

A METHOD FOR OBJECTIVE EVALUATION OF THE SENSATION OF THE LOCALISATION DIRECTION OF A SOUND SOURCE

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This paper proposes a new method allowing the evaluation of the sensation of the localisation direction of an image sound source, generated by signals transmitted by a two - channel loudspeaker stereo - system. This evaluation can be carried out on the basis of the localisation parameters of signals reaching the membrane of the listener's ear drum, i.e. it does not require psychoacoustic investigations. The method proposed takes into account the properties of the human hearing organ, as known from psychoacoustic investigations, which are essential in localisation.

In the final part of the paper, by using the method proposed, a critical evaluation is carried out of a two - channel loudspeaker intensity stereo - system.

1. Introduction

The sensation of sound spatiality, generated by signals reaching the listener's ears directly or through a stereophonic system, is shaped by a large number of factors [3, 4, 9-11, 18]. The evaluation of their effect on the sensation of spatiality is commonly carried out by using psychoacoustic investigations. However, as the number of these factors is large and the psychoacoustic investigations are very time - consuming, the results obtained do not permit comprehensive and thorough analysis of the phenomenon, particularly in the case of stereophonic systems.

The sensation of sound spatiality is also evaluated, in addition to psychoacoustic investigations, by means of other research methods. These methods can be divided into two principal groups. The basic element of one of them is constituted by measurements of changes in the parameters of signals on the path between the sound source and the listener's ears, performed by using an acoustic head analogue or a listener [1, 5, 8, 16]. The other is based on analysis

of phenomena occurring during the transmission of the signals, by using mathematical models [2, 9, 11, 12]. These objective methods permit the determination of the parameters of signals reaching the listener's ears, but give no answer as to what sensation of spatiality is caused by these signals. Thus, there is the need for developing a method which would provide an answer to this question. This need has been pointed out e.g. by BLAUERT [3].

Spatial hearing is a property of the human hearing organ which consists of a large number of elements. FURDUEV [4] distinguishes three components of spatial hearing:

- localisation of sound sources,
- distinguishing signals,
- acoustic atmosphere.

This paper presents a method permitting objective evaluation of the basic component of spatial hearing which is the localisation of sound sources. In developing the method, both the results of psychoacoustic investigations and those of objective measurements of the responses of the external auditory system of man were used. The development of this method required:

- the definition of the so - called localisation parameters, i.e. those parameters of signals reaching the membrane of the listener's ear drum on the basis of which he localises the sound source (section 2),
- the development of a criterion linking the localisation parameters with the sensation of the localisation direction of a sound source which the listener has (sections 3-7).

The method proposed requires knowledge of the localisation parameters of signals reaching the membranes of the listener's ear drums. For given conditions and a given signal emitted by a sound source these parameters can be determined from measurements or calculated from mathematical models, which describe the passage of the signal from the sound source to the listener's ears.

In the final part of this paper, as an illustration of the possibility of application of the method proposed, the results of evaluation of a two - channel loudspeaker intensity stereo - system are given.

2. Definitions of the localisation parameters

As a result of psychoacoustic investigations, it has been established [6, 7, 9, 15, 17] that the listener evaluates the localisation direction of a sound source above all on the basis of the following parameters of signals reaching his ears:

- uniaural sound pressure levels,
- interaural difference between sound pressure levels,
- interaural time delay.

The following parameters of the signals, called localisation parameters below, can be subordinated to the above parameters:

— short-term power spectrum levels L_{SL} and L_{SP} of signals reaching the left and right ears:

$$L_{SL}(f, t_0) = 10 \log \frac{S_L(f, t_0)}{S_0}, \quad L_{SP}(f, t_0) = 10 \log \frac{S_P(f, t_0)}{S_0}, \quad (1)$$

where $S_L(f, t_0)$ and $S_P(f, t_0)$ are the short-term power spectra of the signals for the left and right ears, S_0 is the reference quantity,

— interaural difference between the short-term power spectrum levels, ΔL_S , of signals for the left and right ears:

$$\Delta L_S(f, t_0) = L_{SL}(f, t_0) - L_{SP}(f, t_0), \quad (2)$$

— interaural time delay C_S :

$$C_S(f, t_0) = \tau_{SL}^*(f, t_0) - \tau_{SP}^*(f, t_0), \quad (3)$$

where τ_{SL}^* and τ_{SP}^* are the time delays of a direct wave, describing the passage of a signal from the sound source to the membranes of the ear drums of the left and right ears.

The short-term power spectra occurring in formulae (1) are defined in the following way:

$$S(f, t_0) = \int_{-\infty}^{\infty} W_w(|\tau|) \cos 2\pi f\tau \left[\int_{-\infty}^{t_0} R(t, t - |\tau|) W_k(t_0 - t) dt \right] d\tau, \quad (4)$$

where $R(t, t - |\tau|)$ is the autocorrelation function of a nonstationary random process, $W_k(t)$ is the window function in the time domain, $W_w(\tau)$ is the window function in the domain of time shift, and t_0 is the moment of the determination of the short-term power spectrum.

In definition (4), the window function $W_k(t)$ permits a property of the human hearing organ, called a time constant, to be taken into account. Among research workers there is difference of opinion as to the shape and length of the window function $W_k(t)$ which describes properly the inertial nature of the hearing organ. In the calculations presented in a further part of this paper, a window function proposed by PENNER [14], described by the following relation, was used:

$$W_k(t) = 1(t) 1(T_{S0} - t) \frac{2t}{T_{S0}} \exp\left(-\frac{2t}{T_{S0}} + 1\right), \quad (5)$$

where $1(t)$ is the Heaviside distribution, T_{S0} is the length of the time constant of hearing; it was assumed that $T_{S0} = 50$ ms.

From the psychoacoustic investigations reported on in MISZCZAK's paper [13], it follows that the listener localises a sound source on the basis of the

initial part of a signal reaching his ears, which results from the resumption of sound emission by the source. This property of hearing is known in the literature as "CREMER's law". The initial part of the signal, as a result of the delay of reflected waves with respect to the direct wave, contains mostly the latter wave. Due to this property of hearing, man can correctly localise sound sources in an interior, despite the fact that the energy of reflected wave is often higher than that of the direct wave. CREMER's law imposes the necessity for evaluating the parameters of signals reaching the listener's ears at the time moments $t_0 = t_c + T_{SO}$, where t_c is the moment when the wave front reaches the listener.

3. Localisation of sound sources in standard conditions

The standard conditions are understood here to be the conditions of direct detection in an open space of a signal emitted by a sound source. In such conditions the signals reaching the listener's ears are determined by the transfer functions of the external auditory system, corresponding to a given localisation of the sound source with respect to the listener, and by the signal emitted by this source [10, 11]. In turn the localisation parameters ΔL_S^B and C_S^B (the index B denotes the standard conditions) depend above all on the transfer functions of the external auditory system and on the localisation direction of the sound source with respect to the listener. In localisation, the parameters L_{SL}^B and L_{SP}^B play an auxiliary function [18] and therefore they will for the time being be neglected. The way of describing the localisation of the sound source with respect to the listener is shown in Fig. 1.

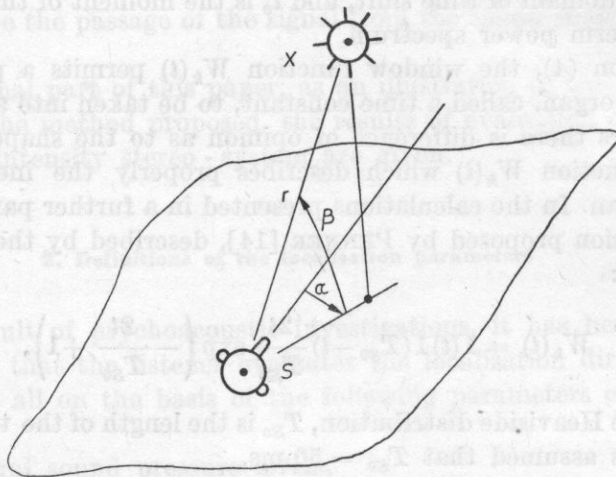


Fig. 1. The localisation of the sound source X with respect to the listener S

The listener can quite easily determine the localisation direction of a real sound source in anechoic conditions. In such conditions localisation is natural. When in the standard conditions the source, situated at an angle (α, β) with respect to the listener, emits a signal with its spectrum concentrated round some frequency f_1 , this signal causes a definite time delay $C_S(f_1) = C_S^B(f_1)$ and the difference between short-term spectrum levels, $\Delta L_S(f_1) = \Delta L_S^B(f_1)$. In the standard conditions there is then a function which subordinates the values of ΔL_S^B and C_S^B to given values of f, α, β . Since this subordination is described by a relationship of five quantities, this situation can be represented in a five-dimensional space $(\Delta L_S, C_S, f, \alpha, \beta)$. For definite f, α, β a point is described by a relationship of five quantities. Assumption by f, α and β of all the values within their variability ranges causes a set H^* to be generated. In the space $(\Delta L_S, C_S, f, \alpha, \beta)$ this set will be represented by a line. Points lying on this line will describe all the interrelationships of the five quantities defining the localisation in the standard conditions which are possible for a given listener. Thus, the set H^* constitutes a description of the localisation for direct detection in anechoic conditions.

As a result of visual and auditory experiences, in the process of learning, the listener memorises the interrelationship of the five quantities describing the localisation in the standard conditions. When the system sound source - listener is in anechoic conditions and the listener cannot see the sound source, then, on the basis of the frequency f , the delay C_S^B and the level difference ΔL_S^B , the listener can determine the localisation direction of the sound source, i.e. the angles α and β . This evaluation is possible due to knowledge of the interrelationship of the quantities $\Delta L_S^B, C_S^B, f, \alpha$ and β , described by the set H^* . It can thus be stated that the set H^* is a description of a decision-making model of the localisation of sound sources.

In view of the individual feature of their bodily constitutions, for each of listeners there occurs a different interrelationship of the quantities $\Delta L_S^B, C_S^B, f, \alpha$ and β . For a given listener this interrelationship is invariable, and so are the shapes of the external auditory system. In the process of learning every one adopts a different subordination, i.e. creates his own set H^* . These individual subordinations constitute individual decision-making models.

4. Averaged decision-making model

In evaluating stereophonic systems it is warranted to use the decision-making model of "the standard listener" who represents the properties of the hearing organs of a human group. Thus, it is necessary to develop an averaged decision-making model. To date, no investigations of the properties of the hearing organ have been carried out with the view to developing an averaged decision-making model, nevertheless measurements have been taken of the "shadow"

effect of the head solid. The results of these measurements were gathered by Shaw and represented in the form of a set of the averaged moduli of the transfer functions from the free field to the membranes of the listener's ear drums, as a function of frequency and the angle α [17]. On the basis of these moduli, using Hilbert's transform, a set of impulse responses of the external auditory system was calculated [10, 11]. In turn these impulse responses became the basic elements of the mathematical model of signal passage from the sound source to the listener's ears in standard conditions. With this model, assuming that the autocorrelation function of a signal emitted by the sound

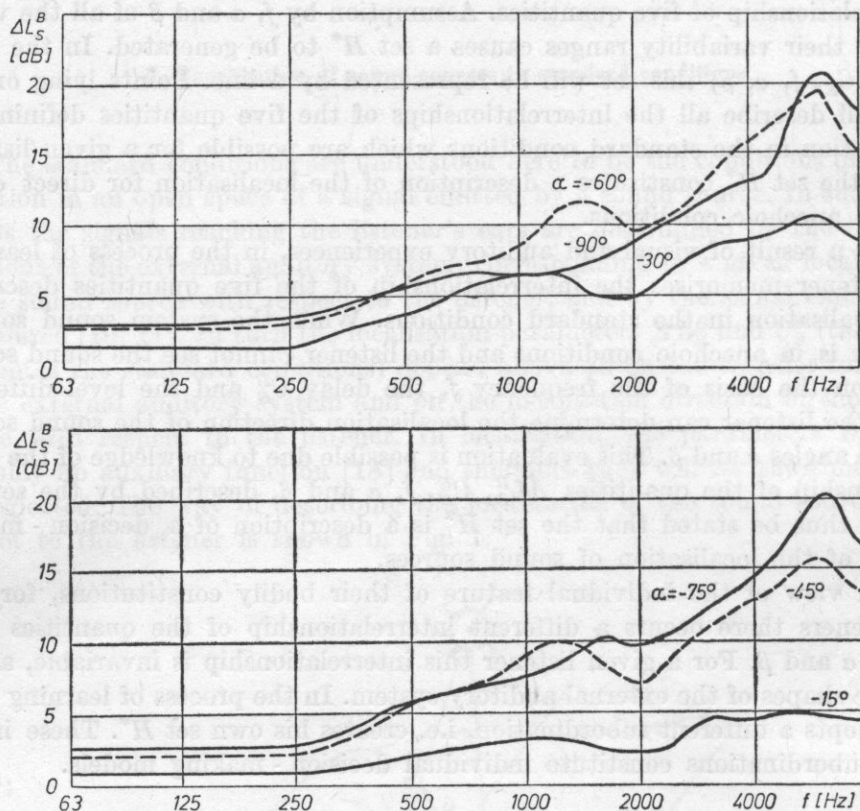


Fig. 2. The differences in the short-term spectral power levels, ΔL_S^B , in standard conditions, for various angles α

source has the form $R_{xx}(t_1, t_2) = {}^2\delta(t_1, t_2)$, where ${}^2\delta(t_1, t_2)$ is a twodimensional Dirac distribution, the dependencies of the level difference ΔL_S^B and the time delay C_S^B on frequency and the horizontal angle α was determined. These dependencies are shown in Figs. 2 and 3.

In view of lack of sufficient data, attempts to determine the effect of the vertical angle β on ΔL_S^B and C_S^B failed. Therefore, considerations were restricted

to the horizontal plane, i.e. to the case $\beta = 0$. This restriction reduces the decision - making model to four dimensions: ΔL_S , C_S , f and α . In practice this has no essential significance, since the localisation of image sound sources in two - channel loudspeaker stereophony should be limited to the horizontal plane. The elevation of an image source, which is observed at times, will from

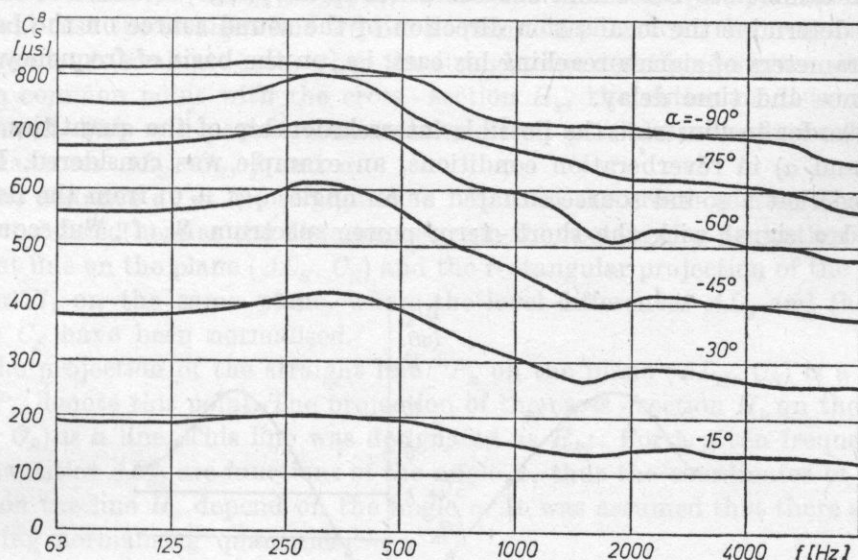


Fig. 3. The time delays C_S^B in standard conditions, for various angles α

this point of view be regarded as disagreement between the effect achieved and the desired one.

The set defined in the fourdimensional space (ΔL_S , C_S , f , α) was designated as H . In this space the set H is represented by a line.

5. Definition of the distance d_k and the angle α_k

The set H describes the interrelationship of the parameters ΔL_S , C_S , the frequency f and the angle α for direct detection in anechoic conditions. In these conditions localisation is obvious and natural. Most frequently, however, the listener will determine the localisation of the sound source in an interior where reflections from the walls occur. These reflections cause a change in the level difference ΔL_S (with respect to anechoic conditions) [10]. There is in turn no change in the time delay C_S , since it is determined by the direct wave. The interrelationship of ΔL_S , C_S , f and α is different from that in standard conditions, and in addition it varies with changing acoustic conditions. When reflections from the walls essentially change the level difference ΔL_S and the listener cannot see the sound source, he can find it difficult to define the locali-

sation direction of the sound source and his localisation can be erroneous [13]. In a case when reflections from the walls do not exert any large effect on ΔL_S , it is not difficult for the listener to determine the localisation direction of the sound source and his localisation is correct. Similar problems can occur for the detection of a signal transmitted by a stereophonic system. In this case, the listener cannot see the sound source. There is an image source. The listener has to determine the localisation direction of the sound source on the basis of the parameters of signals reaching his ears, i.e. on the basis of frequency, level difference and time delay.

In order to illustrate the possible interrelationship of the quantities (ΔL_S , C_S , f and a) in reverberation conditions, an example was considered. It was assumed that a sound source situated at an angle α ($\beta = 0$) from the listener, emitted a signal with the short-term power spectrum $S_{xi}(f)$. Subsequently,

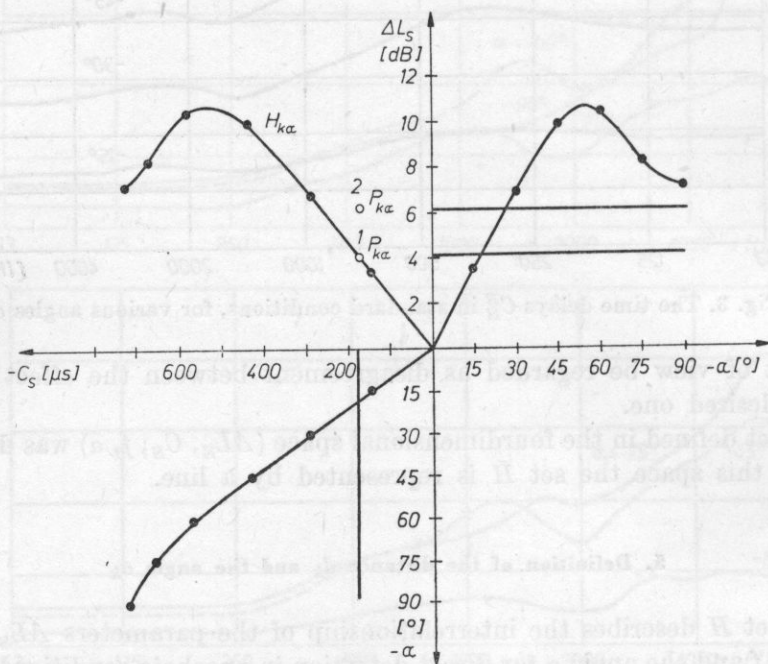


Fig. 4. Projections of the cross-section H_k and the straight lines 1P_k and 2P_k on the planes $(\Delta L_S, C_S)$, $(\Delta L_S, \alpha)$ and (C_S, α)

a spectrum-band with width df and the centre frequency f_k was considered. In standard conditions this band generates in the listener's ears $\Delta^1 L_{Sk}$ and ${}^1C_{Sk}$ — situation A, and with reflections from the walls, e.g. $\Delta^2 L_{Sk}$ and ${}^1C_{Sk}$ — situation B. As a narrow frequency band is considered, analysis of these situations requires only the cross-section H_k of the set H for the frequency $f = f_k$. This cross-section can be represented in the space $(\Delta L_S, C_S, \alpha)$. In the

three-dimensional space, situations A and B are described by the parallel straight lines $(\Delta^1 L_{Sk}, {}^1 C_{Sk}, \alpha)$ and $(\Delta^2 L_{Sk}, {}^1 C_{Sk}, \alpha)$ called respectively ${}^1 P_k$ and ${}^2 P_k$ below. Fig. 4 shows the projections of the cross-section H_k and the straight lines ${}^1 P_k$ and ${}^2 P_k$ on the planes $(\Delta L_S, C_S)$, $(\Delta L_S, \alpha)$ and (C_S, α) . The straight line ${}^1 P_k$, corresponding to anechoic conditions, intersects the cross-section H_k at a point with the coordinates $(\Delta^1 L_{Sk}, {}^1 C_{Sk}, \alpha_1)$ where the equation $\alpha_1 = \alpha_i$ is satisfied. In a general case the straight line ${}^2 P_k$ has no common point with the cross-section H_k . In order to evaluate cases when the straight line describing a given situation has no common point with the cross-section H_k , the notion of distance was introduced. This will permit consideration of all possible interrelationships of the quantities ΔL_S , C_S , f and α .

The distance \bar{d}_k^* of the straight line ${}^m P_k$ from the cross-section H_k of the set H is defined here as the distance between the rectangular projection of this straight line on the plane $(\Delta L_S, C_S)$ and the rectangular projection of the cross-section H_k on the same plane, where the level differences ΔL_S and the time delays C_S have been normalised.

The projection of the straight line ${}^m P_k$ on the plane $(\Delta L_S, C_S)$ is a point. Let ${}^m P_{ka}$ denote this point. The projection of the cross-section H_k on the plane $(\Delta L_S, C_S)$ is a line. This line was designated as H_{ka} . For a given frequency f_k the quantities ΔL_{Sk}^B are functions of the angle α , thus the coordinates of points lying on the line H_k depend on the angle α . It was assumed that there are the following normalising quantities:

- for the level difference ΔL_S — the number M_{Lk} , which is the maximum value of the level difference taken by the cross-section H_k for $\alpha \in (0, \pi/2)$,
- for the time delay C_S — the number M_{Ck} , which is the maximum value of the time delay taken by the cross-section H_k for $\alpha \in (0, \pi/2)$.

After normalisation the coordinates of the point ${}^m P_k$, (η_1, η_2) , are as follows:

$$\begin{aligned} \eta_1 &= \frac{\Delta^m L_{Sk}}{M_{Lk}} \\ \eta_2 &= \frac{{}^m C_{Sk}}{M_{Ck}}. \end{aligned} \quad (6)$$

In turn the coordinates of points lying on the line H_{ka} , (λ_1, λ_2) :

$$\begin{aligned} \lambda_1(\alpha) &= \Delta L_{Sk}^B(\alpha) / M_{Lk}, \\ \lambda_2(\alpha) &= C_{Sk}^B(\alpha) / M_{Ck}. \end{aligned} \quad (7)$$

The distance \bar{d}_k^* can be determined from the relation

$$\bar{d}_k^* \stackrel{\Delta}{=} \min \{ \varrho_k^*(\alpha) \}, \quad (8)$$

where

$$\varrho_k^*(\alpha) = \sqrt{[\eta_1 - \lambda_1(\alpha)]^2 + [\eta_2 - \lambda_2(\alpha)]^2}. \quad (9)$$

It should be borne in mind that in keeping with the definition the distance d_k^* is a number subordinated to the given frequency f_k .

The distance of the straight line mP_k from the cross-section H_k is conceived as a measure useful in determining the localisation direction of a sound source. Such a measure must in particular account for those properties of the human hearing organ that cause the time delay to have a decisive effect on source localisation at low frequencies and the level difference to do so at high frequencies. At medium frequencies the level difference and the time delay are equally essential [7, 18]. It is therefore necessary to introduce the weighted difference. After introducing the weight coefficients the definition of the distance becomes

$$d_k = \min \{\varrho_k(\alpha)\} = \min \left\{ \sqrt{W_{Lk}(\eta_1 - \lambda_1)^2 + W_{Ck}(\eta_2 - \lambda_2)^2} \right\}, \quad (10)$$

where W_{Lk} is the weight coefficient for ΔL_S for $f = f_k$; W_{Ck} is the weight coefficient for C_S for $f = f_k$.

At present there is no knowledge of the dependencies describing the properties of the human hearing organ as represented by the coefficients W_{Ck} and W_{Lk} in formula (10), except for their general trends. Therefore, it was assumed that they depended linearly on the logarithm of frequency and that they take values of zero or unity for frequencies of 16 Hz and 16.4 kHz giving the following expressions:

$$\begin{aligned} W_{Lk} &= 0.1 \log_2 f_k - 0.4, \\ W_{Ck} &= -0.1 \log_2 f_k + 1.4. \end{aligned} \quad (11)$$

The quantity α_k was also introduced. Let α_k denote an angle at which d_k occurs, i.e. α_k is such a value of the angle α at which the function $\varrho_k(\alpha)$ reaches a minimum. The quantity α_k is a function of the frequency f_k , just as so is the distance d_k .

6. Interpretation of the distance d_k and the angle α_k

As an example, Fig. 5 shows a projection of the cross-section H_k of the set H and projections of various possible straight lines mP_k , marked by points A , B , C and D , on the plane $(\Delta L_S, C_S)$. Fig. 5 shows the cross-section of the set H determined in section 4 for $f_k = 1200$ Hz.

In situation A the straight line mP_k has a common point with the cross-section H_k . The distance d_k is zero. α_k is about -0.15π (-27°). This situation can occur for direct detection in anechoic conditions, where there is the equation $\alpha_k = \alpha_i$ (α_i being the angle at which the sound source actually is). Situation A can also be generated by a signal transmitted by a stereophonic system. There is then full analogy to the standard conditions and it is not difficult to

determine the localisation direction of an image sound source. The listener evaluates that the image sound source is localised at the angle α_k . This stereophonic effect is a natural one.

In situation *B* in Fig. 5, the straight line mP_k has no common point with the cross-section H_k . However, the distance d_k is short ($d_k < d_g$, d_g — the limiting distance). The angle α_k is about -0.16π (-29°). A situation such as *B* can be generated for direct detection in an interior with good acoustic condi-

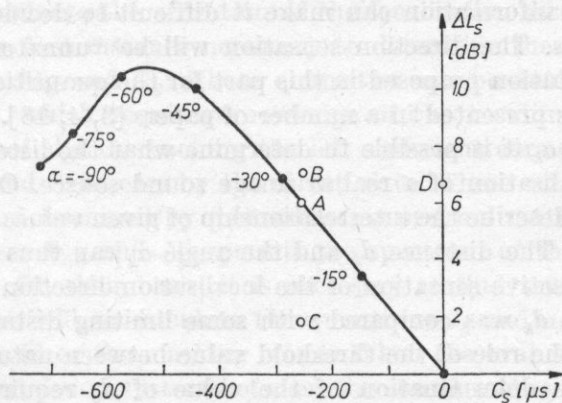


Fig. 5. Projections of the cross-section H_k and the straight lines mP_k on the plane $(\Delta L_S, C_S)$, $f_k = 1200$ Hz

tions. In such conditions, as a result of reflections from the walls of the interior reaching the listener's ears, there is a change (with respect to anechoic conditions) in the level difference ΔL_S . However, this change does not exceed 2 dB [10]. From psychoacoustic investigations, it follows that in such conditions localisation is good [11]. The angle at which the listener localises the sound source, is close to the angle α_i . Situation *B* can also be generated by a signal transmitted by a stereophonic system. It is then not difficult for the listener to determine the localisation direction of the image sound source.

In situation *C* the distance d_k is large ($d_k > d_g$). α_k is about -0.083π (-15°). Situation *C* can be generated for direct detection in an interior with poor acoustic conditions [10]. In such conditions it is difficult, sometimes impossible, to localise, and the listener may evaluate wrongly the localisation of the sound source [13].

In situation *D* the distance d_k is also large ($d_k > d_g$). The angle α_k is about -0.11π (-20°). Such a situation can occur for detection through a stereophonic system [10, 11]. Since the distance d_k (as in situation *C*) is large, the listener will find it difficult to determine the localisation direction of the sound source, as it is not important for the listener whether the given localisation parameters ΔL_{Sk} and C_{Sk} have been generated by signals reaching his ears directly from the source or through a stereophonic system.

The difficulties in determining the localisation direction of the sound source can be caused by "unnatural" interrelationships of the parameters of signals reaching the listener's ears, i.e. the frequency, the level difference ΔL_S and the time delay C_S . This can be an unnaturally large value of ΔL_S for a given frequency, e.g. for $f_k = 2$ kHz $\Delta L_{Sk} > 20$ dB, which in standard conditions does not occur for any angle α . This can be a zero value of the time delay C_{Sk} and a large value of ΔL_{Sk} at medium frequencies, for the zero time delay corresponds to the angle $\alpha = 0$, a large value of ΔL_S to a large negative angle α . The contradictory pieces of information can make it difficult to decide where the sound source actually is. The direction sensation will be "unnatural".

The interpretation proposed in this part for the quantities \bar{d}_k and α_k agrees with observations presented in a number of papers [3, 4, 13]. With the distance \bar{d}_k and the angle α_k it is possible to determine what the listener's decision will be as to the localisation of a real or image sound source. On the other hand, these quantities describe the interrelationship of given values of f , ΔL_{Sk} and C_{Sk} with the set H_k . The distance \bar{d}_k and the angle α_k can thus be objective measures of the subjective sensation of the localisation direction of a sound source.

The distance \bar{d}_k was compared with some limiting distance \bar{d}_g . The latter distance played the role of the threshold value between natural and unnatural localisations. The determination of the value of \bar{d}_g requires psychoacoustic investigations to be carried out. However, from the discussion given, it can be stated that localisation becomes more natural as the distance \bar{d}_k decreases.

The above considerations are concerned with a narrow spectrum band with a centre frequency f_k of a signal emitted by a sound source. Since a source emits most frequently signals with composed spectrum, the determination of the listener's decision as to the localisation of the sound source, requires that \bar{d}_k and α_k should be found for all the components of the spectrum of the signal detected by the listener. For direct detection in anechoic conditions, for all the spectral components the equation $\alpha_k = \alpha_i$ is satisfied, i.e. for all the components the source will be localised in its actual direction. At all frequencies the distance \bar{d}_k will be zero. In evaluating the localisation of a real sound source in reverberation conditions, \bar{d}_k will most frequently be non-zero and the angles α_k will be different from α_i . When the interior has good acoustic conditions, the distances \bar{d}_k will be short, the angles α_k close to the angle α_i . As the acoustic conditions worsen \bar{d}_k will increase and so will the scatter of the values of α_k about α_i . In determining the localisation direction of an image source, very diverse dependencies of α_k on frequency are possible (see Fig. 8). This depends above all on the way of transmitting the signal from the source to the listener's ears, i.e. on the properties of the stereophonic system [10, 11]. Therefore, in such a case, the use of the quantities \bar{d}_k and α_k is a very useful way of evaluating the localisation sensation.

The interrelating, by means of the distance \bar{d}_k and the angles α_k , of the

localisation parameters of signals reaching the listener's ears and the localisation sensation is the proposed criterion for evaluating the sensation of the localisation direction of a sound source.

7. Procedure of objective evaluation of the sensation of the localisation direction of a sound source

In order to evaluate the sensation of the localisation direction of a sound source, it is in the first instance necessary to determine the localisation parameters L_{SL} , L_{SR} , ΔL_S and C_S . These quantities can be determined from dependencies (1)-(4), which requires knowledge of the twodimensional autocorrelation functions of signals reaching the left and right ears and the delays in reaching the left and right ears by the direct wave. These correlation functions and delays can be determined by measurements or from mathematical models.

The parameters L_{SL} and L_{SR} permit the spectrum of the signal detected to be determined. On their basis, it is necessary to find the frequency bands for which it is justified to calculate the quantities d_k and α_k , i.e. in terms of the hearing threshold or the masking effect. When the evaluation is to be carried out on a transmission system between the sound source and the listener's ears, the spectrum of the signal emitted by the source should cover all the frequency range under consideration.

In evaluating a stereophonic system it is also necessary to consider the effect of a later sound being masked by an earlier one. This phenomenon was described in paper [13]. As a result of investigations, it was established that when for a given time delay the level difference between signals emitted by two sound sources exceeds some threshold value, the sound source emitting the signal with the lower level will not be observed by the listener.

Having definite localisation parameters ΔL_S and C_S and using a decision-making model represented by the set H , the distances d_k and the angles α_k should be determined for particular spectrum bands. Subsequently on the basis of d_k and α_k it is possible to carry out an objective evaluation of a given localisation sensation. The direction sensation improves as the angles α_k show less and less scatter about some mean value and as the distances d_k decrease. In addition, in evaluating the direction sensation generated by means of a stereophonic system, it should be considered whether the system provides the possibility of the image sound source being localised at any point of the base.

Apart from their dependence on frequency, the localisation parameters ΔL_S and C_S , and thus accordingly the quantities d_k and α_k , depend on a large number of other factors. In the case of a stereophonic system the localisation parameters are affected by such factors as the angles at which sound sources are localised with respect to the microphones, the characteristics of the micro-

phones and loudspeakers, the parameters describing reflections from the walls etc. [10, 11]. Thanks to the method proposed, the effect exerted by all of these factors on localisation can be analysed thoroughly.

8. Evaluation of a two-channel loudspeaker intensity stereo-system

Using the mathematical model of the stereophonic system under consideration [10, 11], on the assumption that: the detecting and transmitting interiors are anechoic in character, the stereophonic system is a symmetrical one, the microphones making up the coincidence microphone have cardioidal characteristics, the angle of the base is $\pi/2$ and that the source emits a signal described by the autocorrelation function $R_{xx}(t_1, t_2) = {}^2\delta(t_1, t_2)$, the localisation parameters were calculated. Because of the symmetry assumed for the stereophonic system, the time delay C_s is zero by identity. The dependence of the level difference ΔL_s on frequency for some angles at which the sound source is localised with respect to the coincidence microphone (the angle φ), is shown in Fig. 6.

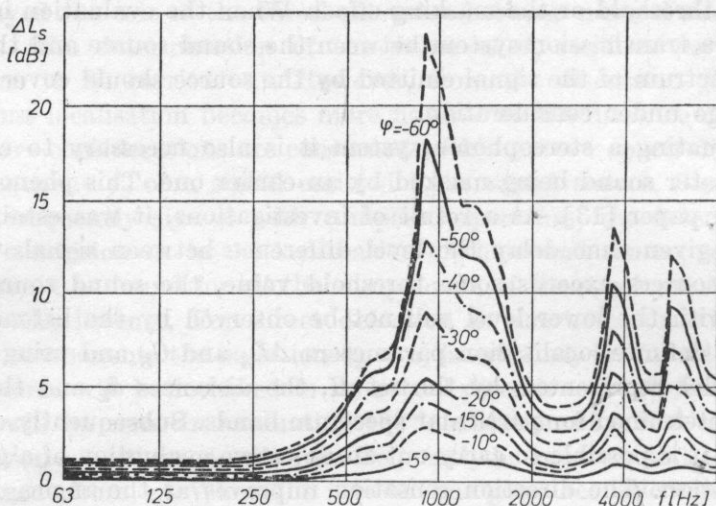


Fig. 6. Differences in the short-term spectral power levels, ΔL_s , for the chosen version of two-channel loudspeaker intensity stereophony, for various angles α

Subsequently, in keeping with the procedure shown in section 7, the distances d_k and the angles α_k were calculated. The results of the calculations are given in Figs. 7 and 8. On the basis of these results and those of calculations carried out for other variants of two-channel loudspeaker intensity stereo-system [10], it was established that:

— for the angles $\varphi \in (-\pi/6, \pi/6)$ at which the sound source is localised with respect to the microphone, the stereophonic system ensures such a representa-

tion in which the larger angle φ corresponds to the larger angle α_k at which the image sound source is situated with respect to the listener. As a result of this, there occurs the sensation of a "blurred" sound source emitting composed spectra. A point sound source will occupy part of the base,

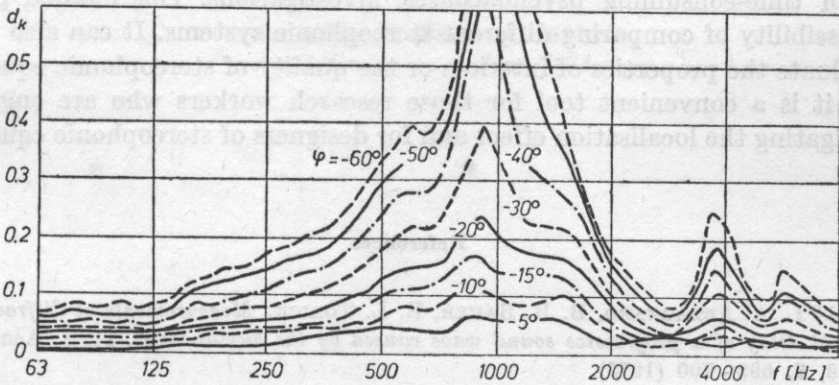


Fig. 7. The calculated distances d_k

- when the sound source emits a signal whose spectrum will be concentrated about some frequency, the image source changes its position on the base as this frequency varies,
- when the sound source emits a low-frequency signal, the image sound source is localised close to the centre of the base,
- in limited frequency bands image sound sources are localised beyond the base section.

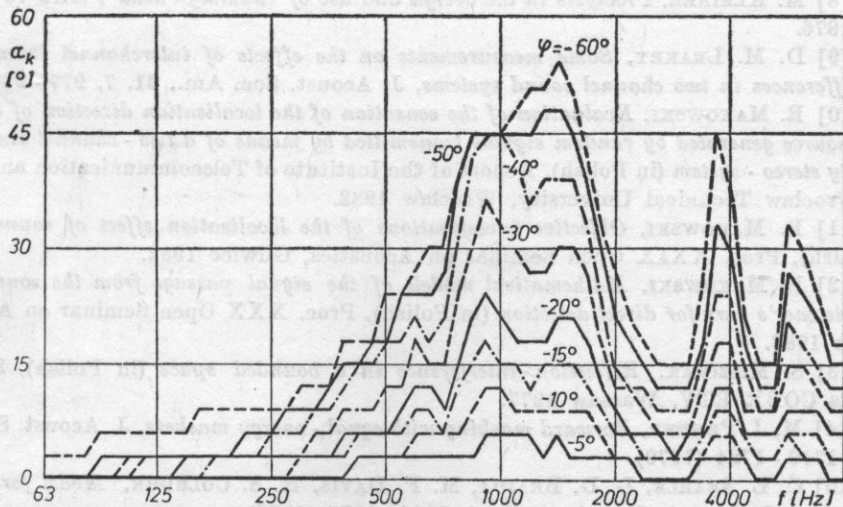


Fig. 8. The angles α_k of the localisation of the image sound source with respect to the listener

9. Conclusion

The method proposed represents a new approach to the evaluation of the localisation effect, permitting this evaluation to be carried out without the need of time-consuming psychoacoustic investigations. This method provides the possibility of comparing different stereophonic systems. It can also be used to evaluate the properties of interiors or the quality of stereophonic equipment. Thus, it is a convenient tool for those research workers who are engaged in investigating the localisation effect and for designers of stereophonic equipment.

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*Received on 17 February, 1983 ;
revised version on 7 June 1984.*

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This paper proposes a method for predicting the sound power level of a reduction line of a gas installation, based on a vibroacoustic cylindrical equivalent model. This model was verified experimentally on a real object. It permits approximate prediction of the sound power level of a reduction line, depending on its technological and structural parameters, particularly for medium and high frequencies.

1. Introduction

Prediction of the value of sound power radiated by sound sources on machinery and industrial facilities is one of the essential problems in machinery vibroacoustics. Attempts to develop methods for predicting the values of the power level of sound sources were undertaken for plates in the case of flexural resonance vibration [2] and for pipes involving turbulent water flow [1, 3].

The present authors are concerned with the problem of predicting values of the sound power radiated by sources with complex geometry, such as occur in gas pipelines. The considerations apply only to radiation of sound power by surface sources [4].

The principal object of this paper is to build and verify experimentally a model of radiation by a system of sound sources with complex geometry, where acoustic energy radiation is caused by gas flow in the process of pressure reduction. This purpose was carried out by forming an equivalent cylindrical