$	1.15	1.45	1.68	1.86	1.99	2.06
$k\left[\frac{\text{mol}}{\text{l}}\right]$	0.70	0.80	1.00	1.25	1.50	1.75
$E\left[\frac{\text{kcal}}{\text{mol}}\right]$	2.12	2.17	2.24	2.32	2.38	2.43
$k\left[\frac{\text{mol}}{\text{l}}\right]$	2.00	2.25	2.50	2.75	3.00	
$E\left[\frac{\text{kcal}}{\text{mol}}\right]$	2.47	2.49	2.50	2.51	2.52	

The critical Bjerrum distance for the formation of ion pairs of the type $Zn^{2+} \times ZnCl_4^{2-}$ is exceed for nearly all of the solutions investigated. It can, therefore, be concluded that zinc chloride occurs mainly in the form of ion pairs $Zn^{2+} \times ZnCl_4^{2-}$ in methanol solutions of the range of concentrations investigated. Considering the above and the fact that the influence of the relaxation process on the absorption of ultrasonic waves increases with increasing solution concentration (and thus increasing concentration of ion pairs), it seems most advisable to consider the detected relaxation process as the disintegration and formation of ion pairs described by the equilibrium

$$\underset{(1)}{Zn^{2+} \times ZnCl_4^{2-}} \underset{k_{21}}{\overset{k_{12}}{\rightleftharpoons}} \underset{(2)}{Zn^{2+} + ZnCl_4^{2-}},$$

where k_{12} and k_{21} denote reaction rate constants, condition (1) denoting the ion pair and condition (2) free ions.

In view of the lack of data for the dissociation constant, the activity coefficients and other quantities describing the solution, the relation between the relaxation time and the solution concentration can be presented in the form

$$\frac{1}{\tau} = k'_{12} + k'_{21}\, k, \tag{10}$$

where k'_{12} and k'_{21} denote apparent reaction rate constants, k is the solution concentration and τ — the relaxation time.

The relaxation time is related to the relaxation frequency by the simple relation

$$f_r = \frac{1}{2\pi\tau}. \tag{11}$$

The apparent reaction rate constants can thus be determined from a graph presenting the relation between the relaxation frequency and the solution concentration. This relation is presented for three selected temperatures in Fig. 4. It is possible to distinguish two concentration ranges for which the

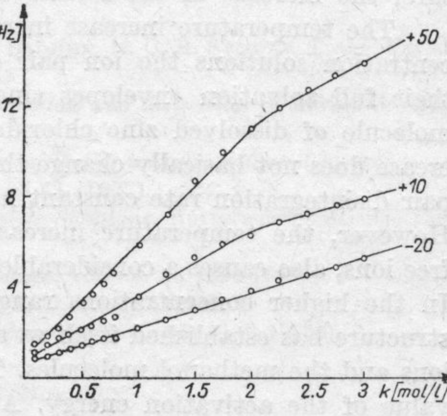

Fig. 4. Relaxation frequency as a function of solution concentration for $ZnCl_2$ in methanol

reaction rate constants differ considerably: low concentrations, ranging from 0.1 to about 2.0 mol/l, and high concentrations, ranging from 2.0 to 3.0 mol/l. The value of the apparent reaction rates for two concentration ranges are given in Table 2.

Table 2. Apparent constant reaction rates as a function of temperature

t [°C]	Lower concentration range		Higher concentration range	
	k'_{12} [$10^6\,s^{-1}$]	k'_{21} $\left[10^6 \frac{1}{\text{mol s}}\right]$	k'_{12} [$10^6\,s^{-1}$]	k'_{21} $\left[10^6 \frac{1}{\text{mol s}}\right]$
−20	5.0	9.4	5.0	9.4
−10	5.0	12.5	13.2	9.8
0	5.0	14.4	18.8	11.7
10	5.0	18.2	19.7	12.1
20	5.0	21.0	20.1	13.2
30	5.0	24.8	22.0	16.6
40	5.0	28.2	30.1	18.0
50	5.0	32.3	31.4	19.2

The increased solution concentration causes increased interaction between the ions and the molecules of the solvent. The concentration of about 2.0 mol/l is a characteristic solution concentration for zinc chloride in methanol. As the concentration increases to reach this value, the activation energy of the process increases quite rapidly, but changes insignificantly with further increase of the concentration above this value. In the lower concentration range, the disintegration rate constant of ion pairs is independent of the temperature, while the recombination rate constant of ion pairs increases with increasing temperature. In the higher concentration range both the disintegration rate constant and the recombination rate constant increase with increasing solution temperature, the increase of the former being greater.

The temperature increase increases the mobility of free ions. In low concentration solutions the ion pair disintegrates into ions which easily obtain their full solvation envelopes since the number of methanol molecules per molecule of dissolved zinc chloride is sufficiently large. The temperature increase does not basically change this situation in the solution and thus the ion pair disintegration rate constant has a value independent of the temperature. However, the temperature increase, while causing increased mobility of the free ions, also causes a considerable increase of the recombination rate constant. In the higher concentrations range, above 2.0 mol/l, a fairly stable solution structure has established itself as a result of increased interaction between the ions and the methanol molecules. This finds its expression in a little changing value of the activation energy. A strong interaction and structural stability imply that, as the temperature increases, the rate of the establishment of a new equilibrium in the solution increases, bringing about an increase in the disintegration rate constant and in the formation of ion pairs in the solution.

4. Conclusions

Investigations of the absorption of ultrasonic waves in methanol solutions of zinc chloride have shown the considerable effect of a molecular relaxation process on the sound absorption. On the basis of the results of other methods of investigating these solutions, it can be concluded that one such process is that of the disintegration and formation of ion pairs, the components of which are zinc ions Zn^{2+} and complex ions $ZnCl_4^{2-}$, described by the equilibrium

$$Zn^{2+} \times ZnCl_4^{2-} \rightleftharpoons Zn^{2+} + ZnCl_4^{2-}.$$

The activation energy of this process increases considerably as the solution concentration increases from 0.1 to about 2.0 mol/l, increasing insignificantly with further concentration increase. At a concentration of about 2.0 mol/l there also occur changes in the value and the tendency to increase of the reaction rate constant. It can be concluded that the structure of methanol solutions of $ZnCl_2$ undergoes considerable stabilization above a concentration of 2.0 mol/l.

The temperature dependence of the relaxation time is strong evidence that the said process is activated thermally. The high absorption of ultrasonic waves, resulting from this process, suggests that it is primarily caused by ion pairs in the solution.

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C H R O N I C L E

ANALYSIS AND EVALUATION OF STANDARDIZATION IN ACOUSTICS IN POLAND

Industrial development and the consequent intensification of the process of urbanization presents an irreversible menace to man's environment, his work-place, his home and his rest. Amongst the numerous hazards for man, the Polish law and national industrial health service include excessive noise and mechanical vibrations which occur in the environment.

Noise and vibration are factors which adversely affect the comfort, the physical and psychological health of man, and the quality and efficiency of his work. The sensitivity to noise and vibration is dependent on the one hand on physical features such as its level, spectral composition, and duration and on the other hand on the degree of man's sensitivity to noise and vibration. This depends not only on his nervous and psychological resistance, and the temporary state of his mind, but also, and primarily, on the nature of his activities, and the places and condition in and under which they are carried out. A number of legal acts have been issued in Poland aimed at reducing the potential of noise and vibration. They include, among others:

(a) the Ordinance of the Cabinet Council of Ministers of August 21, 1959 on general hygienic and sanitary conditions in newly built or reconstructed industrial plants (Official Law Gazette No. 53 item 316) which reads that the permissible noise intensity in working rooms should not exceed the level detailed in obligatory standards viz:

– in design offices – 40 dB,
– in precision workshops – 50 dB,
– in noisy factory halls – 90 dB,

(b) the ordinance of Minister of Labour, Wages and Social Welfare of March 17, 1976 concerning the highest permissible concentrations and intensities of agents harmful for health, in work establishments (Off. Law Gaz. No. 13 item 77); which reads that at work-places in factory halls, pits and in the open air the maximum permissible noise intensity should be 90 dB (A).

In addition the Council of Ministers issued a resolution in 1971 which formulated guidelines for increased noise and vibration control in work establishments. A number of other executive instruments have also been published.

The Council of Ministers is in the process of issuing a new resolution to supersede the previous resolution. While retaining its most important decisions, it will introduce a number of new aspects to cover those areas so far not dealt with by the existing resolution, particularly concerning noise in cities, towns and settlements, and in housing etc.

Standardization and metrological work in acoustics is carried out in accordance with the schedule laid down by the Polish Committee of Standardization and Measures.

The problems covered by the program between 1972 - 1975 were implemented by the inclusion of particulartopics in the yearly plans of the ministries.

The program covered, among other topics, the following:

(a) in standardization:

– the establishment of laboratory and practical measurement methods for testing the acoustic and vibrating properties of machines and equipment,

– the delineation of permissible levels of noise and vibration for particular types of machines and equipment, transport and construction equipment together with measurement procedures,

— the establishment of a basic classification of the types, quality requirements, and methods of investigation of the properties of sound-absorbing, sound-proofing and antivibration materials elements and systems;

— the determination of methods for the calibration of devices for the measurement of noise and vibration;

— the establishment of quality requirements and methods of investigating the efficiency of devices for protecting the individual against noise and vibration;

(b) in metrology:

— the organization in selected centres of measurement laboratories authorized to calibrate measuring and testing devices,

— the selection and organization together with the respective ministries of units entitled to test machines, equipment and tools, including transport and construction equipment, for noise and vibration.

The quantitive state of standards in acoustics on December 31, 1976 was as follows:

— 21 Polish standards (PN),

— 19 branch standards (BN),

— 6 drafts of Polish standards for experimental use,

— 6 analytical and research papers.

In addition to basic standards dealing specifically with acoustics many standard specifications PN and BN defining the requirements for and testing of particular machines and equipment give consideration to the problem of noise measurement, and also to its permissible level. Thus, for example all acoustical problems related to earth-working machines produced by the Construction Machinery Union are dealt with in internal standards.

A detailed list of issued Polish standards and branch standards concerning acoustics and vibration is given in the table below. In addition, the following drafts of standards have been submitted for issuance:

— methods of measurement and the evaluation of noise at workplaces (PN),

— permissible noise level for electrically driven hand tools (PN),

— permissible level of noise and vibrations in telecommunication rooms. Requirements and methods of investigation (BN),

— permissible noise level in post-office rooms. Requirements and methods of investigation (BN).

Although the state of standardization, in principle, meets the actual requirements, it needs supplementing with standards concerning: further basic problems, the method of measurement of a noise source in operating conditions, and the problems of the measurement of and the criteria for the hazards presented by noise and vibration.

These problems are included in the «basic plan of standardization and metrological work for the years 1976 - 1980». The following investigations will be carried out in the part related to the control of noise and vibration:

— the establishment of general measurement methods for noise infrasound, ultrasound and vibration in places of work;

— the construction and testing of meters for measuring sound levels, including octave filters, and also of devices for measuring vibration infra- and ultrasound;

— the determination of permissible noise levels for power boiler equipment, for automobile engines, for agricultural tractors, for machines for the timber industry and the textile industry, for aircraft (at airports and in their neighbourhood and on flight paths at different flight altitudes) and in post office rooms.

— the establishment of methods for measuring the sound power level of boiler equipment, combustion engines (inside and outside the automobile cabin), industrial pumps, refrigerating compressors, machines for the timber industry, mine fans and power-operated tools.

The plan will be realized by the inclusion of particular topics in the yearly plans of the respective ministerial departments.

List of issued Polish Standards and branch standards concerning noise and vibration as of December 31, 1976

Ser. No	Standard No	Title
1	2	3
		A. Polish Standards
1	PN-70 B-02151	Building acoustics. Soundproof protection for rooms in buildings.
2	PN-61 B-02153	Building acoustics. Terminology and definitions.
3	PN-68 B-02154	Building acoustics. Tests on acoustics properties in building partitions.
4	PN-73 E-04255	Electrical rotating machines. Measurement of vibrations.
5	PN-76 E-04072	Transformers. Determination of parameters of the noise.
6	PN-72 E-04257	Electrical rotating machinery. Determination of acoustic parameters of noise.
7	PN-72 E-06019	Electrical rotating machinery. Admissible sound level.
8	PN-73 E-06020	Rotating electric machines. Vibration limits.
9	PN-75 E-06260	Appliances for domestic and similar purposes. Noise level. Examinations and principles of fixing of admissible level.
10	PN-75 M-35200	Admissible sound levels in rooms with energetic objects.
11	PN-72 M-43120	Fans. Methods of noise determination.
12	PN-75 M-47015	Earth moving machinery. Operator's stand. Admissible noise level and methods of tests.
13	PN-75 M-53527	Instruments for mechanical vibration measurements. Terms and definitions.
14	PN/M-55725	Machine tools for metal. Test methods and admissible noise levels (Polish Draft Standard for experimental application).
15	PN-75 M-78030	Driven carriageway cars. Admissible noise level and methods of tests.
16	PN-71 N-01300	Noise of machines and equipment. Methods for determination of acoustic parameters.

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1	2	3
17	PN/N-01301	Acoustics. Measuring frequencies (Polish Draft Standard for experimental application).
18	PN/N-01302	Acoustics. Method of determination of the risk of weakened hearing (Polish Draft Standard for experimental application).
19	PN/N-01303	Acoustics. Method of determination of audibility and audibility levels (Polish Draft Standard for experimental application).
20	PN/N-01305	Acoustics. Plotting curves of equal sound levels and threshold of audibility of normal tones (Polish Draft Standard for experimental application).
21	PN-76 / N-01309	Ear protectors. Method of determination of attenuation and dB (A) reduction.
22	PN-75 / 0-79166	Transport packages. Methods of vibration tests.
23	PN-76 / R-36125	Agricultural tractors and machinery. Noise level at the operator's workplace. Measurement method.
24	PN-71 / S-04051	Automobile vehicles. Test methods and admissible outside noise level.
25	PN-71 / S-04052	Automobiles. Test methods and admissible inside noise level.
26	PN-74 / S-47013	Lorries, buses and trolleybuses. Drivers cabins. Requirements.
27	PN-75 / S-76006	Audible warming devices for privileged motor vehicles. Requirements and tests.
28	PN/T-01009	Electroacoustics. Terms and definitions (Polish Draft Standard for experimental application).
29	PN-64 / T-06460	Sound level meter. General requirements and technical tests.
30	PN-73 / Z-70050	Medical equipment. Clear tone classification audiometres. General requirements and tests.
31	PN-76 / Z-70051	Medical equipment. Audiometres for general purposes. General requirements and methods of testing.
		B. Branch standards
32	BN-73 / 1340-14	General purpose gears. Noise measurement methods.

ctd. tab.

1	2	3
33	BN-75 / 1340-14	Noise of piston combustion engines. Methods of determining acoustic parameters.
34	BN-76 / 1807-01/00	Textile machinery and equipment. Noise measurement methods.
35	BN-76 / 1807-01/01	Textile machinery and equipment. Noise measurement methods at a distance of 1 m from the machine outline.
36	BN-76 / 1807/01/02	Textile machines. Technical permissible sound power levels of looms.
37	BN-76 / 1807-01/03	Textile machines. Technical permissible sound power level of spinning machines.
38	BN-76 / 1807-01/04	Textile machines. Technical permissible sound power levels of plaiting machines.
39	BN-71 / 3209-01	Permissible noise level in rooms of automatic telephone and telegraph exchanges. Requirements and investigation.
40	BN-73 / 3209-02	Permissible noise levels in rooms with telephonic connection and auxiliary hand-operated stands. Requirements and investigation.
41	BN-73 / 3209-03	Permissible noise level in rooms with telegraphic apparatus stands. Requirements and investigation.
42	BN-74 / 3209-04	Permissible level of noise and vibrations in buildings of telecommunication power plants. Requirements and investigation.
43	BN-75 / 3209-05	Permissible level of noise and vibration in operational post-office rooms. Packet distribution rooms. Requirements and investigation.
44	BN-69 / 3510-08	Rolling stock. Acoustic properties. Requirements.
45	BN-74 / 3612-02	Automobiles. Exhaust silencer. Requirements and investigation.
46	BN-75 / 3801-02	Aircraft with maximum take-off weight of 5700 kg. Inner noise. Methods of determining acoustic parameters.
47	BN-76 / 3801-03	Propeller aircraft. Noise in pilot's compartment. Methods of determinating acoustic parameters.
48	BN-75 / 5615-01	Piezoelectric transducers of accelarations with ceramic element. General requirements and investigation.
49	BN-75 / 8824-01	Building acoustics. Measurement of sound absorption coefficient in reverberation chamber.
50	BN-75 / 9360-17	Aircraft. Permissible noise levels. Methods of determining aircraft noise characteristics.

Copies of these standards (in Polish only) are available from: Centralna Księgarnia Normalizacyjna, Sienna 63, 00-820 Warszawa.

Marek Tucholski (Warszawa)

XXIII OPEN SEMINAR ON ACOUSTICS

Wisła, September 6-11, 1976

The XXIII Open Seminar on Acoustics was held in Wisła between 6-11 September 1976. It was organized by the Committee of Acoustics of the Polish Academy of Sciences, the Polish Acoustical Society and the Physics Institute of the Silesian Technical University.

The Seminar was attended by over 200 participants from Poland and also by invited guests from the Soviet Union, the German Democratic Republic, France and Holland.

The sessions of the Seminar were held in three parallel sections:

Section A – Quantum and molecular acoustics, ultrasonic physics, ultrasonic transducers, the applications of ultrasound in engineering and medicine (65 lectures).

Section B – Methods of noise control, architectural acoustics (38 lectures).

Section C – Psychological and physiological acoustics, musical acoustics, cybernetic and telecommunication acoustics (49 lectures).

Each day began with a plenary lecture. The individual lectures delivered in the different sections together gave a review of the actual state of the work being carried out in Poland in the different fields of acoustics.

The next XXIV Acoustics Seminar will be held in September 1977 in Gdańsk.

Plenary lectures

1. A. Rakowski – *Fundamentals of musical hearing.*
2. J. Kacprowski – *Acoustic modelling of speech organs in medical diagnosis.*
3. B. Zapiór – *A review of the problems of present-day sonochemistry.*
4. F. Kuczera – *The theory of the liquid state and acoustical information.*

Section lectures

Section A

1. J. Tabin – *Some problems in the calculation of the field of the ultrasonic echo from long objects.*
2. M. Szalewski – *Properties of active piezoelectric CdS layers obtained by single-source evaporation.*
3. A. Opilski, Z. Cerowski, T. Pustelny, M. Urbańczyk – *Measurement of the velocity and attenuation of Rayleigh surface waves.*
4. M. Szustakowski – *The reverberation echo of magnetoelastic waves in YIG monocrystals.*
5. A. Opilski, T. Pustelny – *The effect of a boundary layer on the propagation of a Rayleigh waves.*
6. R. Dyba – *Analysis of the size of the discontinuity region in an inert gas for waves with continuous spectra.*
7. A. Kawalec, B. Węcki – *Some effects of surface waves in piezosemiconductors in the semiconducting layer.*
8. J. Lewandowski, J. Ranachowski, F. Rejmont – *Propagation of plane longitudinal waves in inhomogeneous solid bodies.*
9. B. Niemczewski – *Interaction of the surface tension and the acoustic impedance of a fluid.*
10. A. Opilski – *The effect of surface trapping on the propagation of surface wave in a piezoelectric-semiconductor.*
11. M. Dobrzański – *Quasicorpuscular properties of phonons.*
12. J. Berdowski, A. Opilski – *Investigations of the acousto-optical anomalies of DADA crystals near phase transitions with photon-phonon interactions.*

13. Z. Kleszczewski, M. Wojewoda – *Materials for the acousto-optical modulation of laser light.*
14. J. Narkiewicz-Jodko, P. Rajchert, A. Leszczyński – *Acousto-optical deflectors.*
15. A. Śliwiński – *Selected acousto-optical properties of liquid crystals.*
16. E. Soczkiewicz – *A new acoustic method for the determination of the volume of holes postulated by the "hole" theory of liquids.*
17. A. Drzymała, M. Cieślak – *Ultrasonic velocity and attenuation coefficient measurement at phase transitions in cholesterol oleate.*
18. J. Krzyk – *Remarks on the problem of the relation of the velocity of sound and the intermolecular interaction potential.*
19. J. Gmyrek – *On an acoustic method for the determination of density as a function of pressure.*
20. S. Szyma – *Investigation of the degree of dispersion and the degree of homogenization of polydispersed systems by acoustic methods.*
21. W. Kasprzyk – *Determination of the size of aggregation zones in the parakinetic theory of acoustic aerosol coagulation.*
22. J. Smela – *The velocity of quasilongitudinal wave in regular crystals of fixed reaction potential.*
23. M. Urbańczyk – *Numerical analysis of a Rayleigh surface wave resonator.*
24. W. Boch, J. Goc – *Investigations of complexity in binary solutions of* $ZnCl_2$ *and* LiCl *in water by molecular acoustical methods.*
25. A. Snakowska, R. Wyrzykowski – *Impedance of the outlet of a semi-infinite pipe with a circular cross-section.*
26. Z. Bartnowski, B. Zapiór, A. Juszkiewicz – *Investigations of the conformation of organic esters using acoustic methods.*
27. A. Juszkiewicz, J. Potaczek – *Investigations of the hydration of polyethylene glycol by acoustic methods.*
28. J. Ranachowski, E. Ryll-Nardzewska – *Ultrasonic investigations of polymorphic transformations in steatite materials.*
29. Nguen Viet Kinh – *Excitation of Rayleigh waves with a laminated transducer.*
30. E. Danicki, J. Filipiak – *The frequency characteristics and impulse response of interdigital transducers determined on the basis of an equivalent diagram.*
31. Z. Jagodziński – *A new method for the calibration of ultrasonic transducers.*
32. W. Pajewski – *The designing of multi-layer transducers.*
33. W. Iłgunas – *Problems of the measurement of sound velocity in liquids at low ultrasonic frequencies.*
34. D. Ciplys – *Acoustic investigations of solids in Vilno University.*
35. A. Wojnar – *A piezoelectric ultrasonic* 630 *Watt transducer.*
36. E. Talarczyk, T. Ciszewski – *An ultrasonic transducer of high quality factor operating with impulse compression.*
37. J. Golanowski, T. Gudra – *Experimental investigations of resonance systems with vibration direction conversion.*
38. L. Lipiński – *An energetic model of dislocations compared versus with the Granato-Lücke theory.*
39. Z. Kaczkowski – *Ferrite ultrasonic transducers for the* 27 kHz*-band.*
40. T. Walecki – *The effect of a magnetic field on the mechanical quality factor of piezomagnetic ferrites.*
41. Z. Kaczkowski, E. Milewska – *The influence of magnetic polarization and thermal treatment on the magnetomechanical coupling, modulus of elasticity and dynamic magnetic capacity of* FeAl 12 *alloys produced on an industrial scale.*
42. T. Walecki – *Internal friction in alfer materials.*

43. Z. Kaczkowski, E. Milewska – *Piezomagnetic properties of high power alfer transducers for the* 22 kHz-*band.*
44. W. Bieńko, Z. Kaczkowski – *The interaction of a transistorized generator with alfer transducer for the* 33 kHz-*band.*
45. Z. Kaczkowski – *Piezomagnetic properties of alfer transducers for the* 82 kHz-*band.*
46. A. Korbicki, W. Pawlak, A. Albinowska – *Ultrasonic generator* CU-22-2500 (UTG 4).
47. J. Górczyński – *The development of techniques for the active use of ultrasound in Poland in the years 1978-1980.*
48. R. Kukulski – *Technical ultrasonic equipment developed in the Intitute of Telephone and Radioengineering.*
49. L. Filipczyński, J. Sałkowski – *An attempt at real time visualization of the heart by means of ultrasound.*
50. L. Filipczyński – *The effect of high temperatures developed in soft tissues under the action of transient focused ultrasonic fields.*
51. J. C. Bamber, D. Nicholas, C. R. Hill, M. J. Fry, F. Dunn – *Measurement of ultrasound scattering and attenuation in excised human tissues.*
52. J. P. Woodcoock – *Transfer function analysis in the study of occlusive arterial diseases.*
53. R. C. Chivers, W. Nasalski – *On the relationship between the object structure and the reconstructed image conveyed by the phase information in ultrasonic holography.*
54. T. Szlagowska, B. Niemczewski – *Results of the measurement of sound velocity and its temperature coefficient in five solvents used for ultrasonic cleaning.*
55. J. Olszewski – *The scaltering ultrasonic waves by bodies similar to erythrocytes and air bubbles.*
56. J. Etiene – *Applications of an ultrasonic method in arterography.*
57. T. Pawłowski – *Evaluation of blood flow rate using a continuous Doppler method.*
58. R. Kubak, A. Hoeks, F. Smeets – *Digital processing of a Doppler signal spectrum.*
59. B. Tal, J. Magiera – *Utilization of ultrasonic energy in periodic fluid-fluid extraction.*
60. J. Somer – *Some impressions of the Third World Congress on Ultrasonics in Medicine.*
61. G. Łypacewicz, L. Filipczyński – *The overall sensitivity of ultrasonic diagnostic apparatus measured by means of a fluid of high attenuation.*
62. T. Waszczuk, J. Kamler – *The grey scale of oscilloscope tubes in the light of ultrasonic visualization.*
63. K. Zasadziński, R. Kozaczewski – *The measuring capabilities of ultrasonic measurement unit* UZP-10.
64. Z. Kozłowski, A. Rekowski – *Ultrasonic flowmeter* PU-10.
65. A. Markiewicz – *On the possibility of developing a non-reflective piezoelectric ultrasonic transducer.*

Section B

1. T. Tyburski – *The investigation of perforated structures used for individual hearing protectors.*
2. W. Jankowski, W. Kusek, W. Birecki – *Intelligibility of speech in tramways during travel.*
3. M. Mirowska – *Tests to establish the relationships of the sound-absorbing properties of fibrous materials to their structure.*
4. A. Rudiuk – *Model investigations related to the selection of acoustic systems in aircraft cockpits.*
5. D. Kozłowska-Kowalczyk, Z. Wrocławski – *Permissible levels of sound power of textile machines.*
6. A. Lipowczan, I. Kubik, H. Olszyczka – *A computer program for the evaluation and classification of noise levels for mining machines.*

7. F. Denderes, J. Zalewski, Z. Worobiec – *Absorbing properties of domestic asphaltic pastes.*
8. J. Jakubczak, W. Tyrchan – *Silent fans for cooling piston compressors.*
9. W. Tyrchan – *The use of a variable cross-section channel as a pressure pulsation damper.*
10. A. Muszyński – *Air bubbles as a sound-absorbing lining for water basins.*
11. N. Mielczarek, A. Puch – *Investigation of acoustic filters.*
12. L. Dunkelmann, G. Budzyński, A. Witkowski – *On the knocks in electronic control.*
13. S. Czarnecki, E. Kotarbińska – *The interaction of sound-absorbing screens with sound-absorbing surfaces in industrial rooms.*
14. R. Makarewicz – *Regional planning in the light of traffic noise.*
15. R. Makarewicz – *The intensity of the acoustic field produced by a moving source in an open space.*
16. P. Schubert – *Experimental results concerning the accuracy of sound power measurement in a reverberant sound field.*
17. W. Bandera, A. Śliwiński – *On the possibilities of using the mechanical impedance measurements to determine the properties of dynamic viscoelastic materials.*
18. Z. Dukiewicz, A. Śliwiński – *Preliminary results of correlation measurements of the propagation of sounds in structures on board M/S A. Garnuszewski.*
19. A. Puch, T. Zamorski – *The real part of the impedance of the radiation from the horn outlet of a dynamic generator.*
20. E. Kozaczka, F. Markiewicz – *Propagation of acoustic turbulence in shallow seas for certain sound velocity profiles.*
21. W. Bartelmus, A. Studziński – *A spectral method for transmission gear diagnosis.*
22. E. Kozaczka, A. Muszyński – *A magneto as a source of acoustic disturbances in water.*
23. E. Kozaczka – *Investigation of the underwater noise produced by a propeller.*
24. J. Motylewski – *Diagnostic acoustic investigations of moulding machines.*
25. J. Karski, P. Pajzderski, W. Śleboda – *Reasons for the development of vibrations in machine tools, with grinding machine* RGF 5/115 *as an example.*
26. M. Rabiega, B. Rudno-Rudzińska, J. Zalewski – *Investigation of the statistical distribution of traffic noise.*
27. P. Leśniewski – *The effect developed when shutting of the sound source and its influence upon the reverberation time.*
28. E. Drescher – *Investigation of the properties of solidifying cement grouts by microscopic and ultrasonic methods.*
29. A. Jaroszewska – *Experimental investigations of the propagation of elastic waves in boreholes.*
30. H. Gawda – *Application of an impulse ultrasonic method to the investigation of the density distribution of ground subject to deformation.*
31. H. Idczak, B. Bogusz, J. Jurkiewicz – *A method for the selection of parameters for spectral analysis.*
32. H. Idczak, A. Jaroch, J. Renowski – *Application of a rotary diffuser in a weakly absorbing measurement chamber.*
33. G. Budzyński, M. Sankiewicz, A. Kulowski – *Acoustics of the cathedral in Oliwa.*
34. Z. Wąsowicz, J. Jaguścik – *Measurement of diffusivity of an acoustic field by a correlation method.*
35. Będkowski, B. Tokarz – *Selected acoustic problems in feature films.*
36. K. Brodnicki – *An anechoic segment chamber.*
37. A. Kulowski, M. Sankiewicz – *Investigation of audio-monitoring systems for tetraphonic recordings.*
38. J. Bień, E. Kowalska, E. Zielewicz – *Filtering properties of sonicated waste sediments.*

Section C

1. J. Renowski, S. Hlibowicki – *Selected problems in modelling hearing properties.*
2. J. Renkowski, K. Budno-Rudnicki, R. Toczyk – *The selection of listeners for sound location tests.*
3. S. Hlibowicki – *An analogue model of the basic membrane.*
4. T. Tyburski – *Results of physiological-ergonomic investigations of noise.*
5. A. Rakowski, A. Jaroszewski, E. Bogdanowicz – *A threat to the hearing of performing musicians operating high power equipment.*
6. M. Makowski – *Properties of the hearing image.*
7. A. Preis – *Representation of the physical spectrum of musical sound in a fixed two-dimensional space.*
8. H. Harajda – *Preliminary measurements of the sound range of a violin.*
9. A. Rakowski, M. Morawska – *Investigations of absolute hearing.*
10. Z. Wójcik – *The determination of the parameters of a speech signal with the aid of an instant memory.*
11. D. Szybista – *The influence of vocal context on the spectral features and the recognition of fricative consonants.*
12. M. Kozak, Cz. Basztura, M. Tyburcy – *Subjective measurement of the duration of phonemes and phrases.*
13. J. Zalewski, M. Myślecki, J. Jurkiewicz – *Application of linear prediction to the description of a certain class of phonemes.*
14. J. Kacprowski, W. Mikiel, R. Gubrynowicz, A. Komorowska, A. Szewczyk, W. Tłuchowski – *Acoustic diagnosis of a voice-producing throat.*
15. B. Adamczyk, W. Kuniszyk-Jóśkowiak, E. Smołka – *Correlation between the effect of echo and reverberation on the speech of stutterers.*
16. H. Kusek, W. Birecki – *Speech perception of a young person employed in school workshops.*
17. J. Jarycki, Z. Worobiec – *Investigation of skin vibrations in the throat region during speech articulation.*
18. W. Cholewa – *Optimization of the numerical parameters in the analysis of acoustic signals.*
19. R. Millner, B. Gramlich, M. Millner – *Ultrasonic measurements on bone tissues.*
20. M. Millner, R. Millner, H. Grossmann – *Glottography with ultrasonic waves.*
21. R. Carre – *Investigations in the field of speech acoustics carried out in research centres in France.*
22. E. Tyburcy, J. Zalewski – *Spectral transitions as distinctive features of phoneme connections.*
23. A. Pawlak, C. Basztura, W. Majewski – *Application of the Bayes optimal decision rule for cases with incomplete probabilistic information for the identification of voices.*
24. A. Gos, J. Zalewski, W. Myślecki – *On the relationship between the parameters of the impulse generator for throat excitation and the articulation channel on the tonal quality of short phrases of the Polish language.*
25. J. Jurkiewicz, J. Zalewski, W. Myślecki – *The dependence of the tone of code phrases of the Polish language on the order of the analyzing digital filter.*
26. K. Mytkowski – *The transmission in real time of a speech signal and its parameters between a MERA minicomputer and its steriperipheral units.*
27. H. Kubzdela – *A model for an analogue-digital system to identify Polish vowels in simplified phoneme progressions.*
28. J. Jarycki – *An objective method for the evaluation of transmitted speech quality using throat transducers.*
29. W. Majewski, C. Basztura, H. Holien – *Short-term identification of speakers.*

30. W. HAMER – *A model of an artificial head.*
31. K. MUSIALIK, W. MAJEROWSKI, W. MYŚLECKI – *Multi-channel acoustic output from Electric Digital Computers with limited information store.*
32. W. MIKIEL – *A computer system for the measurement and processing of acoustic data.*
33. J. ZALEWSKI, W. MYŚLECKI – *Investigations concerning the optimal parameters of throat excitation for the synthesis of short phrases of Polish speech.*
34. A. RAKOWSKI, T. ŁĘTOWSKI, R. LITWIN – *Analysis of the possibilities of using a microphonic system in an artificial head for recording music.*
35. J. KONIECZNY – *An attemp at a theoretical definition of the active surface of a dynamic microphone membrane.*
36. J. FLORKOWSKI, B. REWIŃSKA – *Directorial properties of an impulse excited loudspeaker.*
37. S. HLIBOWICKI – *Relationships between the loudspeaker efficiency and its characteristic frequencies.*
38. A. DOBRUCKI, CZ. ROSZKOWSKI – *Measurement of the complex Young's modulus of the cellulose used for loudspeaker membranes.*
39. K. RUDNO-RUDZIŃSKI – *Multiple-loudspeaker systems.*
40. A. DOBRUCKI – *Equations of the vibrations of loudspeaker membranes.*
41. Z. SOŁTYS, Z. G. WĄSOWICZ – *The rooms for the audiometric investigation of loudspeakers in the Institute of Telecommunications and Acoustics.*
42. J. FLORKOWSKI – *The location of virtual sound sources for signals of various durations.*
43. A. GABOR, J. ZARZYCKI – *The accuracy of the determination of the (linear) transmittance function of real electroacoustic channels.*
44. J. ZARZYCKI, A. GABOR – *A sinusoidal method for the measurement of quantities fully characterizing non-linear deformations.*
45. S. NUCKOWSKI, J. SZYMBOR – *A method for the automatic measurement of multidimensional functions of the transmittance of non-linear inter systems.*
46. W. GŁOWACKI, W. SUŁKOŃSKA – *Analysis of deformations and noise in electroacoustic channels.*
47. B. ROGALA, R. SZKOP, R. ZMONARSKI – *Evaluation of the effect of the nonlinearity and the inertia of an electroacoustic system on its impulse response.*
48. S. NUCKOWSKI, B. ROGALA – *Measurement of non-linear deformations in the low frequency channels of radiophonic receivers using a "break in the spectrum of the measurement signal".*
49. W. KULESZA, B. ROGALA, J. SOBOLEWSKI – *An attempt at the evaluation of transmittive properties of musical signals on the basis of their statistical characteristics.*

Aleksander Opilski (Gliwice)

IV CONFERENCE ON «NOISE CONTROL»

Warszawa, October 13-15, 1976

The conference was organized by the Acoustical Committee of the Polish Academy of Sciences and the Polish Acoustical Society with the assistance of the Institute of Fundamental Technological Research of the Polish Academy of Sciences and the International Institute of Noise Control Engineering I/INCE.

The conference was the Fourth National Conference of a series of conferences organized every three years, and at the same time the First International Conference. The international character of the conference contributed to its high level and permitted closer contacts to be made between Polish and foreign specialists.

The Conference was held in the Palace of Science and Culture, and included an exhibition of measuring devices, sound-proofing and anti-vibratory materials and systems.

Lectures were presented in three forms: invited papers delivered by internationally eminent scientists at the invitation of the organizing committee, contributed papers presented in two parallel lecture sections, and lectures presented in poster sessions.

The poster form is a relatively new form of information exchange and the organizing committee therefore devoted much attention to it. The participants who had expressed their desire to present their lecture in poster form were each given a numbered stand, confined from three sides and open at the front. This enabled them to lecture to standing groups of up to 15 listeners. In order to facilitate discussion in groups of 2 to 4 people, each stand was equipped with a small table and chairs.

Materials in the form of large-scale illustrations were hung on the internal walls of the stands whose area was 2 m^2. Participants were provided with note-pads for use in discussions.

The procedure for the presentation of poster lectures was the following: the authors were given 5 minutes at the plenary sessions to present briefly the assumptions and theses of their work. During this time they could project slides and use a print projector. The purpose of this was to communicate to all participants the scope of the lectures, thus enabling them to select poster lectures which would be of particular interest to them.

The main part of the poster lectures took place for one-hour session on each of 3 days during which the authors were present at their stands for a predetermined half hour to present more extensively their results and for discussion. The remaining time allowed authors to hear other poster lectures of interest to them.

The poster form created optimal conditions for discussion between participants working on similar problems and anxious to establish closer contacts. Its popularity with the participants proved both its practicality and its contribution to the success of the conference.

The conference proceedings «76 Noise Control Proceedings», of 462 pages, have been published and distributed to the participants and other interested. They are available from prof. S. Czarnecki, Institute of Fundamental Technological Research of the Polish Academy of Sciences, 00-049 Warszawa, Poland.

LIST OF PAPERS PRESENTED

Invited papers

F. Ingerslev – *Planning against transportation noise*

M. J. Crocker – *Reduction of diesel engine noise*

H. G. Leventhall – *Developments in active attenuators*

F. P. Mechel – *Why are silencers symmetrical?*

W. Schirmer – *The vibro-acoustical transmissibility of machine structures*

Z. Maekawa – *Noise shielding on highway*

P. Liénard – *Acoustic propagation in the low atmosphere*

C. Bardone-Sacerdote, G. Sacerdote – *A critical survey of the damage criteria on noise exposure in industry*

Contributed and poster-form papers

S. Gruhl – *Sound propagation in workshops with different arrays of sources*

U. J. Kurze – *Methods and examples of noise reduction in industrial halls*

J. Regent, L. Kałmucki – *Investigation of sound absorption efficiency by absorption lining in the field of the reflected sound wave*

R. Friberg – *Industrial noise control obtained by acoustic enclosures and acoustical treatment of ceilings and walls*

R. Makarewicz, G. Kerber – *Relationship between road traffic and the values of equivalent level L_{eq}*
S. Czarnecki, E. Kotarbińska – *Model investigations of the efficiency of acoustic barriers in industrial halls and urban areas*
J. Grabek, R. Kucharski – *Acoustic map of Warsaw; graphical presentation of acoustic climate trends (reasons and conclusions)*
M. Stawicka-Wołkowska – *Investigations on traffic noise annoyance on the territories adjacent to express routes*
P. François – *Reference sound sources-characteristics, calibration and utilization*
Cz. Cempel, M. Majewski – *Estimation of the plant rooms acoustical properties by means of coherence function*
E. Ozimek – *Fast Fourier transform in acoustic diagnostic research of diesel engine*
Z. Kyncl – *On the spectral analysis of single acoustic impulses*
W. Cholewa – *Problems concerning the practical realisation of Fourier's fast transform with the purpose of analysing acoustic signals*
A. Darwen, J. Ludlow – *A computer based system for monitoring aircraft noise exposure*
J. Adamczyk, P. Krzyworzeka – *A method of identification for diagnostic in vibroacoustics*
W. Rajchert, A. Grzejszczyk, K. Szymański – *Infra- and ultrasounds in building construction equipment*
J. Sentek – *Reduction of noise accompanying the outflow of compressed gas or steam to the atmosphere*
R. R. Armstrong, H. V. Fuchs, A. Michalke, U. Michel – *Influence of Mach number on pressure fluctuations relevant to jet noise*
A. Bigret, J. Delcambre – *Steam turbine generator design and noise*
R. Agnon, M. Bartenwerfer, T. Gikadi, W. Neise – *Noise reduction at the source in centrifugal fans*
W. M. Jungowski, W. C. Selerowicz, K. J. Witczak – *Some features of choked air jets generating discrete frequency noise*
M. Czechowicz, S. Czarnecki – *Attenuation of choked airflow by chamber-disc suppressor*
R. Stuff – *Analytic solution for the sound propagation through the atmospheric wind boundary layer*
S. Till – *Silencing main ventilation mining fans avoiding energetic power loss*
V. D. Naylor – *Experiments on elemental efficiencies and damped vibrations*
L. Millei – *Shock and vibration response of art memorial church to sonic boom and road traffic*
G. Siudyła, M. Zabawa – *Vibroacoustical model of a plate driven by an impulse force*
W. J. Stojanowski – *Theoretical vibroacoustic model of a driving system*
Z. Engel – *Noise reduction of chosen casting machines*
A. Lipowczan, L. Fajfrowski, H. Olszyczka, T. Malinowski, T. Rabsztyn, W. Beble, R. Wagstyl, W. Mrukwa, J. Klepacki – *Noise prevention system in the Polish coal industry*
J. Rutkowski – *Analysis of the possibilities on improvement of acoustic conditions in prefabrication plants by means of acousticobuilding protections*
M. Makomaski, J. Kaźmierczak – *Total absorption of rolling mill house*
J. Kaźmierczak, M. Makomaski – *Investigations concerning the means and ways of reducing noise emitted by an electric arc furnace for the smelting of steel*
A. Zmysłowski – *Some aspects of the reduction of noise in the case of an electric steel arc furnace*
L. Pimonow – *Discomfort caused by noise*
I. Ratajska, R. Makowiecka – *The hearing and equilibrium condition in the ultra-sound defectoscope operators*

Z. Bochenek, I. Ratajska, K. Stelmaszek — *The dynamics of the chronic acoustic trauma in subjects with the postinflamatory changes of tympanic membranes*

L. W. Tweed, D. R. Tree — *Close fitting acoustical enclosures*

Bo Nyström — *Noise reductions in heavy constructions by means of thin viscoelastic damping layers*

I. Żuchowicz — *Acoustic properties of sound absorbing perforated construction*

A. Gackiewicz, W. Orliński — *Searching and practical usage of sound-proofed materials basing on building machines*

St. Czareński, J. Kowal — *Vibroisolation of rubberlike materials*

M. Menżyński, B. Niewczas, A. Troszok — *Sound insulating covers for industrial fittings*

S. Będkowski, S. Duda, Z. Jakubek, M. Staffa — *Prefabricated sound-absorbing cabins for industry*

M. Jessel — *Noise control by means of active absorbers* (part I: theory)

G. Mangiante — *Noise control by means of active absorbers* (part II: Sound absorbers in a long duct)

M. Vogt — *Correlation analysis of phase cancellation in an acoustic field*

M. Vogt, S. Czarnecki — *Analysis of sound source phase cancellation conditions*

G. Canévet — *Noise control by means of active absorbers* (part III: Experiments)

W. F. King III and D. Bechert — *Radiated noise from high-speed trains*

G. Engler — *The possibilities to reduce the noise outside of vehicles by the construction of the body*

H. Kusek, W. Birecki — *Industrial noise influence on the audibility of acoustic warning signals*

B. Buna, J. Miazga — *Estimation of noisiness of motor vehicles admitted to traffic with the aid of the measuring device* AS-2

J. Miazga — *Acoustic estimation of the country vehicle stock*

A. Rudiuk — *Methods applied on the territory of the Polish People's Republic for investigating external noise produced by light aircraft*

O. J. Pedersen — *Standards for the measurement of noise from household appliances*

A. G. Jhaveri — *Compatibility of the noise control legislation with those involving energy environment and land use planning*

S. Majoros, A. Csekö, Z. Ficsor — *Combustions noise amplification and feedback generation*

P. L. Timár — *Inverter fed induction motor and its noise*

Stefan Czarnecki, Ewa Kotarbińska (Warszawa)

MOLECULAR AND QUANTUM ACOUSTICS SECTION

The Central Board of the Polish Acoustical Society (PTA) has established a Molecular and Quantum Acoustics Section (AMK), which has its seat in Gliwice. The activities of the Section will comprise:

1. the initiation of investigations in Poland in molecular and quantum acoustics and in sonochemistry,

2. the education of young scientists and the exchange of information on the state and conditions of investigations being carried out in this field. The implementation of this aim will be effected during winter schooling organized every year in cooperation with the Upper Silesian PTA-Department.

At the meeting on September 8th, 1976, in Wisła, the Board of the Section was formed. It comprises a chairman: Aleksander Opilski (Gliwice), a vice chairman: Antoni Śliwiński (Gdańsk), a secretary: Joachim Gmyrek (Gliwice), and two members: F. Kuczera (Gliwice), M. M. Dobrzański (Warszawa).

BOOK REVIEW

Reduction of Machinery Noise
Purdue University 1975

The book, edited by prof. Malcolm Crocker, contains materials from a training course entitled «Reduction of Machinery Noise» organized on December 10-12, 1975, in Purdue University. It contains part of the revised and supplemented lectures from a previous course (May 13-17, 1974) whose conference materials were briefly discussed in issue 3, vol. 10 (1975) of Archiwum Akustyki.

The essential advantage of the book under discussion is a very good balance of the theoretical problems and its practical application. Add to this a very careful and clear elaboration of present day problems it can be seen that the book should arouse interest among acousticians engaged in problems of noise control at all levels: from the fundamental to concrete practical solutions. The book has forewords by the main promotor of the course prof. M. J. Crocker, and prof. Ray Cohen, director of the Ray W. Herrick Laboratories School of Mechanical Engineering Purdue University, the centre at which the course was organized.

LIST OF PAPERS PRESENTED

Fundamentals of Noise Control

J. W. Sullivan – *Sound waves and acoustical definitions*
F. R. Fricke, D. R. Tree – *Room acoustics*
P. G. Vaidya – *Sound propagation, outdoors*
W. A. Cooper – *The effects of noise on people*
D. R. Tree – *Instrumentation and noise measurements*
M. J. Crocker – *Use of anechoic and reverberant rooms for measurement of noise from machines*
M. J. Crocker – *Noise control approaches*
J. F. Hamilton – *Fundamentals of vibration and noise control by vibration isolation*
W. Soedel – *Noise control by absorption*
M. J. Crocker – *Noise control by use of enclosures and barriers*
M. J. Crocker – *Noise control with mufflers*
M. J. Crocker – *Noise legislation and regulations*

Reduction of machinery noise

D. F. Fowler – *Instrumentation for noise and vibration measurement*
A. J. Schneider – *Noise measurements*
P. K. Baade – *Identification of noise sources*
R. L. Staadt – *Truck noise control*
R. S. Lane – *Sources and reduction of diesel engine noise*
A. F. Seybert, M. J. Crocker – *Noise source identification in diesel engines*
W. R. Thornton – *Noise control of new and existing petrochemical facilities*
J. B. Moreland – *Controlling industrial noise by means of room boundary*
R. C. Locke – *Automatic strip feed press noise and its reduction*
R. J. Alfredson – *Noise source identification and control of noise in punch*
J. M. Guinter – *Noise from electrical equipment*
G. W. Kamperman – *Operator noise control in construction machinery*
L. F. Yerges – *Noise reduction in metal cutting operations*
R. L. Bannister – *Large steam turbine-generator noise control*

K. Arcuri – *Valve and pipeline noise causes and cures*
G. M. Diehl – *Centrifugal compressor noise reduction*
J. B. Graham – *Noise of fans and blowers*

Noise case histories

W. Soedel – *Manifold design of piston machinery using a Helmholtz resonator approach*
L. W. Tweed, D. R. Tree – *The use of acoustical enclosures to quiet small internal combustion engines*
S. L. Applegate, M. J. Crocker – *Reducing the noise of a rotary lawn mower blade*
M. J. Crocker, D. R. Tree – *Acoustic enclosures for diesel engines in trucks.*

The questions dealt with in the book do not, of course, include all the problems of noise control but concentrate mainly on the group of problems whose solution is most important in the USA. Amongst them are primarily Diesel engines, where without undertaking long-term systematic investigations, based on modern measuring methods, positive results could not be expected.

Many papers are devoted to concern flow equipment, such as compressors, blowers, valves and fans, and this gives evidence of an extensive development of the problems of aerodynamic noise thus enabling the practical utilization of fundamental investigations.

Many of the papers deal with methods of reducing noise by sound-proofing rooms, and the use of barriers and enclosure for the devices producing the noise. It should be stressed that, independent of their acoustic advantages, the practical developments described are remarkable for their long life, simple light construction, and pleasing appearance.

An essential feature of the papers is that they are concise and well arranged, thus making the book of great practical and didactic importance.

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