ARCHIVES OF ACOUSTICS
2, 4, 281 (1977)

DETECTABILITY OF PULSE DISTORTION IN AN ACOUSTIC LOUDSPEAKER FIELD

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This paper presents results of the evaluation of the distortion of transient signals produced by a set of loudspeakers. They were obtained on the basis of certain objective signal parameters and subjective tests. The separate objective parameters provided complex information on the degree of signal distortion and were correlated with the psychoacoustic ability of the sense of hearing to detect distortions. These parameters were related to the rising transients with complex spectra and to the decay transients with simple spectra.

1. Introduction

It was shown [7] that the behaviour of differential thresholds in time depends very little on the duration of transient for $t \geqslant 1$ s, while for shorter durations the threshold values increase. This is true for both rising and decaying transients of tone pulses.

The differential thresholds are independent of changes both in the tone pulse intensity, over a range from 40 to 100 phons, and for component frequencies in the pulsed signals above 400 Hz. From the detailed analysis of the psychoacoustic evaluation of differential thresholds of pulse tones with transient duration smaller than 1 s, it can be concluded that the organ of hearing detects differences in the transient durations primarily on the criterion of the rate of rise or decay of the signal intensity. A secondary criterion is the detection of differences in the intensity of audible signals. Similar results were obtained [7, 11] for complex tone and noise pulses.

Another important factor affecting the psychoacoustic evaluation of sound pulses is the shape of their envelope. Investigations of sound pulses with rising and decaying transients of the same duration but with different envelopes [2, 3, 4], provided initial information on the ability of the organ of hearing to detect, changes in the signal envelope. It was shown [7] that the differentiation of sound pulses with rising transients longer than 10 ms is related to the detectability of differences in the rising transients. For transients shorter than 10 ms, the detectability is associated with differences in the signal timbre. Similar experiments were performed for sound pulses with a "distorted" non-exponential envelope for the decay transient The decay signal transient should be either extended or shortened (at a level 40 phons below the maximum) by 10 to 20 % if these changes are to be detectable. There must be a time interval 70 to 300 ms [3, 4] between two successive maxima in the decaying transient envelope described by $\sin x/x$, in order to make them detectable. The amplitudes of these maxima are then 1/2 and 1/3 of their values in the steady state. The longer this time interval is, the higher the detecting ability becomes, with a peak detectability for a 250 ms time interval between the maximum values. In the author's opinion, this is related to the occurrence of a minimum in the auditory threshold of amplitude modulation (the modulation frequency being 2 Hz).

The results presented in this paper show that the transients play an important role in the psychoacoustic evaluation of sound pulses. They also permit the determination, in a general way, of the transient parameters (duration, envelope shape which are of vital importance in this evaluation.

2. The purpose of the paper

Within the framework of research on the dependence of the detectability of changes in sound pulse forms on their objective parameters, we have made an attempt to determine the parameters that would be capable of providing information on the degres of distortion of the entire signal or of selected parts. The primary selection of these parameters was aimed at obtaining detailed information on the role of the rising and decaying transients in the psychoacoustic evaluation of signals taken as a whole.

The problems, which relate the psychoacoustic evaluation of sound pulses with the objective parameters of the pulses, constitute a group of problems dealing in general with the subjective evaluation of complex acoustic fields in which the effects of diffraction, interference, and amplitude and phase non-uniformities produce major distortions in the acoustic signals.

We describe here the results of an attempt to determine the signal parameters by evaluating the acoustic field of a set of loudspeakers[1] (2 broadband loudspeakers, 1 high-pitch loudspeaker enclosed in a housing 0.250 m^3). We have objectively determined the degree of distortion of the sound pulses in both transient and steady states at particular points in the field and analysed the psychoacoustic detection of these distortions. From these results conclusions have been drawn on the interrelation between the psychoacoustic evaluation and the objective parameters of the signals under consideration.

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[1] A set of loudspeakers produced by firm Klein-Hummel, type Studio-Regielautsprecher 02.

3. Measurements

3.1. The measuring system

The schematic diagram of the measuring system employed for the assumed investigation program is shown in Fig. 1.

The set of loudspeakers was excited by various acoustic generators in accordance to a selected measuring method. The signal from the loudspeakers was received simultaneously by two microphones, M-2 being at a fixed point, while M-1 was moved from one place to another as the measurements were in progress. The signals received by the microphones were amplified and fed to the recording systems. In the case of pulse methods, these signals were first recorded on magnetic tape and transformed by means of an A/C converter. The time –response of the loudspeaker of a given pulse excitation was obtained on paper tape at the output of punch unit.

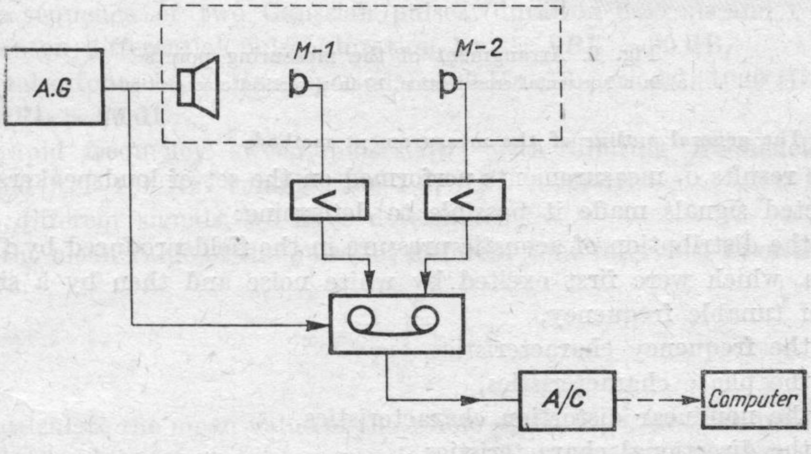

Fig. 1. Block diagram of the measuring apparatus

AG – acoustic generator, *M*-1, *M*-2 – measuring microphones, *AC* – analog-to-digital converter, *COMPUT* – IBM computer

3.2. Measuring conditions

All tests were performed in an anechoic chamber. Fig. 2 shows the arrangement of the measuring points and the location of the different loudspeakers in the set. The measuring points were located along three axes. Two axes *a* and *c* were coincident with the axes of the loudspeakers, while the third *b* axis was symmetrical to the axes of the two loudpseakers. The signals which excited the loudspeakers were selected in accordance with the programme of investigation.

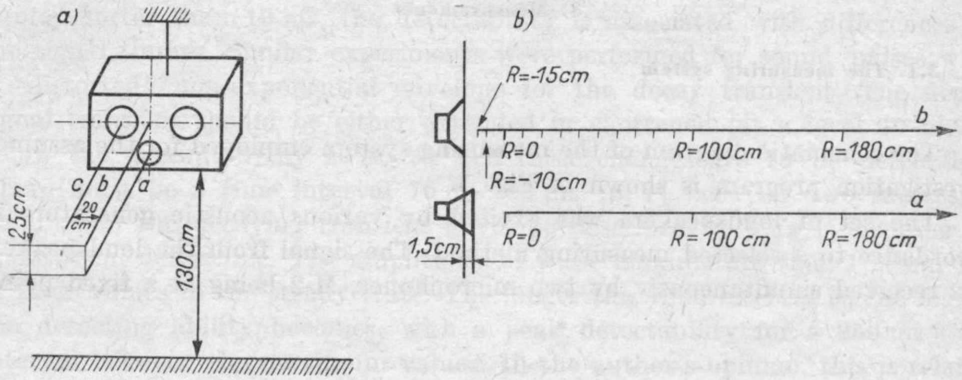

Fig. 2. Arrangment of the measuring points
a) in a horizontal cross-section, b) in a vertical cross-section

3.3. The general outline of the measurement methods

The results of measurements performed on the set of loudspeakers excited by selected signals made it possible to determine:

— the distribution of acoustic pressure in the field produced by the loudspeakers, which were first excited by white noise and then by a sine-wave signal of tunable frequency,

— the frequency characteristics,

— the phase characteristics,

— the nonlinear distortion characteristics,

— the directional characteristics.

Since measuring methods for the above quantities are well known, they will not be described in this paper.

The reaction of the loudspeakers excited by pulse signals were recorded on the basis of the following three signals:

signal 0 — obtained from a generator and recorded by a purely electrical method,

signal 1 — (acoustic) received by microphone M-1 located at different points of the acoustic field,

signal 2 — (acoustic) received by microphone M-2 located at a fixed measuring point of the acoustic field (a, $R = 180$ cm). The magnetic tape recordings of the reaction of the loudspeaker to a pulse excitation were used to prepare psychoacoustic tests. A series of measurements were made for a single location of microphones in order to eliminate possible inaccuracy related to the positions of the measuring points.

3.3.1. *Methods employed for the unsteady state*

From the theoretical point of view, the maximum information on a transmission system is obtained by employing a Dirac delta pulse signal [8]. However such a experiment is very difficult to perform if an acoustic pressure level

$SPL = 70$ to 90 dB is required at a distance 1-2 m from the loudspeaker plane. For this reason, experimenters select pulsed signals of a finite duration strictly related to the frequency under test [10]. The other parameters of the exciting pulse such as rise and decay times, and shape and amplitude of the envelope, are selected in accordance with the testing programme of the transmitting system [5]. The following pulses were used in these experiments:

– a pulse with a rectangular envelope, of durations 1 ms or 0.25 ms; providing acoustic pressure levels $SPL = 70$ and $SPL = 90$ dB at a distance of 1.80 m from the loudspeaker plane,

– pulses of Gaussian envelope, durations 1 ms (rise and decay times 0.3 ms), and 1.5 ms (rise and decay times 0.5 ms); $SPL = 80$ dB.,

– a sequence of two Gaussian pulses (duration 0.25 ms and 1 ms) and one Thomson differential pulse (duration 1 ms); $SPL = 80$ dB,

– pulse tones at 80 Hz (3 periods), 800 Hz (16 periods), 1020 Hz (16 periods); SPL = 80 dB,

– rapid frequency sweep pulses [6] with limiting frequencies $f_1 = 30$ Hz and $f_2 = 150$ Hz; tuning time 1 s; $SPL = 80$ dB.

For the different signals, we have determined:

– the mean value of the pulse for different time intervals, T, of the signal

$$\overline{x} = \frac{1}{T}\int_0^T x(t)\,dt.$$

To calculate the mean value of the whole signal $\overline{x}_{11}$[2]), we took into account all the data, while for the rising transient $\overline{x}_{01}$ only the data from 0 to N was used, the mean value being determined for a given kind of pulse recorded at a the measuring point a, $R = 180$; and being related to the maximum walue of the amplitude after the rising transient has heen completed. The mean value of the decay transient $\overline{x}_{13}$ was determined for the range from N to Z, where Z was measured at the point a, $R = 180$ and related to the background level. The mean values of the transients of the signals $\overline{x}_{22}$, $\overline{x}_{02}$, $\overline{x}_{23}$ recorded at the remaining measuring points were calculated for unchanged values of N and Z for a givan type of pulse. With this approach, it was possible to compare the corresponding mean values of signals recorded at different measuring points by determining the differences $\Delta\overline{x}$.

The standard deviation S from the mean value within the time intervals mentioned above is given by:

$$S = \sqrt{\frac{1}{T}\int_0^T [x(t) - \overline{x}]^2\,dt}.$$

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[2]) Indices used for signal parameters are described in subsection 4.2.

The standard deviation S was assumed to be a measure of the variation of the process. The values of S are equal to zero only if all the numerical data describing the time interval of a pulse are identical. The higher the value of S (which is always positive), the greater is the dispersion of the numerical data describing a given signal. As for the mean values, the differences ΔS for the standard deviations were also determined.

The coefficient of skewness Q for time interval T is given by:

$$Q = \frac{1}{T} \frac{\int_0^T [x(t) - \overline{x}]^3 \, dt}{S^3}.$$

The coefficient of skewness of a distribution Q provides information on the shift of the distribution with respect to the zero-axis. If the greater part of the distribution lies on the positive side of the axis, then Q is positive, since the weighted sum of the third powers of large positive deviations is greater than the sum of the third powers of negative deviations. In this case, the skewness is positive. A negative skewness corresponds to such a distribution that the greater part of the process lies below the axis. For the symmetrical distribution, the sum of the positive third powers is equal to the sum the negative third powers and thus $Q = 0$. ΔQ was also determined for this parameter.

By direct comparison of signals 1 and 2, we have determined their correlation coefficient ϱ,

$$\varrho_{12} = \frac{\mathrm{cov}_{12}}{S_1 \cdot S_2},$$

where cov_{12} is a covariance

$$\mathrm{cov}_{12} = \frac{1}{n+1} \sum_{i=0}^{n} [x_i(t) - \overline{x}_1] \cdot [x_i(t) - \overline{x}_2]$$

and S_1, S_2 are the standard deviations.

The correlation coefficient is positive if the two characteristics either decrease or increase at the same time, and is negative if one characteristic increases while the other decreases. If the characteristics are independent of each other, the correlation coefficient is zero. This coefficient takes the value -1 or $+1$, if the two characteristics are interrelated in the form of a simple linear function. If a number od values of $x_2(t)$ correspond to a fixed characteristic $x_1(t)$, the correlation coefficient is either a fraction or zero. The correlation coefficient was determined here for the time intervals mentioned above of the signals to be compared.

3.3.2. The methods employed for psychoacoustic tests

As has already been mentioned in section 2, the psychoacoustic tests were aimed at determining the relation between the changes in particulas objective parameters characterizing the degree of distortion of the pulses recorded at different points of the acoustic field, and the ability of the human sense of hearing to detect these changes.

The psychoacoustic test involved successive pairs of pulses which were always repeated three times. The observer was asked to listen to a pulse pair repeated three times and to give an answer to the question: Are the signals of the pulse pairs identical? The answer could be either «yes» or «no». If the observer hesitated he should answer «no». Fig. 3 presents the time diagram of the test. The succession of signals in a pair, selection of signals and signal reception points were determined at random. For the set of loudspeakers, the tape recordings were taken at the five points marked in Fig. 2 (c, $R = -15$; b, $R = -15$; a, $R = 0$; b, $R = 100$; c, $R = 180$) and this provides $2 \times 5 \times 10$ signal pairs. These signals were heard by three normally hearing observers over 10 successive days using dynamic earphones type HD 414 produced by the firm of Sennheiser.

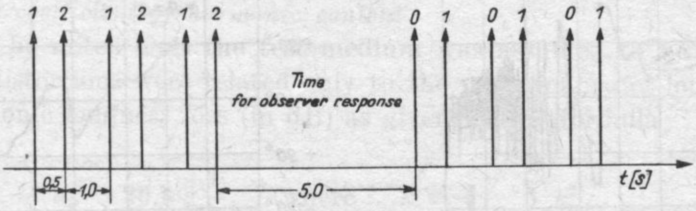

Fig. 3. Succession of acoustic signals in a psychoacoustic evaluation test with a 5 s break for the observes answer

0 – acoustic signal from generator, electrically recorded on magnetic tape, 1 – acoustic signal received by microphone M-1 positioned at successive points of the loudspeaker radiation field, 2 – acoustic signal received by microphone M-2 positioned at a fixed measuring point (a – axis, distance between microphone and loudspeaker R – 180 cm)

4. The results of the investigations

The changes in the sound signal structure in the acoustic field produced by a set of loudspeakers were evaluated with loudspeakers excited by both continuous and pulsed signals.

4.1. The continuous wave excitation

4.1.1. Amplitude and phase characteristics

The amplitude and phase characteristics were measured at different points of the acoustic field produced by the set of loudspeakers (Fig. 4). It was found that these characteristics depend significantly on the position of the measuring point.

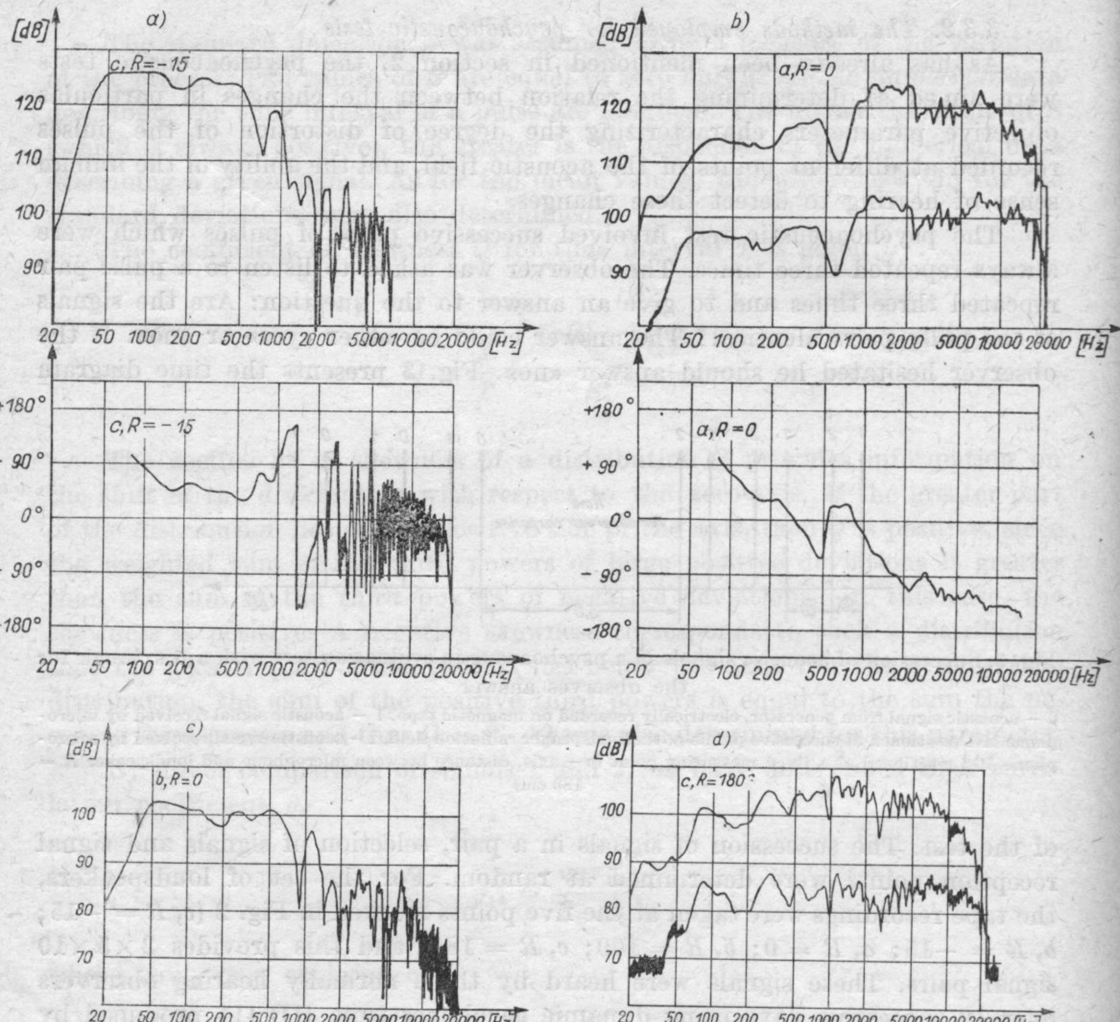

Fig. 4. Amplitude and phase characteristics of the tested set of loudspeakers. Measuring points were located as follows

a) c-axis, distance $R = -15$ cm; b) a-axis, $R = 0$ cm; c) b-axis, $R = 0$ cm; d) c-axis, $R = 180$ cm

The diagrams 4b and 4d represent amplitude characteristics for two intensities corresponding to acoustic pressure levels of $SPL = 70$ dB (the lower curve) and $SPL = 90$ dB (the upper curve) at the point a, $R = 180$; the remaining curves (Figs 4a, 4c) correspond to the case $SPL = 90$ dB

Differences occur not only between the near and far fields but also between axes at a fixed distance from the measuring points and the front plane of the set of loudspeakers.

The selected amplitude and phase characteristics shown in Fig. 4 indicate that there are major differences in both the band width and nonuniformity of the transfer function. It is known that the nonuniformity of the transfer function results in amplitude distortions. Its effect on frequencies is obvious. Nonuniformity also causes major distortions in the wave envelope (distortion of the pulse shape [9]). It follows from analysis of the phase characteristics that they are nonlinear. Thus different frequency components are subject to different attenuation and this results in additional distortion of the signal

4.1.2. The coefficient of harmonic content

It should be noted that the test medium was assumed to be linear and all nonlinear distortions were related only to the performance of loudspeakers.

The harmonic loudness loss (in dB) as given by the formula

$$a_k = 20\log\frac{1}{k}, \quad \text{where} \quad k = \sqrt{\sum_{n=2}^{N} k_n^2}, \tag{1}$$

was measured at points located on different axes of the set of loudspeakers and at different distance from its front plane. In (1), k is the coefficient of harmonic content (on a linear scale) and n is the degree of distortion. From the data obtained we have

$$a_k = \begin{cases} z(L) & \text{with} \quad f = \text{const}, \\ g(f) & \text{with} \quad L = \text{const}, \end{cases} \tag{2}$$

where f is the signal frequency, and L is the acoustic presure level for a measuring point located on the axis of the set loudspeakers (a, $R = 180$ cm).

We also have the relations of the type

$$a_k = \begin{cases} S(R), & f = \text{const}, \quad L = \text{const}, & \text{(3a)} \\ j(r), & f = \text{const}, \quad L = \text{const}, & \text{(3b)} \end{cases}$$

where R is the distance between the measuring point and the front plane of the set of loudspeakers, and r is the distance between the measuring point and the axis of the set of loudspeakers.

It is known in general how the harmonic loudness loss depends on the acoustic pressure level and signal frequency. The dependence of the quantity a_k, and the parameters related to the position of the measuring point within the radiation field of the set of loudspeakers, will be discussed.

Specimen results of the measurement of a_k as a function of R (Table 1) indicate that the relation described by (3a) is appropriate, especially at higher frequencies.

Table 1. Figures of nonlinear distortion of the 3rd order coefficient as a function of the frequency of the loudspeaker excitation signal and the position of the measuring points

SPL [dB]	*f* [Hz]	Axis	*R* [cm]	a_{k3} [dB]
90	80	*C*	180	48
			10	45
			−10	46
			−15	45
	1*k*	*C*	180	50
			10	43
			−10	48
			−15	47
	5*k*	*C*	180	55
			10	50
			−10	>70
			−15	>70

To interprete the processes described by expression (3), it is possible to utilize the factors affecting the forms of amplitude and phase characteristics.

The recorded differences in the amplitude and phase characteristics are reflected in the changes of the coefficient of harmonic content in view of the related signal distortions. In the author's opinion, numerous problems arise in the determination of the relationship between the psychoacoustical detectability of changes in the form of the acoustic processes and the objective parameters related to the type of signals in a piece of music or speech. These problems involve testing material, its description by means of objective methods and the ability of the observers to memorize the compared test samples. For this reason, pulse methods were used for the tests described, which are regarded as a first step to the whole problem under consideration. The set of loudspeakers was excited by pulses with different envelope forms, different times of rise and decay, and different spectra. It was intended to use the signals most frequently encountered in practice, and to perform a detailed analysis of the selected pulse parameters by objective methods.

4.2. Pulse signal excitation

In subsection 3.3.1 the detailed information is provided on the pulse signals used in the tests. The diagrams given in Fig. 5 illustrate the responses of the set of loudspeakers for various pulse excitations recorded at particular points in the acoustic field. These responses were processed in the A/C converter and in the computer. All the calculations described in subsection 3.3.1 were carried out each of the responses.

Specimen computer results for the sound pulses which were then used for the psychoacoustic tests are given in Table 2. The three groups of columns

Table 2. Values of the calculated parameters and the psychoacoustic evaluation of pulse signals from the set of loudspeakers

			Data of total signal				Data of rising signal transient				Data of decaying signal transient				Psychoacoustic evaluation Answer «yes» in %
Kind of pulses	Axis	R	$\bar{x}_{11}/\bar{x}_{22}$	S_{11}/S_{22}	Q_{11}/Q_{22}	ϱ	$\bar{x}_{01}/\bar{x}_{02}$	S_{01}/S_{02}	Q_{01}/Q_{02}	ϱ_{01}	$\bar{x}_{13}/\bar{x}_{23}$	S_{13}/S_{23}	Q_{13}/Q_{23}	ϱ_{02}	
Sequence of pulses	*a*	180	510	108	−0.88	—	518.0	194	−0.71	—	485	54	−2.5	—	—
	b	−15	511	100	−1.1	0.85	530	185	−94	0.87	481	26	−0.08	0.62	60
	b	−15	507	160	−0.55	0.44	596	215	0.17	0.54	490	93	0.33	0.08	20
	d	0	510	122	−0.20	0.78	540	221	−0.48	0.81	470	37	0.82	0.54	20
	b	100	509	97.6	−0.67	0.91	526	181	−0.64	0.94	488	35	−0.37	0.68	100
	c	180	512	102	−0.21	0.92	533	187	−0.43	0.94	490	46	−0.82	0.74	90
Gauss shape	*a*	180	509	66.8	−1.35	—	628	103.4	0.41	—	492	105	−0.97	—	—
	c	−15	510	97.6	−0.44	0.64	571	69.2	0.97	0.92	510	158.9	−0.30	0.67	90
	b	−15	509	152.2	0.09	0.43	532	18.4	0.72	0.94	547	201.5	−0.01	0.62	0
	a	0	511	85.4	−1.07	0.87	643	90.8	−0.01	0.63	506	132.7	−0.81	0.86	30
	b	100	510	68.8	−1.57	0.93	653	110.4	0.17	0.95	501	105.7	−1.29	0.93	90
	c	180	509	62.8	−1.33	0.95	631	97.5	0.26	0.95	500	96.6	−1.03	0.95	90
Tone pulse (80 Hz)	*a*	180	523	118.6	0.37	—	734	112.7	−0.47	—	501	152	0.25	—	—
	c	−15	524	81.9	2.18	0.83	730	109	−0.49	0.98	514	86.2	1.31	0.81	90
	b	−15	526	120.6	0.24	0.62	716	102.8	−0.46	0.98	521	158.5	−0.03	0.52	60
	a	0	522	89.8	0.90	0.76	712	100.4	−0.49	0.98	513	109.0	0.11	0.70	40
	b	100	523	106.9	1.20	0.88	746	119.1	−0.49	0.98	511	128.4	0.71	0.88	80
	c	180	522	106.4	0.53	0.95	719	102.2	−0.49	0.98	506	131.6	0.46	0.97	100
(800 Hz)	*a*	180	513	21.2	4.75	—	646	105.8	−0.58	—	518	45	0.36	—	—
	c	−15	514	23.8	4.90	0.58	571	36.6	−0.82	0.85	536	60.3	1.31	0.58	—
	b	−15	513	58.2	1.24	0.48	617	79.5	−0.63	0.85	518	141.2	0.86	0.52	90
	a	0	511	32.2	2.76	0.72	705	146.5	−0.47	0.85	516	77.0	−0.12	0.61	0
	b	100	514	23.5	5.73	0.52	648	132.1	−0.34	0.83	523	51.6	1.77	0.34	90
	c	180	514	25.8	−2.63	0.45	644	111.9	−0.45	0.85	525	46.0	1.17	0.31	90

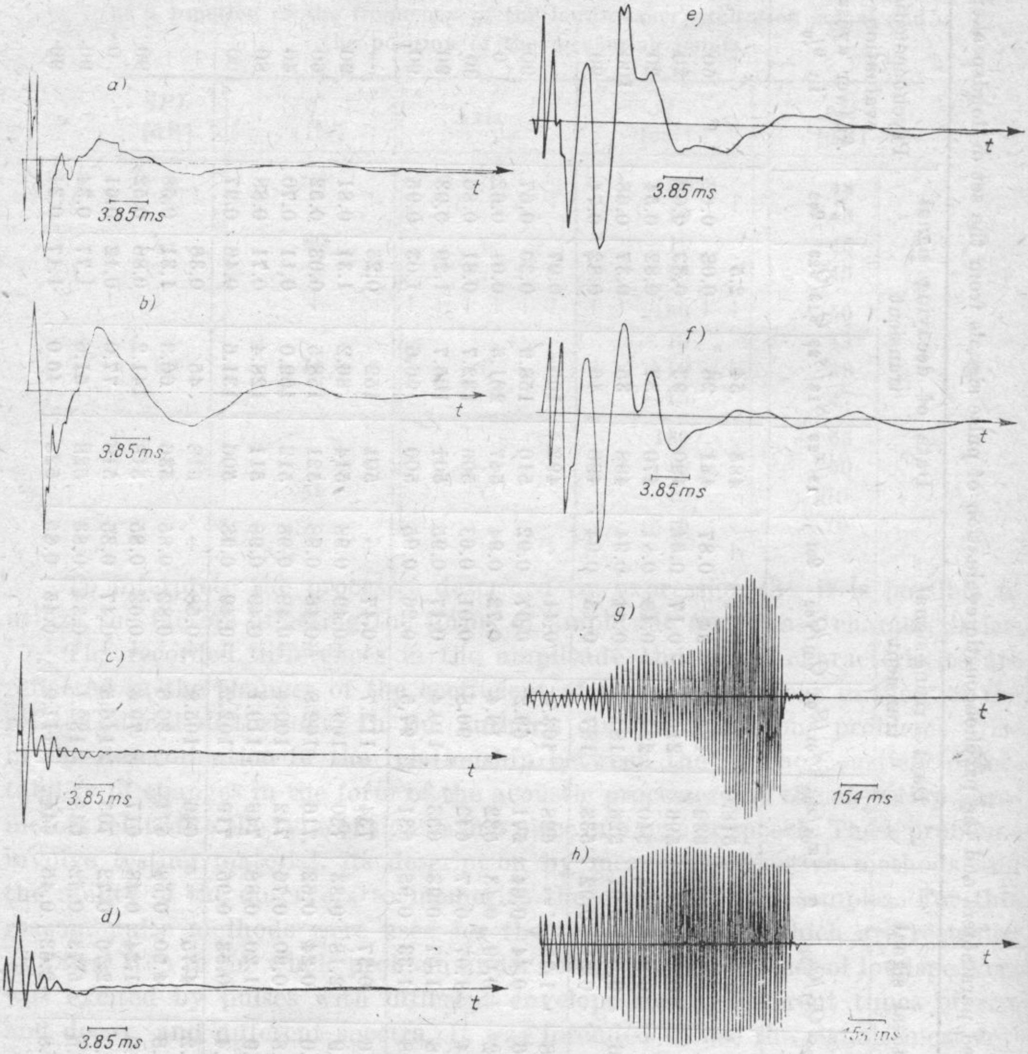

Fig. 5. Responses of the set of loudspeakers to various types of pulse excitation; these responses were recorded at selected points in the acoustic field

Exciting pulses: a) rectangular pulse, duration 1 ms, measuring point a, $R = 0$; b) rectangular pulse, duration 1 ms, measuring point b, $R = 10$; c) tone pulse of frequency 900 Hz, measuring point a, $R = 0$; d) tone pulse of frequency 800 Hz, measuring point c, $R = 100$; e) sequence of pulses, measuring point a, $R = 0$, f) sequence of pulses, measuring point a, $R = 100$; g) rapid frequency sweep pulses with limiting frequencies $f_1 = 30$ Hz and $f_2 = 150$ Hz, measuring point a, $R = 0$; h) rapid frequency sweep pulses with limiting frequencies $f_1 = 30$ Hz and $f_2 = 150$ Hz; measuring point a, $R = 180$

include the results calculated for the whole signal, and its rising and decaying transients recorded at the measuring point a, $R = 180$, and at corresponding points in the acoustic field of the set of loudspeakers. The parameters $\bar{x}$, S, Q characterize each signal independently, while the parameters $\Delta\bar{x}$, ΔS, ΔQ and ϱ describe the differences between the signals recorded at different points of the acoustic field and their relation to the reference signal recorded at the measuring point a, $R = 180$. The indexes of the parameters are defined as:

11 – the whole signal recorded at the measuring point (a, $R = 180$),
22 – the whole signal recorded at particular measuring points
01 – the rising transient of the signal recorded at the measuring point (a, $R = 180$),
02 – the rising transient of the signal recorded at particular measuring points,
13 – the decaying transient of the signal recorded at the measuring point (a, $R = 180$),
23 – the decaying transient of the signal recorded at particular measuring points.

The numerical values of the selected objective parameters of the pulse signals recorded at various points in the acoustic field of the set of loudspeakers have been compared with the recorded distortions of these signals. It can be concluded:

1. The mean values $\bar{x}_{11}$ and $\bar{x}_{22}$ determined for the total duration of the time varying signals recorded at different measuring points are very close to each other; they provide no information on the signal distortion.

2. The differences in the mean values $\Delta\bar{x}_{02} = \bar{x}_{02} - \bar{x}_{01}$ related to the rising transients and $\Delta\bar{x}_{13} = \bar{x}_{23} - \bar{x}_{13}$ related to the decaying transients, of the reference and other signals provide initial information on the distortion of the rising and decaying transients.

3. The numerical values of the differences ΔS and ΔQ for succesive rising and decaying transients of the different signals provide information on the degree of distortion, the sign of the difference ΔQ determining the direction of deviation of the distortion.

4. The correlation coefficients ϱ, ϱ_{01}, ϱ_{02} related respectively to the entire time intervals of the different signals, and their rising and decaying transients, contribute general information on the degree of distortion only for signals with complex spectra (e.g. a sequence of pulses, Gaussian pulses).

4.3. The psychoacoustic evaluation of pulse distortion

The selected sound pulses were used for psychoacoustic evaluation on the basis of the test described in subsection 3.3.2. It was intended to investigate the question of the extent to which the objective parameters are related to the ability of the organ of hearing to detect distortion of the pulses. Analyzing the detailed measured results, part of which are given in Table 2, we have assumed the number of «yes» answers $z \geqslant 70\,\%$ of all answers, to be the criterion of significance. In this way, if $z \geqslant 70\,\%$ the observer considers the pulses contained in a pair to be identical and he detects no distortion in either of the two compared signals. Using this criterion, it is possible, for instance, to state that the signals produced by a set of loudspeakers excited by a sequence of pulses and recorded at various points in the accoustic field are not identical (one has been distorted) for both $z = 60\,\%$ and $z = 20\,\%$ (see Table 2).

It follows from the table that the selected objective parameters x, S, Q, ϱ assume such values that for the compared «sequence of pulses», psychoacoustic evaluation can be based on the following statements:

1. The value of the correlation coefficient ϱ connects the entire durations of the compared signals and for $\varrho \geqslant 0.90$ the «yes» answers were estimated at $z \geqslant 90\,\%$ (i.e., if $\varrho \geqslant 90$, then the compared signals are considered to be identical).

2. The correlation coefficients ϱ_{01}, ϱ_{02} relate the corresponding rising and decaying transients $z \geqslant 70\,\%$ was determined for $\varrho_{01} \geqslant 0.94$ and $\varrho_{02} \geqslant 0.68$ (if $\varrho_{01} \geqslant 0.94$ and $\varrho_{02} \geqslant 0.68$ the compared signals were considered to be identical).

3. The psychoacoustic evaluation used the differences of the mean values $\Delta\bar{x}_{02} = \bar{x}_{02} - \bar{x}_{01}$, $\Delta\bar{x}_{13} = \bar{x}_{23} - \bar{x}_{13}$ and the coefficients of skewness $\Delta Q_{02} = Q_{02} - Q_{01}$, $\Delta Q_{13} = Q_{23} - Q_{13}$. The «yes» answers $z \geqslant 70\,\%$ were recorded for $\Delta\bar{x}_{02} \leqslant 15$ and $\Delta Q_{02} \leqslant 0.28$, and $z \geqslant 70\,\%$ for $\Delta\bar{x}_{13} \leqslant 5$ and $\Delta Q_{13} \leqslant 1.7$.

It should be remarked that the numerical values are not unique. Considering the parameters as functions of the position of the measuring points in the accoustic field of the set of loudspeakers, it can be found that they depend on the distance from the set of loudspeakers and from the measuring axis (e.g. for $c, R = -15$ wo have $\varrho = 0.85$ and $z = 60\,\%$, and for $b, R = -15$, $\varrho = 0.44$, $z = 20\,\%$). It follows from the psychoacoustic evaluation of signals recorded at points close to, and far away from the plane of the set of loudspeakers, that the signals have little similarity ($z \leqslant 60$ for $R \leqslant 0$ and $z \geqslant 90\,\%$ for $R \geqslant 100$).

From a similar analysis performed for Gaussian pulses (duration 1.5 ms, rise and decay times 0.5 ms) it can be concluded that:

1. For «yes» answers $z \geqslant 70\,\%$, the experiments give $\varrho \geqslant 90$; this relation is not unique since it was recorded that for $z = 90\,\%$, the other parameters were $\varrho = 0.64$, $\varrho_{01} = 0{,}92$ nad $\varrho_{02} = 0.67$. These responses were then evaluated psychoacouslicly mainly on the basis of the physical similarity of their rising transients (very high ϱ_{01}). This thesis is confirmed for example by high values $\varrho = 0.82$ for $z = 30\,\%$ and with $\varrho_{01} = 0.63$ and $\varrho_{02} = 0.86$. Thus, in spite of high values of ϱ and ϱ_{02} a smaller ϱ_{01}, the psychoacoustic evaluation in «no» (the compared signals are not identical), i.e. it is stated that one pulse is distorted.

2. The dispersion of the values $\Delta\bar{x}_{02}$ is not expressed in the values of ϱ_{01}, but changes in $\Delta\bar{x}_{13}$ affect ϱ_{02}.

3. Changes in ϱ_{01} and ϱ_{02} are related to changes in ΔS_{02} and ΔS_{13}.

From the specimen measured results obtained for tone pulses with frequencies of 80 Hz (3 periods) and 800 Hz (16 periods), it is possible to draw the following conclusions:

1. For higher frequency signals $f = 800$ Hz, the coefficients ϱ are small and their values are not uniquely related to the results of the psychoacoustic evaluation; such relations were found for low frequency signals ($f = 80$ Hz).

2. The value of ϱ_{01} is approximately constant for all the tested signals ($\varrho_{01} \geqslant 0.80$) and is not related to the psychoacoustic evaluation of the signals. This value is independent of changes in the parameters $\Delta\bar{x}_{02}$, ΔS_{02}, ΔQ_{02}.

3. For lower frequencies ($f = 80$ Hz), the psychoacoustic evaluation is strictly related to ϱ_{02} (the number of «yes» answers $z \geqslant 70\,\%$ is determined for $\varrho_{02} \geqslant 0.80$). The relation $\varrho = \text{const.}$ is obvious the pulses are sections of sine-wave tones; only major distortions of these signals changed the values of ϱ (e.g. for $f = 80$ Hz) and this was reflected in the psychoacoustic evaluation of the signals.

4. The differences in the number of «yes» answers are related to changes of $\Delta\bar{x}_{02}$ for $f = 800$ Hz.

5. Conclusions

From the results described in this paper it is possible to draw the following final conclusions:

1. The correlation coefficients ϱ are most strongly related to the psychoacoustic evaluation of the different signals.

2. It was found that the observes are most able to detect differences in the sound pulses with a wide frequency spectrum, for correlation coefficients $\varrho \leqslant 0.90$.

3. The high values of the correlation coefficient ϱ for sound pulses with a wide frequency spectrum cause high values of the coefficients ϱ_{01} which are related to the rising transients of these signals.

4. For tone pulses with a spectrum, in which one frequency is clearly determined, the correlation coefficients ϱ provide no information of the signal. However, signals of this type are distorted by a set of loudspeakers (especially, in the low frequency range) and, consequently, the correlation coefficient of the signal is $\varrho \neq \text{const.}$ In the psychoacoustic evaluation, the distortion of a signal is detected for a correlation coefficient $\varrho \leqslant 0.80$.

5. The high value of the correlation coefficient ϱ for a tone pulse which has been distorted for example in transmission causes high values of the correlation coefficient ϱ_{02} which is related to the decaying transients of the signal.

Based on conclusions 1 to 5, it is possible to formulate two more, general, conclusions:

6. In the psychoacoustic evaluation of sound pulses with complex spectra, the rising transients play the most important role, while in the case of tone pulses with simple spectra (limited, moreover, to the low frequency range), the decisive role is played by the decaying transients.

7. It is true that the objective parameters and the results of psychoacoustic evaluation depend on the positions of the measuring points in the acoustic field as may be expected. However, this dependence is not a simple function of the distance between the measuring points and the plane of the set of loudspeakers. (This problem will be discussed by the author in another paper «Analysis and the principles of perception of signals with a varying structure in a complex acoustic field» which is in preparation.)

Acknowledgement. I am very grateful to Prof. Dr. H. Ryffert for the discussion on the present paper.

I am also indebted to prof. dr E. Zwicker and to the Humboldt Foundation for the assistance in conducting the research in the Institute of Electroacoustics in Münich (München).

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Received on 29th December 1977