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QUASI-NONLINEAR DISTORTION OF SIGNALS IN CLOSED SPACE FOR AN UNSTEADY STATE

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This paper discusses the possibility of signal distortions occurring in closed space for the rise and decay of signals. A theoretical analysis is conducted for a simplified case in which the character of phenomena is analyzed in relation to the value of the ratio of the time delay between successive signals and their period.

The later part of this paper gives experimental results which confirm the possibility of distortions occurring in real conditions. It also discusses the possibility of the occurrence of phase, frequency and amplitude modulation and gives examples of signal shape distortion in its rising, obtained in two concert halls in the Academy of Music, Warsaw.

A subjective analysis was performed of the audibility of distortions in simulated signals which were programmed with a regular kind of distortion. The results of distortion audibility in % and the subjective analysis of the audibility of distortions in real conditions have shown that these distortions can be perceived with their short duration of the order of several or a dozen or so milliseconds.

1. Introduction

Among the variety of criteria for evaluating the acoustic quality of concert halls, theatres and auditoria, which will here be called auditoria, some refer to the initial time of sound rise and decay in enclosures.

Both in evaluating the masking effect of the first reflection with respect to the direct wave [6, 16] and in energy analysis of the first reflection with respect to all reflected waves [11, 13], these criteria involve a value of 50 ms calculated from the arrival or decay time of the direct wave as an essential value from the point of view of subjective analysis. The arbitrarily assumed notions of the rise time *TR* and the early decay time *EDT* of signals contain, according to Jordan [7], essential data characterizing the properties of interiors.

These criteria can in general be called macroscopic, since they involve primarily energy relations without investigating the structure of signals [12].

It is well known that reflected waves affect the structure of a signal in an enclosure for an unsteady state. However, investigations in this field, which could be called microscopic approach investigations, have not been performed extensively so far. This approach was suggested in the 1950s and partly developed in the following years [2, 4, 14]; however, the lack of subjective evaluation methods for a „microscopic distinguishing" of the structure of the acoustic field failed to provide conditions to show the necessity for these investigations to be carried on.

The recently developed methods for subjective evaluation of acoustic signals and the use of FFT analysis of signals have encouraged the present authors to continue the microscopic approach investigations mentioned above.

The starting point in the present paper is the treatment of the delays with which particular reflected waves reach the observation point at the rise time of a signal as nonlinear, step-like time functions [2, 3]. For particular reflected waves these functions take a value of zero in the period in which the wave of a given reflection has not yet reached the observation point and a value of one when the wave of a given reflection has reached the observation point.

Assuming the linearity of phenomena in the amplitude domain, it is possible to superpose particular reflected waves, each of which contains a relevant step-like, i.e., nonlinear, time function resulting from the delay in its arrival at the observation point. As a result of the nonlinearity mentioned above, when the sound rises or decays at the observation point, the resultant signal shows the properties of a distorted signal.

Since the nonlinear properties of the resultant signal result from time characteristics and not from amplitude ones, the distortions thus caused will be called quasi-nonlinear distortions.

It is much more difficult to define nonlinear distortions resulting from nonlinear amplitude dependencies for unsteady-state than those for steady-state conditions. It is still more difficult to define quasi-nonlinear distortions.

In view of the above, the present paper does not attempt to define precisely the notion of transient signal distortions in an enclosure and their quantitative evaluation and is limited to a comparative analysis of the instantaneous spectrum of a distorted signal caused by the effect of the delays of particular reflected waves with respect to a reference signal in which the effect of reflected waves is absent.

reference is in this case a signal obtained in anechoic conditions, or (in practice) a signal in which the direct wave dominates. Thus, each signal for an unsteady state of sound in the enclosure in which reflected waves occur is a distorted signal. This paper attempts to grasp the character of these distortions and to evaluate their subjective audibility.

The aim of this paper is to analyze the effect of quasi-nonlinear distortions in the three aspects:

– theoretically, for a simplified case which can be described analytically;

– experimentally, in order to determine whether the phenomena described in the theoretical part occur in real conditions;

– subjectively, in order to define the audibility of differences between a distorted signal and an undistorted one for different durations of distortion.

The present investigations are preliminary and are limited to an examination of sinusoidal signals. In the future, these investigations will include real speech and music signals in a larger number of halls, and also other methods of psychoacoustic evaluation.

2. Theoretical analysis

It is known that Sabine's statistical theory is good for description of the acoustic field of enclosures in the case when the condition of a regular distribution of sound energy density in reflected waves, i.e. for a good diffusion of sound energy, is satisfied [1].

In unsteady-state of sound in an interior, i.e. when the sound rises or decays in an interior, this condition leads to an exponential increase or an exponential decrease in the SPL of waves reflected in an arbitrarily chosen observation point. In practice the curves of the rise or decay of sound in an interior deviate greatly from theoretical curves (most so for sinusoidal signals), in particular in the beginning of the rise or decay of sound. This is caused by an insufficient, from the point of view of the laws of statistics, number of the waves of the first reflections and by the effect of phase displacements between particular reflected waves whose sound pressure amplitudes are much higher than the amplitudes of the waves of the later reflections. This effect is one of the basic reasons why sinusoidal signals are not used in measurements of the reverberation time and why the first time section corresponding to a drop in SPL of 5 dB with respect to the steady-state level is eliminated from the decay curve.

This approach is fully justified in terms of measurement technique; it leads, however, to the loss of some data about the properties of a hall in the early rise or decay period. These data could be valuable in the future development of new criteria for evaluating halls.

In an unsteady state of sound in an enclosure the method of image sources [10,15] is useful in analysis of the acoustic field, particularly for the first reflections. This method consists in replacing the surfaces limiting the interior with an array of image sources with such properties and spatial arrangement that the sound distribution of the acoustic field in the interior represented by means of image sources corresponds to the sound distribution of this field in real conditions.

Of the major properties of the above representation used in the present paper, it is necessary to mention the following;

— the surfaces limiting the interior are planes;

— the sound power of the image source of the nth order is equal to the sound power of the real source multiplied by the factor β^n, where β is the coefficient of reflections from the planes limiting the interior;

— all image sources are started at the same time as the real source with the same phase;

— the distance of a given image source from the observation point is equal to the passage path of the wave emitted by the source Z and reaching the observation point X after a given number of reflections;

— in terms of amplitude acoustic signals fall within the limits of linear acoustics, which permits the use of a superposition of signals.

These assumptions make it possible to write the following formula for the instantaneous value of sound pressure in an unsteady state when the signal rises in the interior, from the time when the direct wave reaches the observation point till the time when the nth reflected wave does so,

$$p(t) = 1(t-\tau_0)p_0\sin\omega_0(t-\tau_0)+\ldots+1(t-\tau_n)\varrho_n\sin\omega_0(t-\tau_n), \tag{1}$$

where p_0 and p_n are the sound pressure amplitudes of the direct wave and of the nth wave reflected at the observation point X, respectively. The functions $1(t-\tau_0)$ and $1\ (t-\tau_n)$ are unit step-functions representing, respectively, the delays τ_0 and τ_n with which the direct and the nth reflected waves reach the observation point X with respect to the time $t = 0$ when the source is started:

$$\begin{aligned} 1(t-\tau_0) &= 0 \quad \text{for } t<\tau_0, \\ 1(t-\tau_0) &= 1 \quad \text{for } t\geqslant\tau_0, \\ &\ldots\ldots\ldots\ldots\ldots \\ 1(t-\tau_n) &= 0 \quad \text{for } t<\tau_n, \\ 1(t-\tau_n) &= 1 \quad \text{for } t\geqslant\tau_n. \end{aligned} \tag{2}$$

Considering relation (2), equation (1) can be represented in the form of the following sum

$$p(t) = \sum_{i=0}^{n} 1(t-\tau_i)p_i\sin\omega_0(t-\tau_i), \tag{3}$$

where n is the number of reflected waves under observation.

For successive values of the number n with known values of the sound pressure amplitudes p_i and the delays τ_i, it is possible to plot changes in the resultant signal as a function of time in the process of sound rise in the interior for some arbitrarily chosen observation point X. For arbitrary values of the amplitudes p_i and the delays τ_i it is, however, impossible to present the sum expressed in formula (3) in simple analytical form. This is possible only in a simplified case under the following assumptions:

– the sound level amplitudes of particular reflected waves are the same and equal to the amplitude p_0 of the direct wave,

$$p_i = p_0; \tag{4}$$

– successive reflected waves reach the observation point X at the same time intervals τ, which leads to the conditions

$$\tau_i = i\tau. \tag{5}$$

This case does not occur in real enclosures and can exist only in the conditions of the plane wave field limited by two parallel planes with the reflection coefficient $\beta = 1$, in the Kundt tube, for example. Consideration of the above simplified case aims, however, at investigating the character of sound pressure variations in an unsteady state, while experimental investigations in halls should verify whether the phenomena occurring in the simplified case occur in real conditions.

Assuming, for simplification, as the initial moment of the sound rise in the interior the moment when the direct wave reaches the observation point, i.e. $\tau = 0$, it is possible to write the expression of the instantaneous value of the sound pressure after n reflected waves have reached the observation point in the form

$$p(t) = \sum_{i=1}^{n} 1(t-i\tau)\, p_0 \sin\omega_0(t-i\tau). \tag{6}$$

The expression of the sum of sinusoidal signals with the same amplitude and argument which changes by a constant value as a step-function can be represented in analytical form, according to the general relation (5),

$$p_{n-1}(t) = \sum_{i=0}^{n-1} p_0 \sin(x+iy) = p_0 \sin\frac{ny}{2} \operatorname{cosec}\frac{y}{2} \sin\left[x+(n-1)\frac{y}{2}\right]. \tag{7}$$

Changing the number of the terms in the series in formula (7) to n and substituting

$$y = \omega_0\tau, \tag{8}$$

$$y = \omega_0\tau, \tag{9}$$

it is possible to write expression (6) of the sum of sinusoidal signals delayed in time and shifted in phase after n reflected waves have reached the observation point X in the form

$$p_n(t) = 1(t_n - n\tau)\, p_0 \frac{\sin\left[(n+1)\dfrac{\omega_0 \tau}{2}\right] \sin\left(\omega_0 t - n\dfrac{\omega_0\tau}{2}\right)}{\sin\dfrac{\omega_0\tau}{2}}. \tag{10}$$

After the time t_n in which the nth reflected wave reaches the observation point, i.e. for $t_n \gg n\tau$, the function

$$1(t_n - n\tau) = 1, \tag{11}$$

while the step-like changes in a signal caused by the arrival of successive reflected waves at the observation point are expressed in expression (10) by varying the number n, which can be written in the form

$$n = \operatorname{ent}\left(\frac{t_n}{\tau}\right) = f(St). \tag{12}$$

This notation signifies that as successive reflected waves reach the observation point, the value of the number n is changed in a step-like manner, taking the least value of the integer in the quotient t_n/τ. The above step-like changes in the number n form a nonlinear time function which has been written in the general form as $f(St)$.

In further considerations it is convenient to use the ratio of the delay τ between successive reflections and the period T of the signal,

$$\gamma = \frac{\tau}{T} = \frac{\omega_0 \tau}{2\pi}. \tag{13}$$

Substitution of relations (12) and (13) into expression (10) makes it possible to write

$$p_n(t) = p_0 \frac{\sin[f(St)+1]\pi\gamma}{\sin \pi\gamma} \sin[\omega_0 t - f(St)\pi\gamma]. \tag{14}$$

In the above expression the step time function $f(St)$ occurs twice: once as a change in amplitude and once as change in phase. It follows therefore that in an unsteady state of sound in an enclosure, i.e. in the time range $0 < t < \tau_n$, an effect corresponding to simultaneous amplitude and phase modulation of the signal occurs at the observation point X.

It is known that signal modulation can only occur under the conditions in which nonlinear effects occur. In the present case nonlinearity does not apply to the amplitude relations of the system but to the time relations resulting from delays introduced by the system. Therefore, in order to distinguish them from the commonly used notion of nonlinearity, signal distortions considered here were called quasi-nonlinear distortions at the beginning of the paper. In considering the hall as a delaying system which is nonlinear as a function of time, it should be expected that the quasi-nonlinear distortions introduced should influence the character of the frequency spectrum in the unsteady-state of the signal.

The effect of amplitude and phase modulation resulting from formula (14) depends essentially on the coefficient γ expressed in relation (13), representing the ratio of the delay τ between two successive waves reaching the observation point X and the period T of the signal.

It is possible to distinguish here four characteristic ranges of the coefficient γ, leading to qualitatively different effects.

I. $\gamma \leqslant 0.5$, i.e. $\omega_0 \tau \leqslant \pi$.

In this case the delay τ between successive waves reaching the observation point X is lower than half the period T of the signal, causing distortions in the shape of the curve of the signal to occur in the first phase of the sound rise in an interior, analogous to nonlinear distortions. This shape can be determined easily from formula (14) by calculating successive values of $p_n(t)$ for successive numbers n. The shape of a distorted signal can repeat at regular time intervals, if the condition

$$\omega_0 \tau = \frac{2\pi}{k}, \tag{15}$$

where k is an arbitrary positive number, is satisfied.

For example, when $\gamma = 0.5$, i.e. $\omega_0 \tau = \pi$, from formula (12)

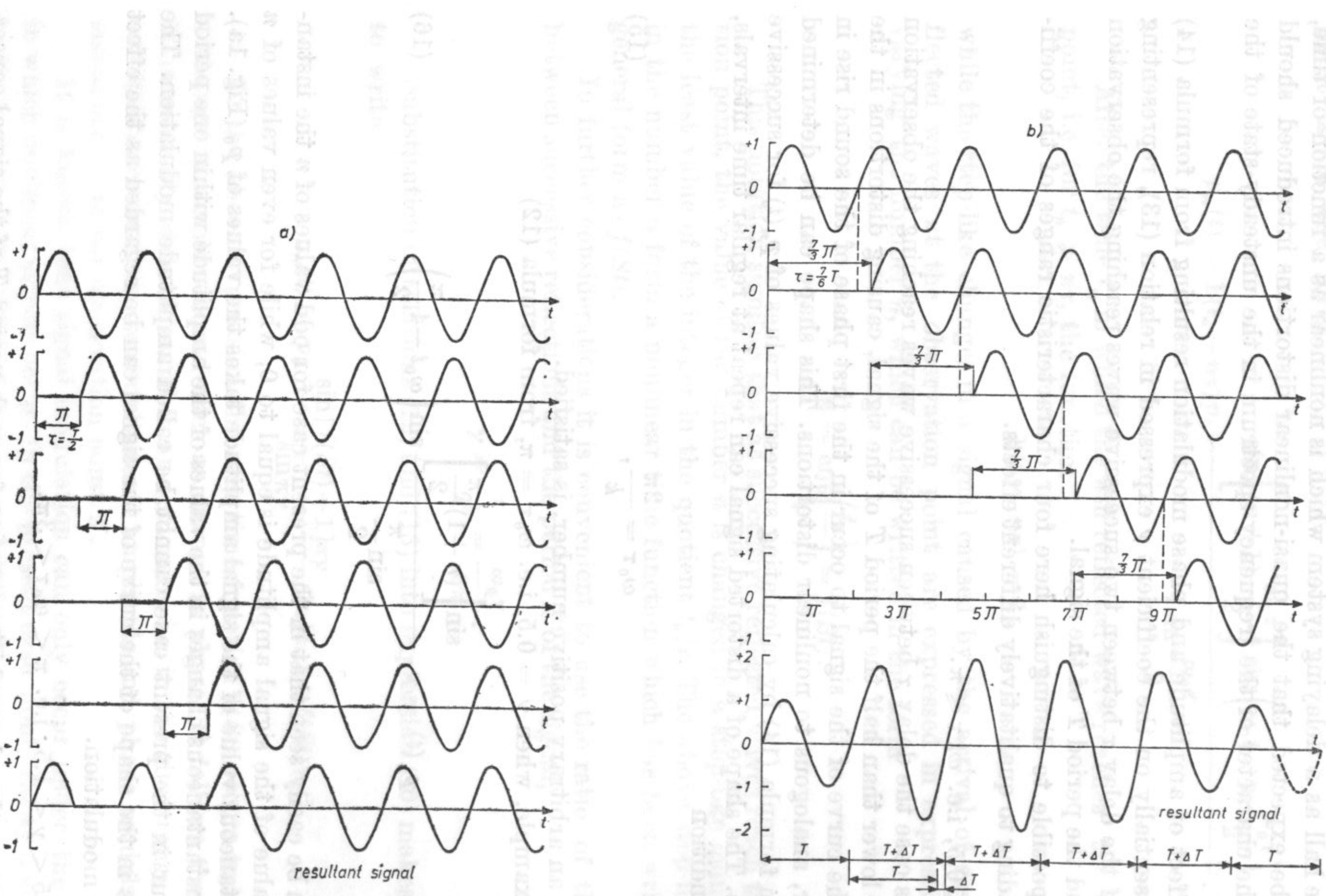
$$p_n(t) = p_0 \frac{\sin\left[(n+1)\dfrac{\pi}{2}\right]}{\sin\dfrac{\pi}{2}} \sin\left(\omega_0 t - 4\frac{\pi}{2}\right). \tag{16}$$

It can be easily seen that in the present case for odd values of n the instantaneous value of the signal amplitude is equal to 0, while for even values of n the instantaneous value of the signal amplitude takes the values of p_0 (Fig. 1a).

The instantaneous changes in the values of the amplitude within one period which occur in the present case cannot be called amplitude modulation. The distortions in the shape of the curve of the signal can be regarded as the effect of phase modulation.

II. $0.5 < \gamma < 1.5$, i.e. $\pi < \omega_0 \tau < 3\pi$.

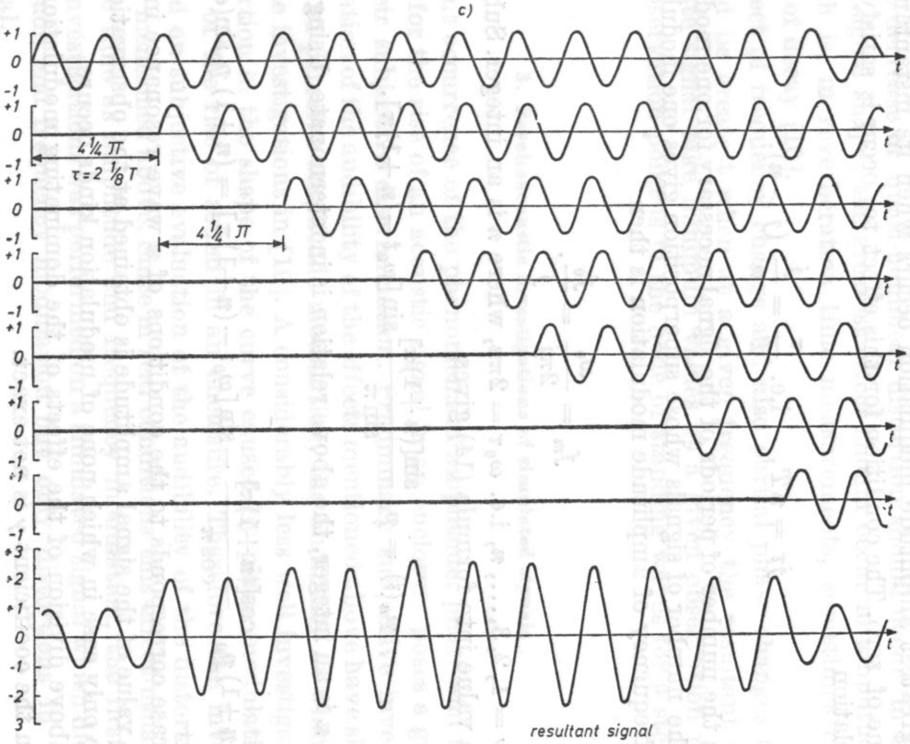
In this case the phase shift occurring for each period T of the signal causes, in addition to change in the signal shape, an effect corresponding to frequency



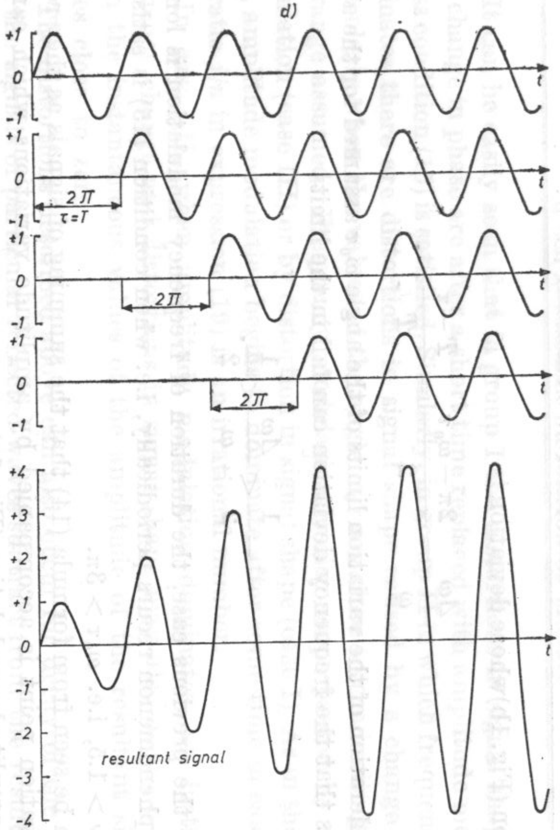
a) distortion in signal shape, $\tau = T/2$ $(\gamma = 1/2)$

b) frequency change $\tau = 7/6\ T$ $(\gamma = 7/6)$

Fig. 1. Characteristic cases of the rise of a sinusoidal signal in enclosure for constant values of the delays of successivesignals by the value τ



c) amplitude modulation $\tau = 2\frac{1}{8}\,T \;\; (\gamma = 4\frac{1}{4})$



d) summing of signals in phase (wave resonance), $\tau = T \;\; (\gamma = 1)$

modulation (Fig. 1b)whose deviation is

$$\frac{\Delta\omega}{\omega} = \frac{2\pi - \omega_0 \tau}{2} = \frac{T-\tau}{T}. \tag{17}$$

Consideration of the variation limits of the angle $\omega_0\tau$ assumed for the above case shows that the frequency deviation can fall in the limits

$$-\frac{1}{2} < \frac{\Delta\omega}{\omega} < \frac{1}{2}. \tag{18}$$

As in the previous case, the duration of frequency modulation is longest when the phenomenon recurs periodically, i.e. when condition (15) is satisfied.

III. $\gamma > 1.5$, i.e. $\omega_0\tau > 3\pi$.

It can be seen from formula (14) that the summing of signals as they reach the observation point is accompanied by amplitude variations which can be treated as amplitude modulation (Fig. 1c).

The greatest amplitude modulation occurs when its instantaneous values ake a value of zero. The condition for this effect to occur is satisfying the following relation

$$j\tau = nT, \quad \text{i.e. } \frac{\tau}{T} = \frac{j}{n} \ (j > n), \tag{19}$$

where j is the number of periods of the signal necessary for one modulation period, n is the number of signals whose superposition gives one modulation cycle.

The frequency of amplitude modulation is then

$$f_m = \frac{\omega_0}{2\pi j} = \frac{f_0}{j}. \tag{20}$$

IV. $\gamma = 1, 2, 3, \ldots, n$, i.e. $\omega_0\tau = 2\pi n$, where n is an integer. Substitution of the above value into formula (14) gives

$$p_n(t) = p_0 \frac{\sin[(n+1)\pi]}{\sin\pi} \sin[\omega_0 t - (n+1)\pi]. \tag{21}$$

Since n is an integer, the above relation is indeterminate. Using de l'Hospital's rule

$$p_n(t) = (n+1)\,p_0 \frac{\cos[(n+1)\pi]}{\cos\pi} - \sin[\omega_0 t - (n+1)\pi] = (n+1)\,p_0 \sin(\omega_0 t). \tag{22}$$

This case corresponds to the conditions of a wave resonance in which the maximum value of the signal amplitude is obtained at the observation point X. This is the only case in which none of modulation kinds occurs.

The above division of the effects of the dominating modulation kinds depending on the coefficient γ is not exact.

It can be easily seen that in group I the distortions in signal shape caused by a change in phase are after a short time replaced with amplitude modulation, unless condition (15) is satisfied. Similarly, in group II in which frequency change dominates there are distortions in signal shape caused by a change in phase, particularly when the values of the angle $\omega_0\tau$ are close to the limiting values of the range assumed. Frequency modulation is also accompanied by amplitude modulation (case III) or by distortion in signal shape (case I). As in the previous case, amplitude modulation begins to dominate after some time in case II when the ratio j/n in expression (19) is an irrational number.

In case III, when amplitude modulation dominates, an instantaneous frequency change and distortions in signal shape may also occur, particularly when the instantaneous values of the amplitude of the resultant signal take values close to zero.

The distortions described above can also occur for decay of the signal but then the signals of particular reflected waves subtract from the resultant signal in a steady state, which decreases the influence of the effects mentioned above. The greatest decrease of quasi-nonlinear distortion should occur in the effect of distorted signal shape; the greatest effect should occur in amplitude modulation, which is, in reverberation time measurements, sometimes called the irregularity of decay [10].

The effect of frequency change and also partial phase changes for sound decay should be greatest when at a given frequency the resultant values of a steady-state signal take minimum values at a given observation point, i.e. when the observation point is in the node of standing waves for a steady-state condition.

3. Psychoacoustic investigations of simulated signals

The joint occurrence of the phenomena of amplitude, phase and frequency modulation for the rise of an acoustic signal in an enclosure poses a great difficulty in their subjective discrimination. Preliminary subjective investigations of the evaluation of the audibility of the effects mentioned above have shown that it is easiest to perceive amplitude modulation, the audibility of which was the object of the investigations in [19]. A considerably less well investigated effect is the distortion in the shape of the curve caused by phase modulation in the first period of the rise of sound in an enclosure. Therefore the investigation concentrated on subjective evaluation of the audibility of the distortion in the curve shape mentioned above. Since in real conditions this effect usually occurs at the same time as amplitude and frequency modulation, the first stage of subjective investigations was performed on a simulated signal which did not contain amplitude and frequency modulation but only distortions in the shape of the signal.

The simulated signal was a sinusoid with a continuously controlled number of half-sinusoids cut-off by the digital system (Fig. 2) [17].

When n denotes the number of distorted periods, the signal under investigation is described by relation (16) for numbers less than n; or it is an undistorted sinusoidal signal for numbers greater than n. The investigations were performed on signals for the numbers n equal to 1, 2, 4, 8, 16, which when recorded on magnetic tape alternating with undistorted signals in a random order were reproduced through headphones. The investigations were performed for frequencies of 250 Hz, 800 Hz and 2.5 kHz. Several measurement series, 10 signals each, were performed for each of the above frequencies.

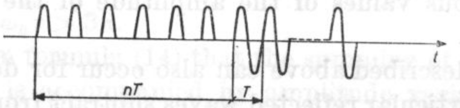


Fig. 2. An example of distorted simulated signal for subjective investigations of the evaluation of the audibility of changes in the signal shape (n is the number of periods of the distorted signal)

15 listeners participated in the investigations. Their task was to evaluate which of successive signals was regarded as distorted one. The audibility S, in %, was achieved from the ratio of the number of correct answers to the total number of signals offered for each frequency. The results, after ordering with respect to the number n of distorted periods, are shown in Fig. 3a.

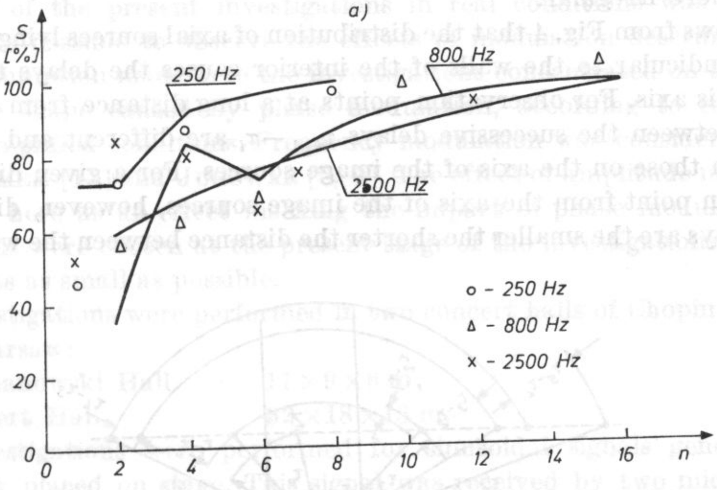
Before the investigations the listeners underwent preliminary training in order to become generally accustomed to the kind of signals emitted.

It follows from Fig. 3a that the best audibility of distortions was obtained for low frequencies. For medium and, particularly, high frequencies high values of the audibility S occurred as the number of distorted periods increased. This results from the fact that the period of a signal decreases proportionally for higher frequencies, which signifies a decrease in the duration of distortions.

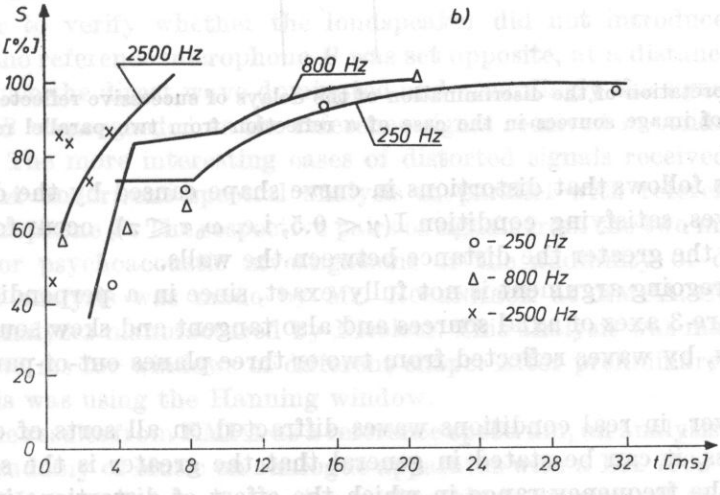
In order to verify in what way the audibility of distortions depends on their duration at different frequencies, the results in Fig. 3a were recalculated as a function of time and shown in Fig. 3b. These results show that the audibility of distortions involves different times necessary to recognize distortions of different frequency. This result differs from those of other psychoacoustic investigations, for example, in the case of perceiving echo, where the time needed for perceiving the phenomenon does not depend on frequency. It is interesting to note that recognition of distortions at a frequency of 2.5 kHz takes their duration of the order of 1-2 ms.

The method of subjective evaluation used in the present paper can provoke a number of objections, both to the principle of evaluation and the criteria of perceptibility used. An additional method which consisted in offering the listener pairs of signals was therefore used in the investigations. These pairs consisted either of a distorted signal and an undistorted one of different duration of distortions or of two undistorted signals, or of two signals distorted in the same way. The listeners' task was to evaluate which of the pairs of signals they regard as different.

The results obtained are shown in Fig. 3a, b in the form of ×, △ and ○. They show a character close to that of the results obtained using the previous method, which to a certain extent increases the reliability of evaluations made.



a) as a function of the number n of distorted periods



b) as a function of the duration of distortion

Fig. 3. The evaluation, in %, of the audibility S of distortions in a case of the simulated signal

4. Distortions in the shape of transient signals in real conditions

Condition (5) accepted in section 2 and regarding the same values of the delays τ of the signal and condition (4) assuming the equality of amplitudes are not satisfied in real conditions.

It is practically impossible to describe mathematically the distortions which occur then and it is only possible to make graphic analysis. It follows from this analysis that the effects of phase, frequency and amplitude modulation described for a simplified case also occur in real conditions although ther character is more irregular.

It follows from Fig. 4 that the distribution of axial sources lying on a straight line perpendicular to the walls of the interior causes the delays to be regular only on this axis. For observation points at a long distance from this axis differences between the successive delays $\tau_{i+1}-\tau_i$ are different and considerably lower than those on the axis of the image sources. For a given distance of the observation point from the axis of the image sources, however, differences between delays are the smaller the shorter the distance between the walls.

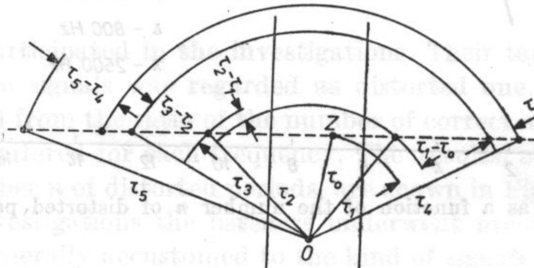


Fig. 4. Interpretation of the discrimination of the delays of successive reflected waves using the method of image sources in the case of a reflection from two parallel reflecting walls

It thus follows that distortions in curve shape caused by the delays of reflected waves, satisfying condition I($\gamma < 0.5$, i.e. $\omega_0\tau \leqslant \pi$), occur for the lower frequencies the greater the distance between the walls.

The foregoing argument is not fully exact, since in a perpendicular-walled hall there are 3 axes of axial sources and also tangent and skew sources caused, respectively, by waves reflected from two or three planes out-of-parallel to one another.

Moreover, in real conditions waves diffracted on all sorts of edges occur. Nevertheless, it can be stated in general that the greater is the size of halls, the lower the frequency range in which the effect of distortions in the shape of the curve of the signal may occur when sound rises in the interior.

It follows from the analysis of the distribution of image sources that the mean delay between successive signals can increase as the number of real sources increases or when soud-diffusing elements are introduced, which should cause an increase in the frequency at which distortions in the shape of the curve of the signal appear in a given hall. The above problem is the object of further investigations.

5. Experimental results

The aim of the present investigations in real conditions was to verify whether it was possible to observe the effects of modulation described above. Under the assumption made here the investigations concentrated on the distortion in curve shape caused by phase modulation, according to case I considered for simplified conditions. Frequency modulation was considered in the papers of Ozimek [14] and Jugowar [8, 9]. The effect of amplitude modulation was regarded here as an effect masking the impact of phase modulation and such conditions were chosen at the present stage of the investigations, in which this effect was as small as possible.

The investigations were performed in two concert halls of Chopin Academy of Music, Warsaw:

I. Szymanowski Hall	17 ×9 ×6 m,
II. Concert Hall	32 ×18 ×13 m.

The investigations were performed for sinusoidal signals generated by a loudspeaker placed on stage. This signal was received by two microphones. The measurement microphone *A* was placed in the hall at a distance of 10 m from the loudspeaker.

In order to verify whether the loudspeaker did not introduce transient distortions, the reference microphone *B* was set opposite, at a distance of 0.5 m. At this distance the direct wave dominated and accordingly the signal from the microphone *B* was regarded as the reference signal received in conditions close to anechoic. The more interesting cases of distorted signals received from the microphone *A* underwent spectral analysis in parallel with reference signals from the microphone *B*. The respective pairs of signals from the two microphones were used for psychoacoustic investigations of the audibility of distortions.

Spectral analysis was made by Mr. Rosenheck at EMPA, Switzerland, on an FFT analyzer manufactured by Nicolett. This analysis was made at time intervals of 40 ms for windows of different shape. After preliminary investigations analysis was using the Hanning window.

Using the results from EMPA as a reference spectrum, an analysis was made at Chopin Academy of Music on analogue apparatus with a BK 7502 digital register and a BK 2109 frequency analyzer. The block diagram of the measurement system for the signal *A* received from a long distance from the loudspeaker and for the signal *B* regarded as the reference signal is shown in Fig. 5 [18].

The changes in the signals A and B as a function of time obtained in Szymanowski Hall at a frequency of 180 Hz are shown in Fig. 6 for a time range of 160 ms calculated from the time of the arrival of the direct wave at the observation point. It follows from these curves that the duration of the distorted signal is relatively large, i.e. about 80 ms.

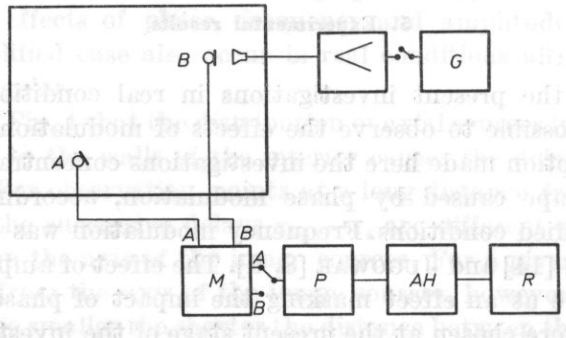


Fig. 5. A schematic diagram of the measurement system in the concert halls

A — the measuring microphone, B — the reference microphone, G — the generator of sinusoidal signals, M — a Nagra IV *SJ* tape recorder, P — digital memory, BK 7502, AH — heterodyne analyser, BK 2109, R — register, BK 2305

The results of the measurements of the frequency spectrum in the time ranges 0 - 40 ms, 40 - 80 ms and 80 - 120 ms for the signals A and B taken at EMPA are shown in Fig.7. The changes in the frequency spectrum of the same signals measured in the same ranges but obtained in the Academy of Music are shown in Fig. 8.

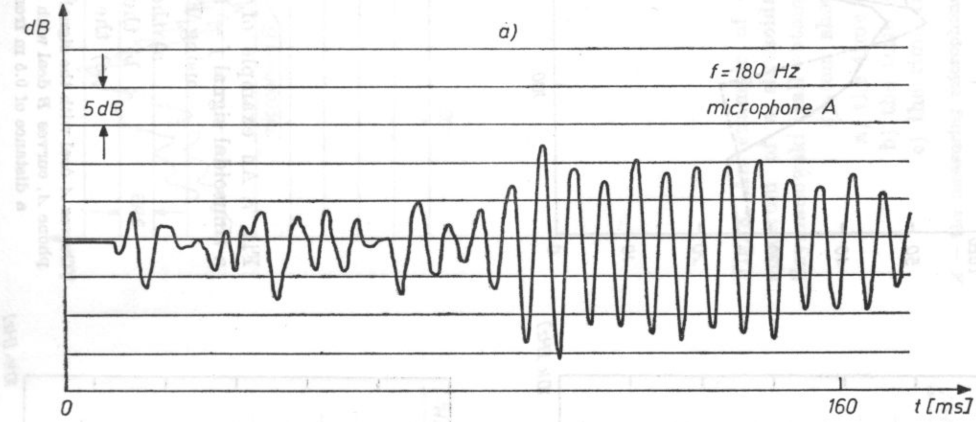
Comparison of changes in the frequency spectrum obtained using the two methods shows some differences which indicate the effect of measuring apparatus on the results. However, the general character of differences between the curves obtained from the measurement microphone A and the reference microphone B is similar. The differences between the spectra indicate a distinctly fuzzy frequency spectrum and the occurrence of an additional frequency in the time range 0-40 ms in the signal A, which are absent from the signal B. In the next time range 40-80 m the differences are smaller and disappear in the time range 80-120 ms.

This result indicates a distinct change in the frequency spectrum caused by quasi-nonlinear distortions in the first stage of the rise of the signal, which agrees with the behaviour of the signal observed in the time domain (Fig. 6).

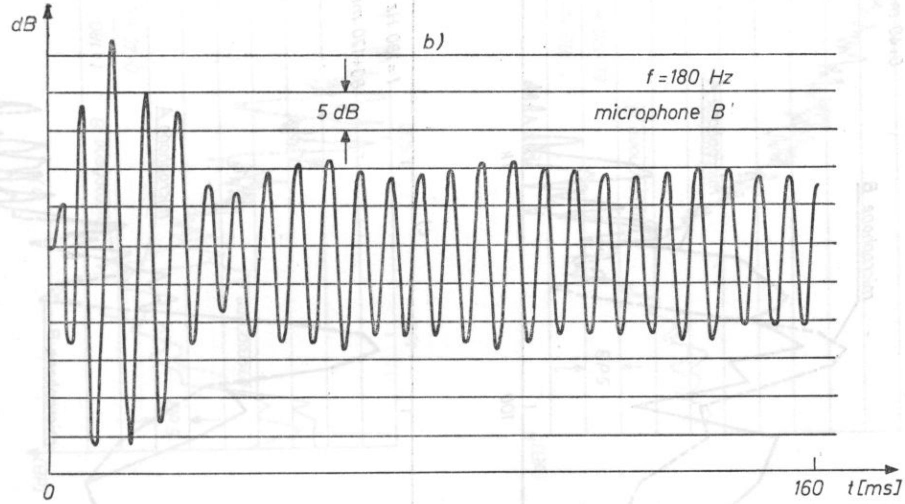
Similar investigations were performed in the Concert Hall. In view of the larger size of the hall it was easier to obtain distortions in curve shape under consideration here for low frequencies of the order of 100-130 Hz. This frequency range has little practical significance, therefore the effect of distortion in curve shape was examined for slightly higher frequencies. In agreement with predictions resulting from theoretical considerations, amplitude modulation dominated increasingly as the frequency of the signal increased. Nevertheless the effect of distortion in curve shape was very distinct for selected microphone positions in the frequency range up to 350 Hz.

Examples of signals obtained from the microphones *A* and *B* for a frequency of 340 Hz are shown in Fig. 9; their frequency spectra obtained in the Academy of Music, Warsaw, in Fig. 10.

As in the case of the measurements in Szymanowski Hall, the duration of the distorted signal was about 80 ms. The changes in the frequency spectrum measured in 40 ms time ranges show, as previously, greatest differences between the signals *A* and *B* in the first stage of sound rise in the interior.



a) the measuring microphone *A*



b) the reference microphone *B*

Fig. 6. An example of the rise of a sinusoidal signal of the frequency $f = 180$ Hz in Szymanowski Hall in Chopin Academy of Music, Warsaw

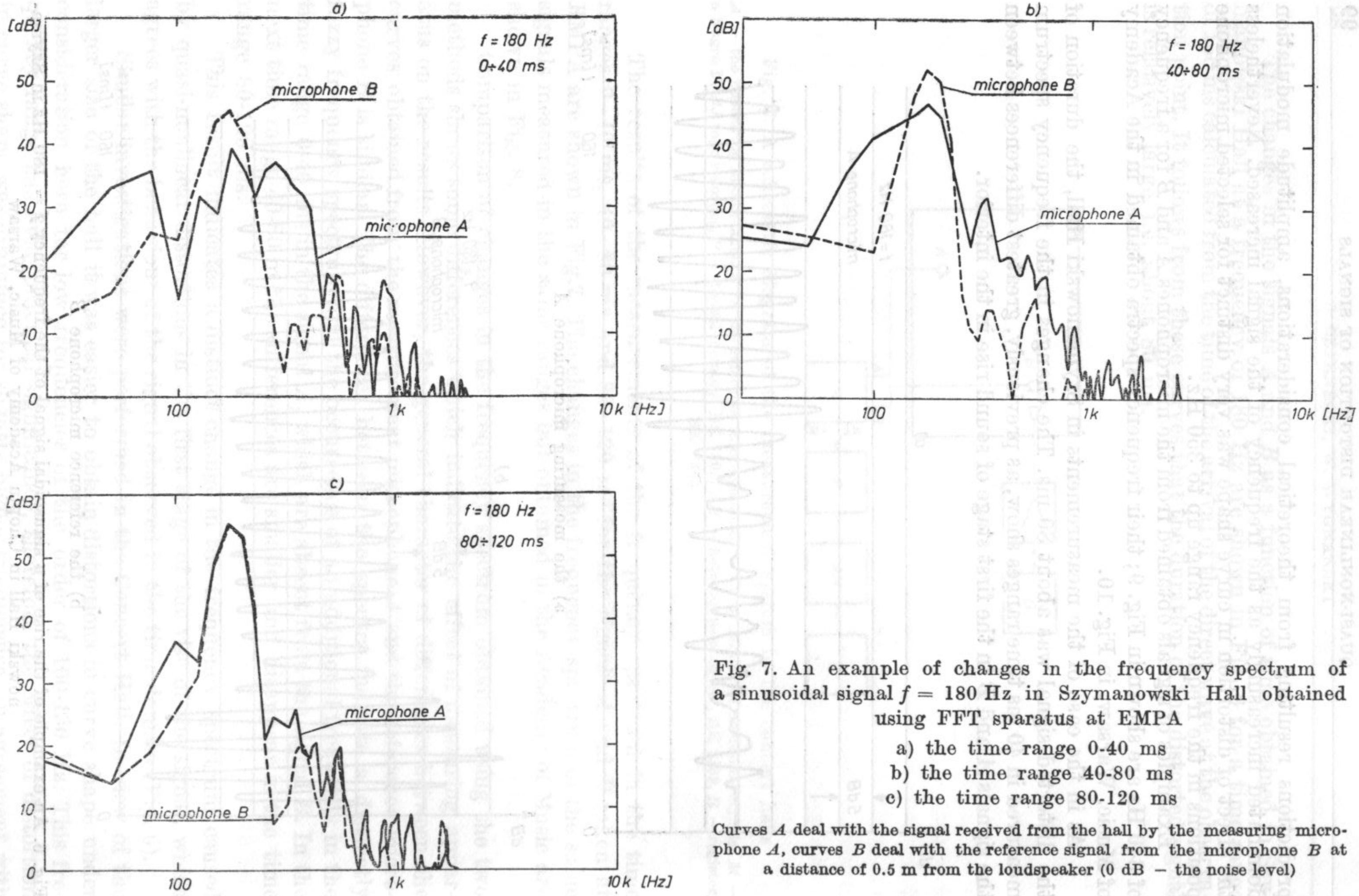


Fig. 7. An example of changes in the frequency spectrum of a sinusoidal signal $f = 180$ Hz in Szymanowski Hall obtained using FFT sparatus at EMPA

a) the time range 0-40 ms
b) the time range 40-80 ms
c) the time range 80-120 ms

Curves A deal with the signal received from the hall by the measuring microphone A, curves B deal with the reference signal from the microphone B at a distance of 0.5 m from the loudspeaker (0 dB – the noise level)

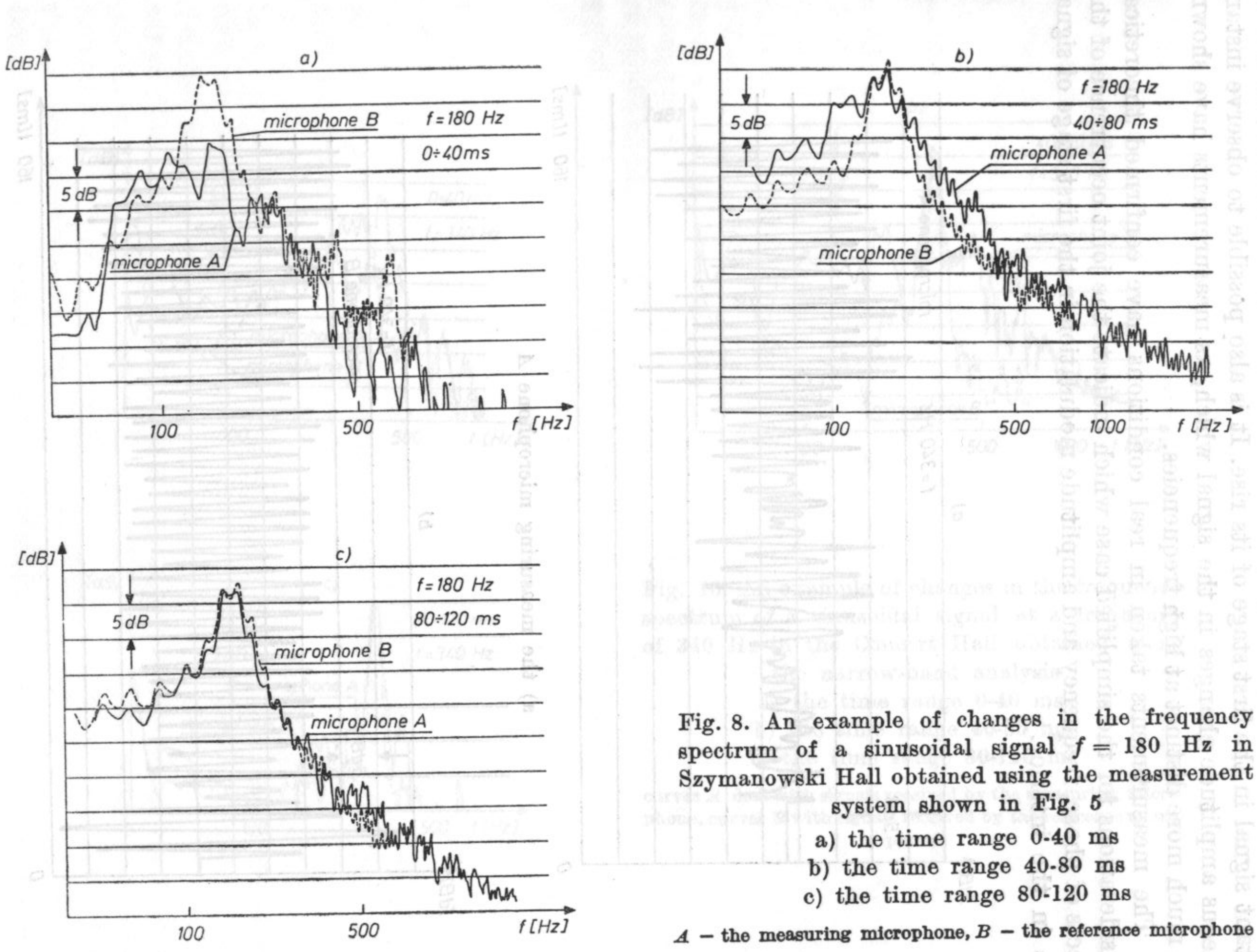


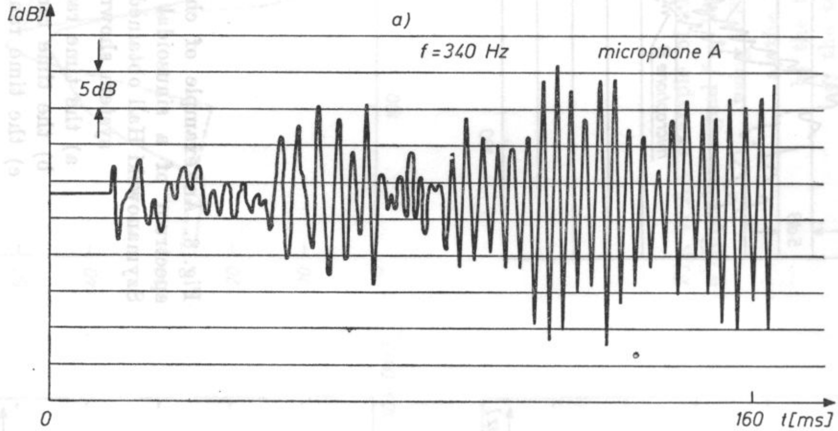
Fig. 8. An example of changes in the frequency spectrum of a sinusoidal signal $f = 180$ Hz in Szymanowski Hall obtained using the measurement system shown in Fig. 5

a) the time range 0-40 ms
b) the time range 40-80 ms
c) the time range 80-120 ms

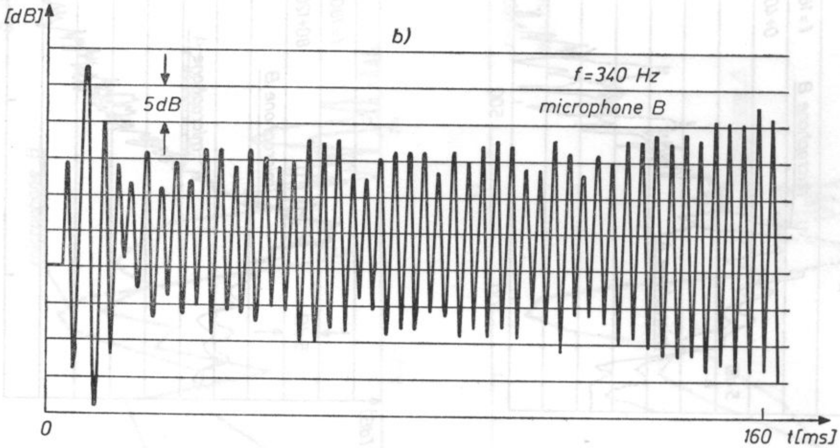
A – the measuring microphone, *B* – the reference microphone

Observations of the signals A in the time domain (Figs. 7 and 9) have shown an irregularity in the time ranges of zero crossings of the signal, indicating instantaneous frequency changes, similar to frequency modulations, in the resultant signal in the first stage of its rise. It is also possible to observe instantaneous amplitude changes in the signal which, as measurements have shown, are much more distinct at high frequencies.

The measurements taken in real conditions have confirmed theoretical considerations for the simplified case which indicate the joint occurrence of the effects of phase, frequency and amplitude modulation in the first stage of signal rise in the interior.



a) the measuring microphone A



b) the reference microphone B

Fig. 9. An example of the rise of a sinusoidal signal at a frequency of 340 Mz in the Concert Hall of Chopin Academy of Music, Warsaw

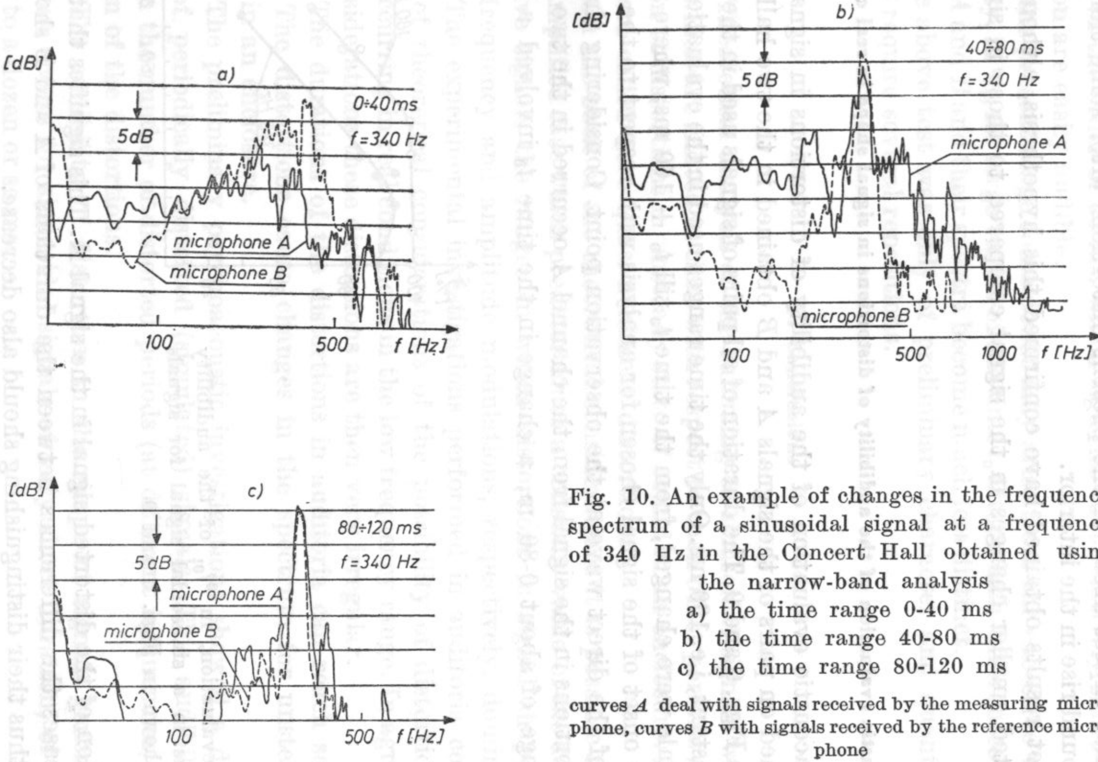


Fig. 10. An example of changes in the frequency spectrum of a sinusoidal signal at a frequency of 340 Hz in the Concert Hall obtained using the narrow-band analysis

a) the time range 0-40 ms
b) the time range 40-80 ms
c) the time range 80-120 ms

curves *A* deal with signals received by the measuring microphone, curves *B* with signals received by the reference microphone

The qualitative evaluation of the phenomena in real conditions performed in section 4 showed that the distortions are irregular and disappear after some time.

In practice, auditoria involve more complex signals, such as signals of speech and music. Naturally their character changes in time and accordingly one should not expect decay of the present distortions after some time in the process of sound rise in the interior.

The pilot results obtained have confirmed this hypothesis, although they have indicated smaller changes in the signal compared to those in sinusoidal signals.

6. Psychoacoustic evaluation of the audibility of distortions in signal shape in real conditions

Psychoacoustic evaluation of the audibility of distortions in signal shape was performed on pairs of the signals *A* and *B* obtained in the two halls which are shown in Figs. 6 and 9. The duration of all pairs of signals used in the evaluation was constant, i.e. 160 ms. Only the time ranges used in the evaluation of the pairs of signals were changed, from the time t_0 till $t_0 + 160$ ms, where t_0 is the delay of the onset of the signal chosen for analysis with respect to the time of the arrival of the direct wave at the observation point. Considering that most distinct distortions in the signal from the channel *A* occurred in the two halls in the time range of about 0-80 m, a change in the time t_0 involved a change in the duration of the distorted signal in the signal *A*. This signifies that as the time t_0 increases the differences between the durations of *A* and *B* should decrease; and thus their distinguishing should also decrease.

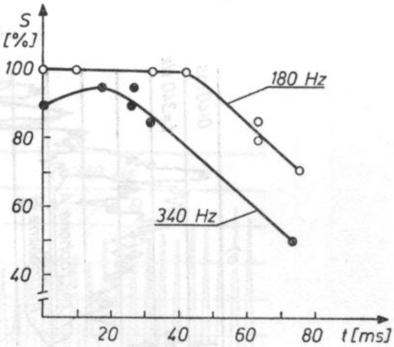


Fig. 11. The evaluation, in $^0/_0$, of the audibility *S* of distortions in a sinusoidal signal for signals shown in Figs. 6 and 9

In the present psychoacoustic investigations the gaps between the signals within one pair compared were 1 s, while the pauses between pairs were about 5 s. The test consisted of 10 pairs for each hall and was estimated by 20 listeners, The order of selection of each pair was random.

The results of the evaluation are shown in Fig. 11. They indicate a very high audibility close to 100 % of the differences between particular pairs of signals when the time $t_0 = 0$, i.e. in the time ranges chosen for evaluation the whole distorted fragments were within the signal A. As the time t_0 increased, i.e. as the distorted fragment became shorter, the distinguishing of signals fell but only after the time t_0 of the order of 40 ms, reaching a value of 70% for the times t_0 of the order of 60 ms, which signifies that relatively short durations of distortion are easily audible.

At the present stage some doubt may arise from the excessive values of the audibility, in %, for large values of the time t_0 after which differences between the signals A and B and their spectra become much less distinct.

The above test was only of preliminary character and quantitative conclusions require several repetitions.

7. Conclusions

1. Theoretical analysis of the summing of sinusoidal signals with different phases which are shifted in time with respect to each other permits simple interpretation of distortions in unsteady state of sound in an enclosure. These distortions were called here quasi-nonlinear distortions.

2. On the basis of the theoretical considerations it is possible to determine in the simplified case which predetermines a constant value of the delay between successive signals the ranges in which distortions in signal shape caused by phase, frequency and amplitude modulations, respectively, dominate.

3. The experimental investigations performed in auditoria confirmed the results of theoretical considerations of the possibility of distortions in curve shape occurring in real conditions in the low frequency range. In agreement with the considerations these distortions are then very irregular.

4. The durations of the distortions in auditoria can reach several scores of ms. The distortions cause changes in the spectrum for unsteady state of signals in an enclosure.

5. The preliminary psychoacoustic investigations showed a distinct audibility of periodically distorted (simulated) signals. This audibility depends more on the number of distorted periods (at different frequencies) than on the duration of the distortions.

6. The audibility of distortion in the shape of signals generated in real conditions occurs for sinusoidal signals of very short duration, of the order of several to a dozen or so ms.

7. At the present stage it is difficult to evaluate the influence of distortions considered on the evaluation of the signals of speech and music quality in enclosures. Therefore the further investigations will follow in the three directions:

– analysis of factors increasing the effect of the distortions considered in this paper,

— investigation of the effect of the interior and its parameters on the distortions,

— psychoacoustic investigations of the effect of the distortions on the audibility of changes in the sound of speech and music signals.

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