Calculation of subjective preference at each seat in a concert hall[a)]

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This paper represents a method of calculating the subjective preference of sound fields in concert halls before construction. Subjective preference judgments (paired comparison tests) were systematically performed using fully independent objective parameters of acoustic information which describe the signals to the two ears. The sound fields with various combinations of listening level, delay of early multiple reflections, subsequent reverberation time, and magnitude of the interaural cross correlation were simulated with the aid of a digital computer. The optimal conditions maximizing the subjective preference could be found for each objective parameter, because the parameters had an almost independent effect on the subjective preference judgments. Based on the linear scale value, which is obtained by applying the law of comparative judgment, we can calculate a total preference value according to the "principle of superposition." Examples of calculating the preference values by use of the plan and the cross section of a concert hall are described.

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INTRODUCTION

The facts of what constitutes a superior sound field in concert halls has only gradually come to light. A series of subjective preference tests for simulated sound fields has revealed that all of the significant objective parameters used to describe the sound signals at both ears can be reduced to the following four factors[1–7]: (1) level of listening; (2) delay time of early reflections; (3) subsequent reverberation time; and (4) magnitude of interaural crosscorrelation. Factors (1) through (3) are called "temporal–monaural criteria" because they may be closely associated with source signals perceived by only one of two ears. Factor (4) is characterized by head and both pinnae, and is almost independent of the source signal. Thus it is called a "spatial–binaural criterion" representing the binaural interdependence.

For the purpose of realizing the optimal design of concert halls, we shall show that the conditions maximizing the subjective preference can be found for each objective parameter, because the parameters influence preference independently of each other. Finally, a method is proposed of calculating the linear value of subjective preference for concert halls at the design stage.

I. SIGNALS AT BOTH EARS

In order to simplify the analysis, we first consider a single sound source on the stage. Let $h_L(r|r_0;t)$ and $h_R(r|r_0;t)$ be pressure impulse responses between the sound source located at r_0 and the left and the right ear-canal entrances, respectively, of a listener sitting at r (located at the center of the head). Then, the pressures at both ears, which must include all acoustic information to be analyzed, are expressed by the following equations:

$$f_L(t) = \int_0^t p(v)h_L(t-v)dv = p(t)*h_L(t),$$
$$f_R(t) = \int_0^t p(v)h_R(t-v)dv = p(t)*h_R(t), \tag{1}$$

where $p(t)$ is a source signal and the asterisk denotes convolution. The impulse responses may be decomposed into a set of the impulse responses $w_n(t)$ describing the reflection property of boundaries and the impulse responses from the free field to the ear-canal entrance $h_{nL}(t)$ or $h_{nR}(t)$, n denoting a single sound with a horizontal angle ξ and an elevation angle η to the listener ($\xi = 0$ and $\eta = 0$ signify the frontal direction). The impulse response may now be written in the form

$$h_{L,R}(r|r_0;t) = \sum_{n=0}^{\infty} A_n w_n(t-\Delta t_n)*h_{nL,R}(t), \tag{2}$$

where A_n and Δt_n are the pressure amplitude and the delay time of the reflections relative to the direct sound, respectively. The amplitude A_n is determined by the "$(1/r)$-law," A_0 being unity. Every n (>1) corresponds to a single reflection; $n = 0$ refers to the direct sound. Equation (1) becomes

$$f_{L,R}(t) = \sum_{n=0}^{\infty} p(t)*A_n w_n(t-\Delta t_n)*h_{nL,R}(t). \tag{3}$$

If the sound source has nonuniform radiation, the radiation pattern of sound source is taken into consideration for each sound direction, i.e., $p(t)$ is replaced by $p_n(t)$ in the above equation. If there are many sound sources distributed on the stage, then the pressures at the two ears may be expressed as a linear sum of $f_{L,R}(t)$ given by Eq. (3), for usual sound pressure levels. It is worth noticing that the preferred condition for the distributed sources may be found by aggregating the

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conditions for the individual and important sources on the stage.

All independent objective parameters of acoustic information, which are included in the sound pressures at the two ears given by Eq. (3), may be reduced to the following as before.[1]

(i) The first parameter is the source signal $p(t)$, which can be represented by its long-time autocorrelation function

$$\Phi_p(\tau) = \lim_{T > \infty} \frac{1}{2T} \int_{-T}^{+T} p'(t) p'(t+\tau) dt,$$

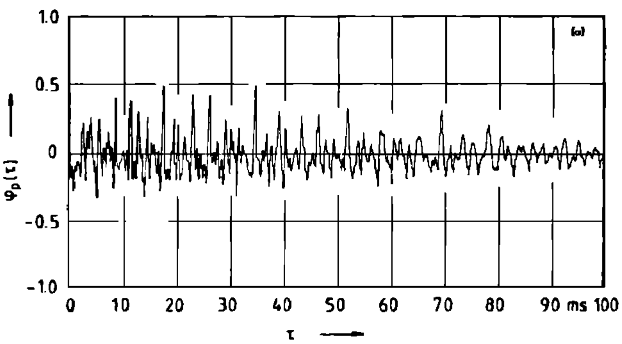
where $p'(t) = p(t) * s(t)$, $s(t)$ corresponds to the ear sensitivity, which might be characterized by the external ear and the middle ear. For practical convenience, $s(t)$ is chosen as the impulse response of an A-weighting filter. The above equation may be divided into the intensity of the sound signal $\Phi_p(0)$ and the normalized autocorrelation function, which is defined by

$$\phi_p(\tau) = \Phi_p(\tau)/\Phi_p(0). \quad (4)$$

Measured autocorrelation functions are shown in Fig. 1 for the first two music motifs listed in Table I (*temporal–monaural criterion*).

(ii) The second objective parameter is the set of impulse responses of the reflecting boundaries, $A_n w_n(t - \Delta t_n)$, which represents the initial time delay gap between the direct sound and the first reflection, as well as the early multiple reflections, the subsequent reverberation, and any spectral changes due to the reflections (*temporal–monaural criterion*).

(iii) The two sets of the head related impulse responses, $h_{nL,R}(t)$, constitute the remaining objective parameter.



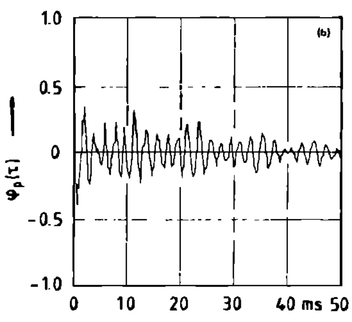


FIG. 1. Measured autocorrelation functions of source signals. (a) Music motif A: Royal Pavane by Gibbons, $\tau_e = 127$ ms. (b) Music motif B: Sinfonietta by Arnold, $\tau_e = 43$ ms.

TABLE I. Music and speech used and the effective duration of the long-time autocorrelation function.

Sound source	Title	Composer	τ_e (ms)[a]
Music A	Royal Pavane	Gibbons	127 (127)
Music B	Sinfonietta, Opus 48; IV movement Allegro con brio	Malcolm Arnold	43 (35)
Music C	Symphony No. 102 in B flat major; II movement; Adagio	Haydn	(65)
Music D	Siegfried Idyll; Bar 332	Wagner	(40)
Music E	Symphony in C major, K-V no. 551, Jupiter IV movement; Molto Allegro	Mozart	38
Speech S	Reading poem spoken by a female	(D. Kunikita)	10 (12)[b]

[a] The effective duration of the autocorrelation function is defined by the delay τ_e at which the envelope of the normalized autocorrelation function becomes just 0.1. The delay τ_e may differ slightly with different radiation characteristics of the loudspeaker being used. The values in the parentheses were previously reported.[1]

[b] The effective durations for different languages are considered to be similar, because the fundamental frequencies do not much differ among the languages.

These responses $h_{nL}(t)$ and $h_{nR}(t)$ play an important role in localization, however, and are not mutually independent objective factors. For example, $h_{nL}(t) \approx h_{nR}(t)$ in the median plane ($\xi = 0°$).

Therefore, in order to represent the interdependence between these impulse responses, one may introduce a single factor, i.e., the long-time interaural crosscorrelation between the continuous sound signals $f_L(t)$ and $f_R(t)$. This becomes a significant factor in determining the degree of subjective diffuseness of sound fields.[8] Subjective diffuseness, or no special directional impression, is the percept for sound fields with a low degree of interaural crosscorrelation. On the other hand, a well-defined direction is indicated, if the interaural crosscorrelation has a strong peak for $|\tau| < 1$ ms. For example, if the peak is observed at $\tau = 0$, then a frontal source direction can be perceived. The interaural crosscorrelation depends mainly on the direction of arrival of reflections at the listener and on their amplitudes.

The interaural crosscorrelation is defined by

$$\Phi_{LR}(\tau) = \lim_{T \to \infty} \frac{1}{2T} \int_{-T}^{+T} f_L(t) f_R(t+\tau) dt,$$

$$|\tau| \geqslant 1 \text{ ms}. \quad (5)$$

First, let us consider the interaural crosscorrelation $\Phi_{LR}^{(0)}(\tau)$ of the direct sound only. The pressures at the two ears are, then, expressed by

$$f_L(t) = p(t) * h_{0L}(t),$$
$$f_R(t) = p(t) * h_{0R}(t).$$

If the listener is facing the sound source, the normalized in-

TABLE II. Measured correlation functions ($\tau = 0$) at both ears for each single sound arriving, as a function of horizontal angle of incidence ξ ($\eta = 0°$). For the range $180° < \xi < 360°$, the value may be obtained by putting $\xi = 360° - \xi$ and interchanging the suffixes L and R.

Sound source		The horizontal angle ξ[a] 0°	18°	36°	54°	72°	90°	108°	126°	144°	162°	180°
Music A	$\Phi_{LR}(0)$	0.99	0.30	−0.32	−0.32	0.09	0.13	0.00	−0.07	−0.09	0.30	0.69
	$\Phi_{LL}(0)$	1.00	0.71	0.42	0.32	0.34	0.65	0.62	0.19	0.24	0.52	0.69
	$\Phi_{RR}(0)$	1.00	1.12	1.31	1.42	1.27	1.51	1.51	0.75	0.84	0.75	0.71
Music B	$\Phi_{LR}(0)$	0.99	−0.17	0.18	−0.28	−0.04	0.04	0.00	−0.06	0.03	0.00	0.63
	$\Phi_{LL}(0)$	1.00	0.54	0.39	0.35	0.28	0.34	0.30	0.23	0.27	0.39	0.66
	$\Phi_{RR}(0)$	1.00	1.38	1.73	2.06	1.42	1.25	1.13	0.87	0.92	0.75	0.66
Music C	$\Phi_{LR}(0)$	0.97	0.18	−0.06	−0.28	−0.16	−0.14	−0.16	−0.12	−0.04	0.22	0.61
	$\Phi_{LL}(0)$	1.00	0.66	0.42	0.40	0.42	0.60	0.57	0.26	0.30	0.45	0.72
	$\Phi_{RR}(0)$	1.00	1.17	1.34	1.50	1.39	1.39	1.30	0.87	0.85	0.74	0.58
Music D	$\Phi_{LR}(0)$	1.00	−0.11	0.02	−0.23	0.06	0.15	0.10	−0.08	−0.06	−0.06	0.88
	$\Phi_{LL}(0)$	1.00	0.65	0.46	0.20	0.28	0.36	0.28	0.22	0.23	0.44	0.90
	$\Phi_{RR}(0)$	1.00	1.40	1.66	1.66	1.56	1.38	1.14	0.97	0.87	0.80	0.96
Music E	$\Phi_{LR}(0)$	0.98	0.03	−0.19	−0.38	−0.42	−0.34	−0.37	−0.46	−0.26	0.18	0.90
	$\Phi_{LL}(0)$	1.00	0.73	0.31	0.39	0.44	0.60	0.47	0.35	0.28	0.63	0.87
	$\Phi_{RR}(0)$	1.00	1.31	1.55	2.61	2.51	2.44	2.30	1.90	1.28	1.13	0.97
Speech S	$\Phi_{LR}(0)$	1.00	0.59	0.12	−0.15	−0.15	−0.18	−0.12	−0.09	0.06	0.44	0.76
	$\Phi_{LL}(0)$	1.00	0.82	0.68	0.65	0.76	0.71	0.76	0.56	0.56	0.68	0.81
	$\Phi_{RR}(0)$	1.00	1.32	1.68	1.85	1.76	1.71	1.76	1.79	1.35	1.06	0.81

[a] For more general angles (ξ, η), the angle given by $\sin^{-1}(\sin \xi \cos \eta)$ is substituted for the horizontal angle ξ.

teraural crosscorrelation defined by

$$\Phi^{(0)}_{LR}(\tau) = \phi^{(0)}_{LR}(\tau)/[\Phi^{(0)}_{LL}(0)\,\Phi^{(0)}_{RR}(0)]^{1/2} \qquad (6)$$

approaches unity because of $h_{0L}(t) \approx h_{0R}(t)$, where $\Phi^{(0)}_{LL}(0)$ and $\Phi^{(0)}_{RR}(0)$ are autocorrelation functions at $\tau = 0$ for each ear.

If discrete reflections are added to the direct sound after the autocorrelation function of the direct sound becomes weak enough, the normalized interaural crosscorrelation is expressed by

$$\phi^{(N)}_{LR}(\tau) = \sum_{n=0}^{N} A_n^2\, \Phi^{(n)}_{LR}(\tau) \times \left(\sum_{n=0}^{N} A_n^2\, \Phi^{(n)}_{LL}(0) \sum_{n=0}^{N} A_n^2\, \Phi^{(n)}_{RR}(0) \right)^{-1/2}$$

$$\text{when } w_n(t) = \delta(t), \qquad (7)$$

where $\Phi^{(n)}_{LR}(\tau)$ is the interaural crosscorrelation of the nth reflection, $\Phi^{(n)}_{LL}(0)$ and $\Phi^{(n)}_{RR}(0)$ are autocorrelation functions at $\tau = 0$ of the nth reflection at the ears, and $\delta(t)$ is the Dirac delta function. If $w_n(t) \neq \delta(t)$, then A_n may approximately rewritten by

$$A_n \int_{\Omega_1}^{\Omega_2} |P(\omega)||W_n(\omega)|d\omega \Big/ \int_{\Omega_1}^{\Omega_2} |P(\omega)|d\omega,$$

where Ω_1 and Ω_2 are the lower and upper limits of the audible frequencies (rad/s), and $P(\omega)$ and $W_n(\omega)$ are the Fourier transforms of $p(t)$ and $w_n(t)$, respectively.

Since these correlations between the two ears have not been theoretically obtained, the long-time interaural crosscorrelations ($2T = 33$ s) for each single sound arriving at a dummy head were measured. The dummy head was constructed according to an acoustic measurement of the threshold level, so that the output signals of the microphones corresponded to the ear sensitivity.[9]

Considering the fact that the peak of the interaural crosscorrelation is usually obtained at $\tau = 0$ in rooms, the measured values only at the origin are listed in Table II with different horizontal angles of a single sound for several sources (Table I). The values of $\Phi_{LL}(0)$ and $\Phi_{RR}(0)$ corresponding to the average intensities of the sound at the left and the right ears are also listed in the table. These values are used to calculate the interaural crosscorrelation by Eq. (7).

The magnitude of the interaural crosscorrelation is defined by

$$\text{IACC} = |\Phi_{LR}(\tau)|_{\max} \quad \text{for } |\tau| < 1 \text{ ms}. \qquad (8)$$

To determine the minimum delay time required for Eq. (7) to hold, measured maximum values of the interaural crosscorrelation as a function of delay time Δt_1 have been discussed[1] (*spatial–binaural criterion*).

II. SIMULATION OF SOUND FIELDS

In accordance with Eq. (3), sound fields in concert halls were simulated by a system as shown in Fig. 2. A reverberation free signal is fed into a digital computer through an analog-to-digital (A/D) converter. The computer program provides the amplitude and the delay of early reflections ($n = 1,2$) and the subsequent reverberation relative to the direct sound ($n = 0$). The block diagram of the reverberator is shown in Fig. 3.[10] By use of four comb filters connected in parallel as shown in the figure, a highly irregular frequency response is produced similar to that in concert halls. The reverberation time of the reverberator is given by the loop gains g_i and delays τ_i of the different comb filters. The sound level decay of $-20 \log g_i$ dB for every trip around the feedback loop τ_i gives

$$T_i = \frac{60}{-20 \log|g_i|}\tau_i = \frac{3}{-\log|g_i|}\tau_i,$$

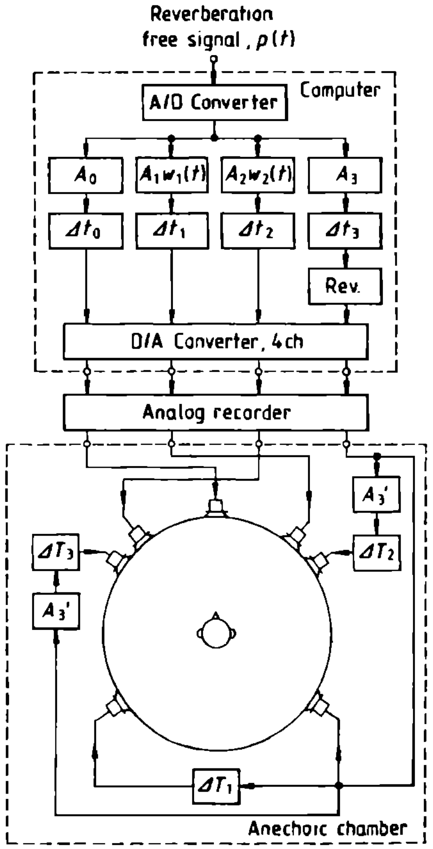


FIG. 2. Simulation system for the sound fields in concert halls. Δt_n: delays of reflections; A_n: amplitudes of reflections; ΔT_1, ΔT_2, ΔT_3: appropriate delays (18–40 ms) producing "subjective diffuseness" in reverberation signals.

and

$$T_{\text{sub}} = [T_i]_{\max} \quad \text{for } i = 1,2,3,4, \tag{9}$$

where T_{sub} is defined as the subsequent reverberation time.

In order to produce a proper density of reflections without changing the frequency response of the reverberator, two all-pass filters are connected in series with the comb filters. The density of reflections at any time t after the impulse excitation is given by

$$n_e(t) = \frac{1}{2} \sum_{i=1}^{I} \frac{1}{\tau_i} \frac{1}{\tau_a \tau_b} t^2. \tag{10}$$

The delays τ_a and τ_b of the all-pass filters should be chosen as $\tau_a, \tau_b < \tau_i$ ($i = 1,2,3,4$) so that they do not influence on the reverberation time from the comb filters. In order to produce the reverberation with "subjective diffuseness," discrete time delays of ΔT_1, ΔT_2, and ΔT_3 as shown in Fig. 2 were inserted and signals were then fed to the loudspeakers. The interaural crosscorrelation was adjusted by the location of loudspeakers around the listener's head.

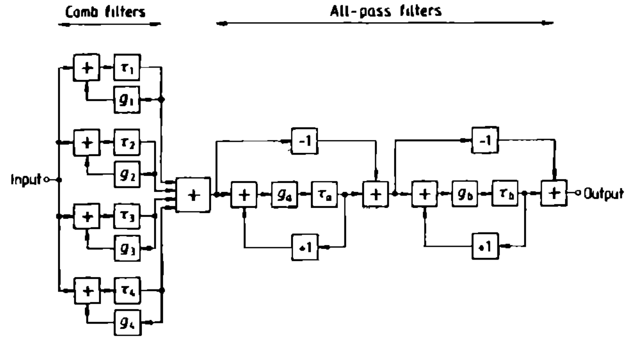


FIG. 3. Block diagram of the reverberator with four comb filters adjusting the subsequent reverberation time and two all-pass filters producing a proper density of reflections (after M. R. Schroeder[10]).

III. RESULTS OF SUBJECTIVE PREFERENCE JUDGMENTS

In order to examine whether the four objective parameters influence the subjective preference judgments independently, two of the four parameters were simultaneously varied while the other two were held constant at the preferred conditions. First, sound fields in a concert hall were simulated with various combinations of early reflections and subsequent reverberation after the early reflections.[6] Second, sound fields with various combinations of the listening level and the IACC were simulated.[5] Third, sound fields with various combinations of the subsequent reverberation time and the IACC[7] were simulated.

A. Preference versus delay time of early reflections and subsequent reverberation

The sound fields in a concert hall with the same plan as the Symphony Hall in Boston were simulated. In order to obtain a natural-sounding reverberator,[10] the values of delays given in Fig. 3 were chosen as

$$\tau_i = (23.5, 25.2, 27.0, 30.5, 7.3, 2.1) \times T_{\text{sub}} \text{ (in ms)}, \quad i = 1,2,3,4,a,b, \tag{11}$$

and

$$g_i = (0.85, 0.84, 0.83, 0.81, 0.70, 0.70), \quad i = 1,2,3,4,a,b.$$

In order to produce low magnitudes of the IACC in the sound fields[8] and to fix the spatial–binaural criterion, $h_{nL,R}(t)$, $n = 0,1,2,\ldots,6$, signals with discrete time delays were fed to the loudspeakers (No. 1,2,...,6) fixed at azimuth angles, i.e., $\xi_{1,2} = \pm 45°$ ($\pm 5°$) (the elevation angle $\eta = 0°$), $\xi_{3,4} = \pm 125°$ ($\eta = 0°$), and $\xi_{5,6} = \pm 55°$ ($\eta = 15°$), respectively. The pressures at both ear entrances of the listener in an anechoic chamber are expressed by

$$\begin{aligned} f'_{L,R}(t) = {} & p_s(t)*h_{0L,R}(t) + p_s(t)*A_1\delta(t-\Delta t_1)*h_{1L,R}(t) + p_s(t)*A_2\delta(t-\Delta t_2)*h_{2L,R}(t) \\ & + \sum_{n=3}^{N} [\, p_s(t)*A_n\delta(t-\Delta t_n)*h_{3L,R}(t) + p_s(t)*A_n\delta(t-\Delta t_n-\Delta T_1)*h_{4L,R}(t) \\ & + p_s(t)*A'_n\,\delta(t-\Delta t_n-\Delta T_2)*h_{5L,R}(t) + p_s(t)*A'_n\,\delta(t-\Delta t_n-\Delta T_3)*h_{6L,R}(t)\,], \end{aligned} \tag{12}$$

where $p_s(t) = p(t)*s(t)$, $s(t)$ is the radiation impulse response of the loudspeakers. The amplitude response of the loudspeakers for the frequency range 110 Hz to 8.5 kHz was in the range of $+3.5$ dB, and the difference between loudspeakers were within 1 dB. Also, $A_0 = 0$, $A_1 = -2.2$, $A_2 = -3.6$, $A_3 = -9.3$, $A'_3 = -11.3$, $A_n \geqslant -49.3$, and $A'_n > -51.3$ when $n \leqslant N$, $A_n = A'_n = -\infty$ $n \geqslant N$ (all in dB); and

$$\Delta t_1 = 22\times\mathrm{SD},\quad \Delta t_2 = 38\times\mathrm{SD},\quad \Delta t_3 = 47\times\mathrm{SD}, \qquad (13)$$

where SD corresponds to the scale of dimension of the auditoriums for early reflections. These delay times were chosen with reference to the result that the preferred delay of the second reflection was found by Eq. (17) of Ref. 4. Properly chosen delays $\Delta T_1 = 18$, $\Delta T_2 = 30$, and $\Delta T_3 = 40$ (all in ms) are inserted to make the four reverberation signals incoherent (Fig. 2). Consequently, the amplitude of the total reverberation decayed smoothly after the two early reflections until it dropped down by about 50 dB.

Paired comparison tests of the 16 sound fields for each source signal were conducted for changes in the temporal-monaural criteria only, i.e., the scale factor of the early reflections, SD, and the subsequent reverberation time, T_{sub}.[6] In order to change the autocorrelation function, several types of source signals were used, i.e., motif A, B, E, and speech. As discussed in the Sec. IIIB, the total sound pressure levels in the sound fields were adjusted to each preferred listening level.

Five second passages of each source signal, except for 10 s of motif A, were used for the paired comparison tests. Test signals were presented with a 1-s interval between the pairs. Each subject (13 subjects for motif A and 14 subjects

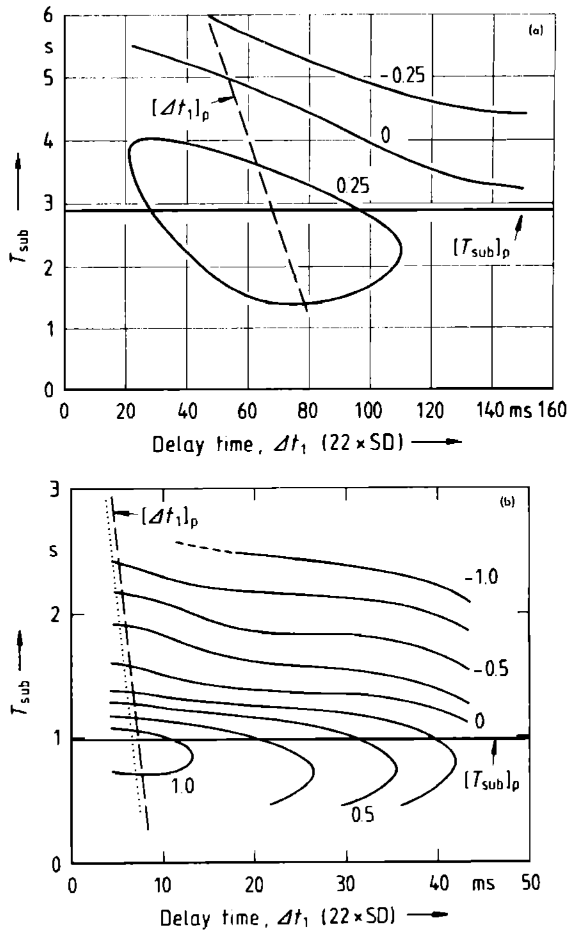


FIG. 4. Contour lines of equal preference for the sound fields versus Δt_1 (SD) and T_{sub}. The dashed line is the preferred value of Δt_1 calculated by Eqs. (14) and (15). The solid line is the preferred value of T_{sub} calculated by Eq. (18). (a) Music motif A (Gibbons); (b) music motif B (Arnold).

TABLE III. Analyses of variance for three tests described in Secs. IIIA, B, and C.

Test	Factor	Sum of squares	Degree of freedom	Mean square	F	Significance level	Contribution (%)
(a) Music motif A[a]							
	Δt_1(SD)	0.20	3	0.07	4.4	<0.05	14
A	T_{sub}	0.73	3	0.24	17	<0.01	65
	Residual	0.13	9	0.01	...		
	L. level	0.99	3	0.33	48	<0.01	27
B	IACC	2.61	2	1.30	187	<0.01	71
	Residual	0.04	6	0.01	...		
	T_{sub}	2.44	3	0.82	68	<0.01	89
C	IACC	0.17	3	0.06	5	<0.05	5
	Residual	0.11	9	0.01	...		
(b) Music motif B[a]							
	Δt_1(SD)	1.20	3	0.40	22	<0.01	13
A	T_{sub}	7.63	3	2.54	141	<0.01	84
	Residual	0.16	9	0.02	...		
	L. level	0.74	3	0.25	12	<0.01	24
B	IACC	1.90	2	0.95	47	<0.01	67
	Residual	0.12	6	0.02	...		
	T_{sub}	2.55	3	0.85	182	<0.01	79
C	IACC	0.64	3	0.21	46	<0.01	19
	Residual	0.04	9	0.01	...		

[a] Reliabilities (95%) of data for each parameter, as scale values plotted in Figs. 7, 8, 9, and 11, are less than ± 0.16 (Music motif A) and ± 0.21 (Music motif B).

for the other signals, 23 ± 3 years of age) judged which of the sound fields they preferred to hear in a concert hall or in a lecture room. The tests were performed for all combinations of the pairs, i.e., 240 pairs for each source signal, interchanging the order in each pair. Since they took about 5 h in total for each subject, sessions were limited to 12 min to avoid fatigue effects.

Scale values of preference, which are regarded as a linear psychological distance between the sound fields, were obtained by applying the law of comparative judgment (case V)[11,12] and were confirmed by the test of goodness of fit,[13] throughout this investigation. The agreement of judgments between the subjects for each source signal was substantial, thus responses of the subjects showed a similar tendency (according to the chi-square test, at 5% significance level).

The contour lines of equal preference were drawn by proportional allotment, as shown in Fig. 4, as a function of Δt_1 (or SD) and T_{sub}. It is obvious that the optimal conditions differ greatly between source signals. The conditions can be easily found at: $\Delta t_1 \simeq 70$ ms (SD$\simeq$3.5) and $T_{sub} \simeq 2.6$ s for motif A (Gibbons), and: $\Delta t_1 \leqslant 10$ ms (SD$\leqslant$0.5) and $T_{sub} \simeq 0.9$ s for motif B (Arnold). The tendency is clear: contour lines drawn at every 0.25 interval become more dense with decreasing coherence of the autocorrelation function τ_e. The most preferred initial time delay gap $[\Delta t_1]_p$ can be calculated by Eqs. (14) and (15). Calculated values of preferred initial time delay gaps, $[\Delta t_1]_p$, are shown by the dashed line in Fig. 4.

By applying analyses of variance to the scale values for each source signal, we learn that factors Δt_1 (SD) and T_{sub} are independent of each other because of the small residual (Table III, test A), even though both factors are temporal–monaural criteria.

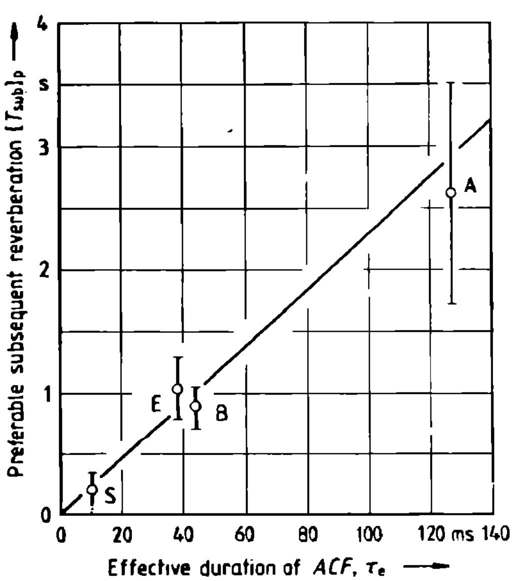


FIG. 5. Relationship between the preferable subsequent reverberation time and the effective duration of autocorrelation function (the ten-percentile delay). Ranges are obtained at 0.1 below the maximum scale values. The capital letters refer to source signals.

Figure 5 shows the relationship between preferred subsequent reverberation time and the effective duration of the autocorrelation function. The statistical correlation between them is 0.99 (at 1% significance level). The preferred subsequent reverberation time $[T_{sub}]_p$ is described by Eq. (18). For example, it has the value 2.9 s (23×0.127 s) for motif A. The solid lines in Fig. 4 are calculated values of $[T_{sub}]_p$.

As is clear from these results, the preferred reverberation time is almost completely independent of Δt_1 or SD. The total amplitude of reflections was set at $A \approx 4.1$, because the sound field at a rear position in a concert hall was simulated. When we examined the sound field at a frontal seat in the hall with $A \approx 1.1$, the preference results showed that a longer reverberation time was acceptable. The most preferred reverberation time, however, agree fairly well with the calculated results of Eq. (18). This will be shown later (Fig. 10, Sec. V). Note that the total amplitude of reflections is inversely related to "Deutlichkeit" as defined by Thiele.[14] It may not, however, determine the preferred condition.[4]

B. Preference versus listening level and IACC

Next we examine the independence of the subjective preference between the listening level and the interaural crosscorrelations.[5] The sound fields were simulated with the computer system described above. In order to obtain the optimal conditions, other parameters were fixed at the preferred values. Accordingly, the subsequent reverberation times were kept constant: 3.0 s for music motif A and 1.0 s for music motif B. The initial time delay gaps between the direct sound and the first reflection were kept at 80 ms for music motif A and 20 ms for music motif B. These values were calculated by Eqs. (14) and (15) with $A = 2.0$. Also, the time delays of reflections were kept at $\Delta t_2 - \Delta t_1 = 0.8 [\Delta t_1]_p$ and $\Delta t_3 - \Delta t_2 = 0.64 [\Delta t_1]_p$. The values of IACC were adjusted in the range of 0.4–1.0 by changing the direction of loudspeakers which produce the reflections. Loudspeaker system (a) included the reflections with the optimal angles $\xi = \pm 55°$ (IACC$\approx$0.4), while system (b) contained early reflections from the listener's median plane and from certain angles in the horizontal plane (IACC$\approx$0.7) and system (c) included only reflections from the median plane (IACC = 0.98: absorbing side walls).

The preference judgments were conducted for the 12 fields with 16 subjects. Most of subjects judged all sound fields in the above two experiments. The contour lines of equal scale values of preference are shown in Fig. 6. Obviously, for a given constant listening level, the sound fields with a smaller IACC are always preferred. Preferred listening levels depend upon the source music being presented, but they are hardly affected by the IACC. The preferred listening levels are found in the ranges of 77–79 dBA for music motif A and 79–80 dBA for music motif B. (Note that these results are not related to "musicality," because of the short music pieces produced.)

Results of the analyses of variance with the scale values of preference for each motif clearly indicated that the factors

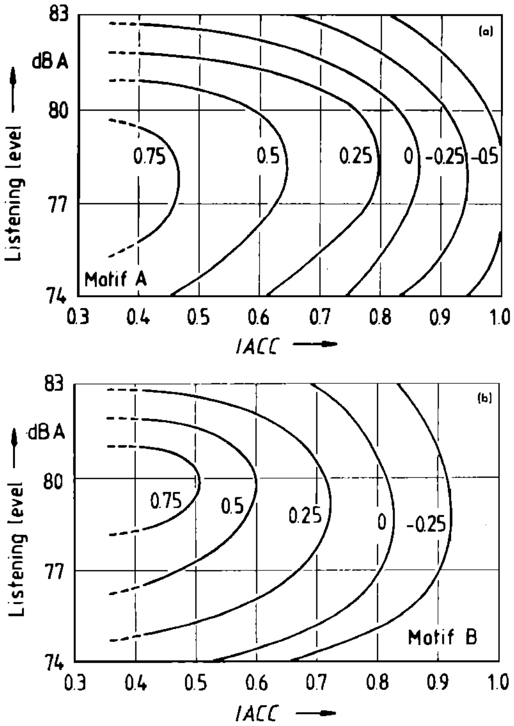


FIG. 6. Contour lines of equal preference for the sound fields versus the IACC and the listening level at a maximum amplitude measured in dBA "slow." (a) Music motif A (Gibbons); (b) music motif B (Arnold).

listening level and IACC, are independent on the subjective preference judgments (Table III, test B). It has been shown by means of example that the calculated scale values applying the "principle of superposition" agree fairly well with those experimental values.[5]

C. Preference versus subsequent reverberation time and IACC

In the third step, the independent effect of the subsequent reverberation time and the interaural crosscorrelation on subjective preference was examined.[7] Also observed was the consistency of scale values of preference from the different series of preference tests. As usual, the other two parameters were fixed at the preferred conditions. The values of IACC were adjusted in the range of 0.31–0.93 by changing the direction of loudspeakers located around the listener. The values of the reverberation time were adjusted in the range of 1.5–6.0 s for music motif A and 0.5–2.5 s for music motif B. The preference judgments were conducted for 16 sound fields with eight subjects.

The results of analyses of variance confirm that the reverberation time and the IACC independently influence subjective preference (Table III, test C).[7]

The most preferred reverberation times obtained here confirm Eq. (18).

As is widely accepted, the preferred condition can generally be obtained by minimizing the magnitude of the IACC. However, the maximum value of the crosscorrelation must be maintained at $\tau = 0$ to ensure frontal localization of the sound source. The preference data satisfied by this condition will be shown later (Sec. V).

D. Agreements with other results of preference judgments

Previous subjective preference judgments were performed in the sound fields with four multiple early reflections and with two early reflections and subsequent reverberation time.[2,3]

It is reconfirmed from the results that the most preferred initial time delay gap can be found by Eqs. (14) and (15). Also confirmed is the fact that factors Δt_1 (SD) and the IACC are independent on the subjective preference judgments.

In Sec. V, consistency of preference values obtained by difference series of judgments with different sound sources will be shown.

IV. OPTIMAL DESIGN OBJECTIVES

The optimal design objectives can be described in term of the preference judgments, which are related to objective parameters describing the sound signals arriving at both ears. They clearly lead to the following comprehensive criteria for the optimal design of concert halls.

A. Listening level (temporal–monaural criterion)

The listening level is expressed by the autocorrelation function of sound signals with the delay $\tau = 0$. Thus this should be included in the temporal criteria. The listening level is, of course, a primary criterion for listening to the sound in concert halls. The preferred listening level depends upon the music motif and the particular passage being performed. As a matter of course, equal listening level distribution throughout the concert hall is ideal. The preferred levels obtained by 16 subjects are in peak ranges of 77–79 dBA for music motif A with a slow tempo, and 79–80 for music motif B with a fast tempo. Both values do not greatly differ from 79 dBA for this kind of music as far as the peak values are concerned.

B. Delay time of early reflections (temporal–monaural criterion)

An approximate formula was derived to describe the relation between the duration of the autocorrelation function, the total amplitude A of the reflections, and the most preferred delay time $[\Delta t_1]_p$. It is well described by the identity[3]

$$[\Delta t_1]_p = \tau_p \tag{14}$$

such that

$$|\phi_p(\tau)| \leqslant kA^c, \quad \text{for } \tau > \tau_p, \tag{15}$$

where $k = \text{const.}\ (= 0.1)$, $c = \text{const.}\ (= 1.0)$. Its amplitude may be chosen by

$$A = \left(\sum_{n=1}^{\infty} A_n^2 \right)^{1/2} \tag{16}$$

and the delay time set equal to that of the strongest reflection,[2] which is usually the first reflection according to the inverse square law. Thus the preferred initial time delay is found to be identical with the time delay at which the envelope of the autocorrelation function reaches a small value, $0.1A$ ($\tau_e = \tau_p$, when $A = 1$. If the envelope of autocorrelation function is exponential, then Eq. (15) is simply $\tau_p = (1 - \log_{10} A)\tau_e$. These relationships also hold for sound fields with a single reflection.[1] The preferred delay of the second reflection, in relation to the direct sound, is given by[4]

$$[\Delta t_2]_p \approx 1.8\tau_p. \tag{17}$$

C. Subsequent reverberation time after the early reflections (temporal–monaural criterion)

The most preferred subsequent reverberation time is approximately described by[6]

$$[T_{\text{sub}}]_p \approx 23\tau_e. \tag{18}$$

The coefficient in the equation was thought to depend on the ratio $1/A^2$ between the energy of the direct sound and that of the total reflections including the subsequent reverberation. However, as was mentioned in Sec. IIIA, the coefficient 23 is unchangeable for sound fields with two early reflections and subsequent reverberation. In our investigations, the energy ratio $1/A^2$ measured at the judgments was between 0.06 and 1. The behavior of scale values of preference as a function of the subsequent reverberation time depends upon the energy ratio as we shall see later (Fig. 10, Sec. V).

D. Incoherence at both ears (spatial–binaural criterion)

All data previously obtained indicate negative correlation between the values of the IACC and the subjective preference. This holds under the condition that the maximum value of interaural crosscorrelation at $\tau = 0$ is maintained. In order to obtain a small value of the IACC in the most effective manner, the directions of arrival of the early reflections at the listeners are kept within a certain range of angle from the median plane: i.e., $\pm(55° \pm 20°)$. It is obvious that the sound arriving from median plane makes the IACC values larger. Also, sound arriving from 90° in the horizontal plane is not always advantageous, because the similar "detour" paths around the head to both ears for the frequency above 500 Hz cannot improve the interaural crosscorrelation effectively. Due to a large interaural level difference and a proper interaural time difference, the optimal angles are, therefore, found in the range centered on 55°.

V. SCALE VALUES OF PREFERENCE FOR EACH PARAMETER

The linear scale value of preference is derived by the law of comparative judgments. As examined in Sec. III, each criterion contributes independently to the scale values of subjective preference. And, we shall show in this section, the scale values are reasonably consistent, i.e., the unit of the scale values is nearly constant for each parameter. Accordingly, the principle of superposition to get a total preference for the sound fields at each seat will be directly applied.[5,15] (The validity of superposition has been discussed by means of examples.[5])

All available data obtained heretofore with more than three plots in relation to each objective parameter will be employed. Each objective parameter is normalized by its most preferred value and the scale value of preference for sound fields can be reduced into a general form.

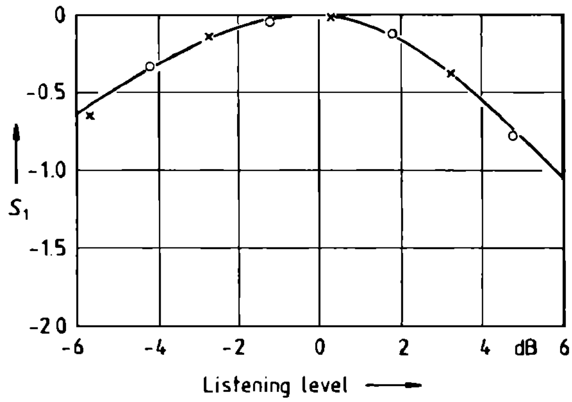


FIG. 7. Scale values of preference as a function of the listening level. Data are obtained by the tests described in Sec. IIIB (Ref. 5). O: Music motif A; ×: music motif B. The scale value at the most preferred listening level is adjusted to zero.

A. Listening level

Scale values of preference as a function of listening level (S_1) are shown in Fig. 7. In order to examine consistency, the scale value at the most preferred level is adjusted to zero. It is clear that the scale values indicated by different symbols are reasonably consistent with each other for different sound sources that were obtained by different series of preference judgments. This, in turn, reveals that the unit of scale values obtained by the law of comparative judgment is nearly constant. An approximate formula at any sound pressure P may be found by the average solid curve in Fig. 7, so that

$$S_1 \approx -\alpha_1 |X_1|^{3/2}, \tag{19}$$

where $X_1 = 20 \log P/[P]_p$, $[P]_p$ being the most preferred sound pressure depending upon music piece and

$$\alpha_1 \approx \begin{cases} 0.07, & X_1 > 0, \\ 0.04, & X_1 < 0. \end{cases}$$

Similar approximations may be obtained for the relationship between the scale value of preference and other objective parameters.

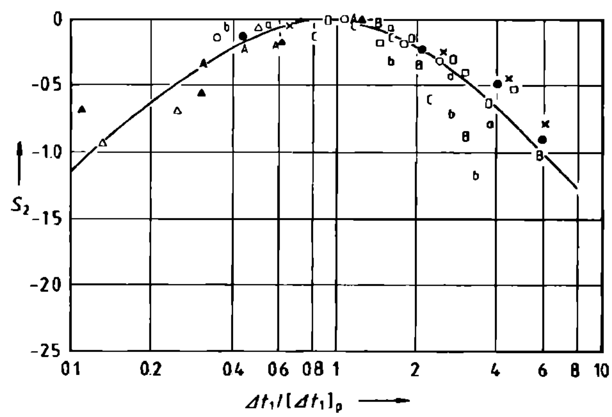


FIG. 8. Scale values of preference as a function of the initial time delay gap, Δt_1. Different symbols indicate the scale values obtained by different test series: Data (O, ×, □, ●) are obtained by the tests described in Sec. IIIA (Ref. 6); (a, b) are in Ref. 7; (A, B, C, D) are in Ref. 3; and (△, ▲) are in Ref. 2. O,a,A,△: music motif A; ×, b,B,▲: music motif B; C: music motif C; D: music motif D; □: music motif E; ●: speech S. The scale value at the most preferred delay time is adjusted to zero.

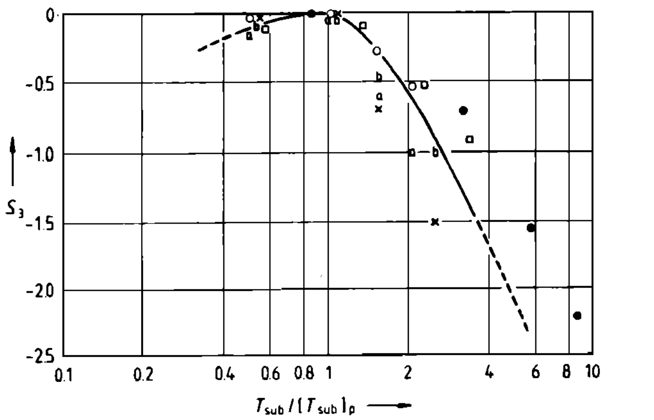


FIG. 9. Scale values of preference as a function of the subsequent reverberation time. Different symbols indicate the scale values obtained by different test series: Data (O, ×, □, ●) are obtained by the tests described in Sec. IIIA (Ref. 6); and (a,b) are in Sec. IIIC (Ref. 7). O,a: music motif A; ×, b: music motif B; □: music motif E; ●: speech S. The scale value at the most preferred reverberation time is adjusted to zero.

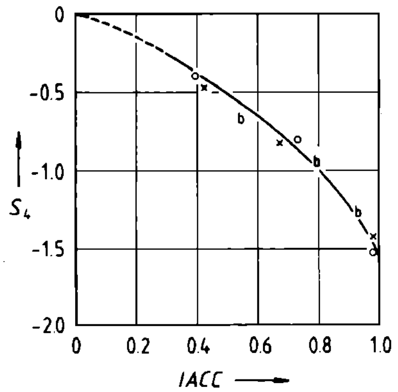


FIG. 11. Scale values of preference as a function of the IACC. Data (O, ×) are obtained by the tests described in Sec. IIIB (Ref. 5); and data (b) are in Sec. IIIC (Ref. 7). O: music motif A; ×, b: music motif B. The maximum value of interaural crosscorrelation must be maintained at $\tau = 0$ to ensure frontal localization of the sound source.

B. Initial time delay gap

Scale values of preference as a function of the initial time delay gap between the direct sound and the first reflection (S_2) are shown in Fig. 8. The abscissa is normalized by the most preferred delay [Eq. (14)]. No fundamental differences among the effects of different sound sources on the behavior of scale values are observed. From the solid curve in Fig. 8, we may have

$$S_2 \approx -\alpha_2 |X_2|^{3/2}, \tag{20}$$

where $X_2 = \log \Delta t_1 / [\Delta t_1]_p$, and

$$\alpha_2 \approx \begin{cases} 1.42, & X_2 \geqslant 0, \\ 1.11, & X_2 < 0. \end{cases}$$

C. Subsequent reverberation time

When $A = 4.1$, scale values of preference as a function of the subsequent reverberation time (S_3) are shown in Fig. 9, which demonstrates a reasonable degree of consistency between data. The abscissa is normalized by the most preferred subsequent reverberation time [Eq. (18)]. If we may compare the average curves in Figs. 8 and 9 at the same value of X greater than unity, we then find the reverberation time is more critical in the preference judgments than in the initial time delay gap. When $A = 1.1$, however, the reverberation time becomes more critical if the value X is smaller than unity as indicated by the dashed curve in Fig. 10. Owing to the two solid curves in the figure and its interpolation, an approximation formula yields

$$S_3 \approx -\alpha_3 |X_3|^{3/2}, \tag{21}$$

where $X_3 = \log T_{sub}/[T_{sub}]_p$, and

$$\alpha_3 \approx \begin{cases} 0.45 + 0.74\,A, & X_3 \geqslant 0, \\ 2.36 - 0.42\,A, & X_3 < 0. \end{cases}$$

It is worth noticing that the total amplitude of reflections A is included in the formula, so that it has an effect on the preference, when $T_{sub} \neq [T_{sub}]_p$.

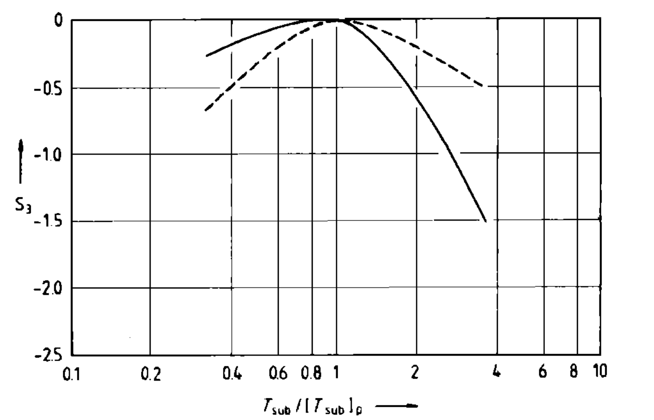


FIG. 10. Average scale values of preference as a function of the subsequent reverberation time and as a parameter of the total amplitude of reflections —: A = 4.1; - - -: A = 1.1. The scale value at the most preferred reverberation time is assumed to be independent of the total amplitude A and is adjusted to zero (note that the value A may not determine the preferred condition as described in Ref. 4, for example).

D. IACC

Scale values of preference as a function of the IACC (S_4) are shown in Fig. 11. The consistency of the data is much more satisfactory. It is remarkable that the preference drops more rapidly when the value of IACC approaches unity. Thus it is recommended to keep the IACC smaller than 0.5 for each sound source location on the stage.

From the figure, we may approximately obtain

$$S_4 \approx -\alpha_4 |X_4|^{3/2}, \tag{22}$$

where $X_4 =$ IACC without any logarithmic transformation and

$$\alpha_4 = 1.45.$$

Thus the scale values of preference have approximately been formulated in term of the 3/2 power of the normalized objective parameters in common, in which the objective parameters are expressed in the logarithmic form for the temporal-monaural criteria, X_1, X_2, and X_3, but a linear value of IACC for the spatial–binaural criterion X_4. In other words, one may assume logarithmic–monaural detectors based on the autocorrelation function process and a linear–binaural detector based on the crosscorrelation process in the auditory pathway. Also, assumed is a preference evaluator based on the "3/2 power process" somewhere in the upper part of the brain.

Considering the fact that T_{sub} is almost independent of the seat position, therefore, the total preference distribution in a given concert hall is mainly determined by the listening level (S_1), the initial time delay gap (S_2) and the magnitude of the interaural crosscorrelation (S_4) as will be discussed in the following section.

VI. CALCULATION OF OBJECTIVE VALUES AND SUBJECTIVE PREFERENCE IN A CONCERT HALL

A. Objective values

As a typical example, we shall discuss quality of sound fields in a concert hall with a similar shape to the Symphony Hall in Boston. The plan, with dimensions of 25 m width and 50 m length, is outlined in Figs. 12–20. The height of the hall is 18 m. For the sake of simplicity, reflections and scattering by balconies and the very early reflection by the floor are not taken into consideration. Also, it is supposed that a single source is located at the center, 1.2 m above the stage floor, which is 1 m above the audience floor level. Receiving points at a height of 1.1 m above the floor level correspond to the ear's position.

According to the image method, 30 reflections with their amplitudes, delay times, and directions arriving at the listeners, were taken into account. The sound pressure level

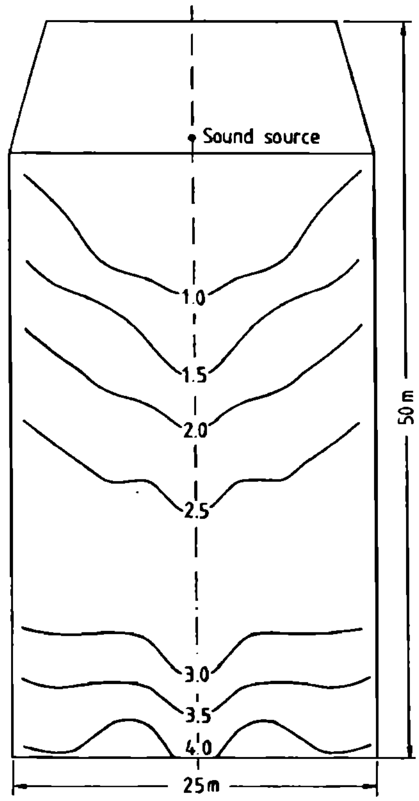


FIG. 13. Contour lines of equal total amplitude of reflections A in the concert hall calculated by Eq. (16).

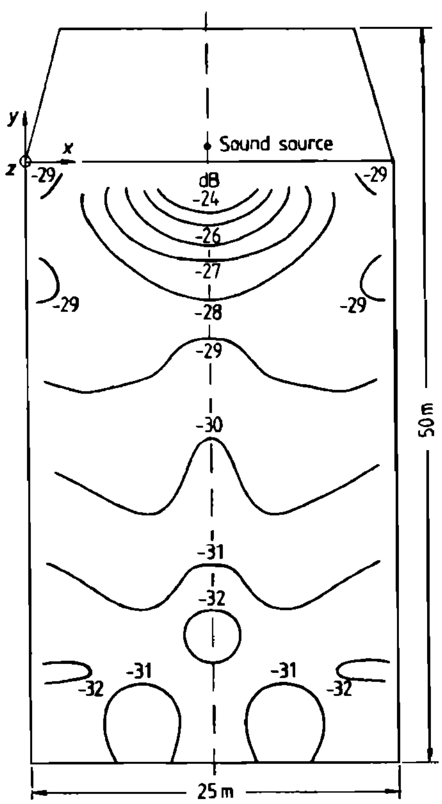


FIG. 12. Contour lines of equal relative listening level in the concert hall calculated by Eq. (23).

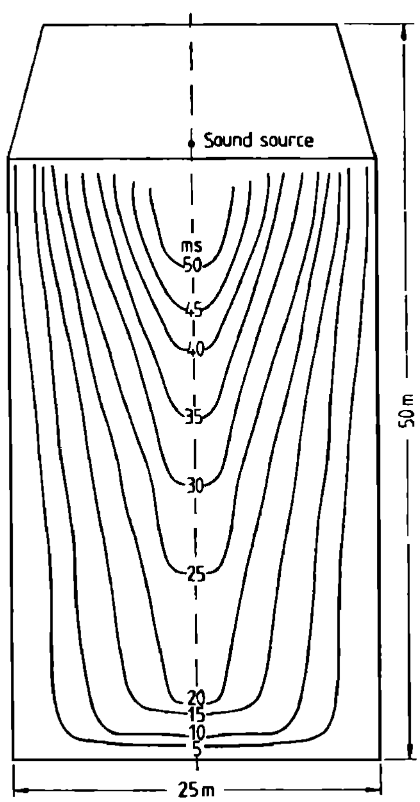


FIG. 14. Contour lines of equal initial time delay gap Δt_1 in the concert hall calulated by Eq. (24).

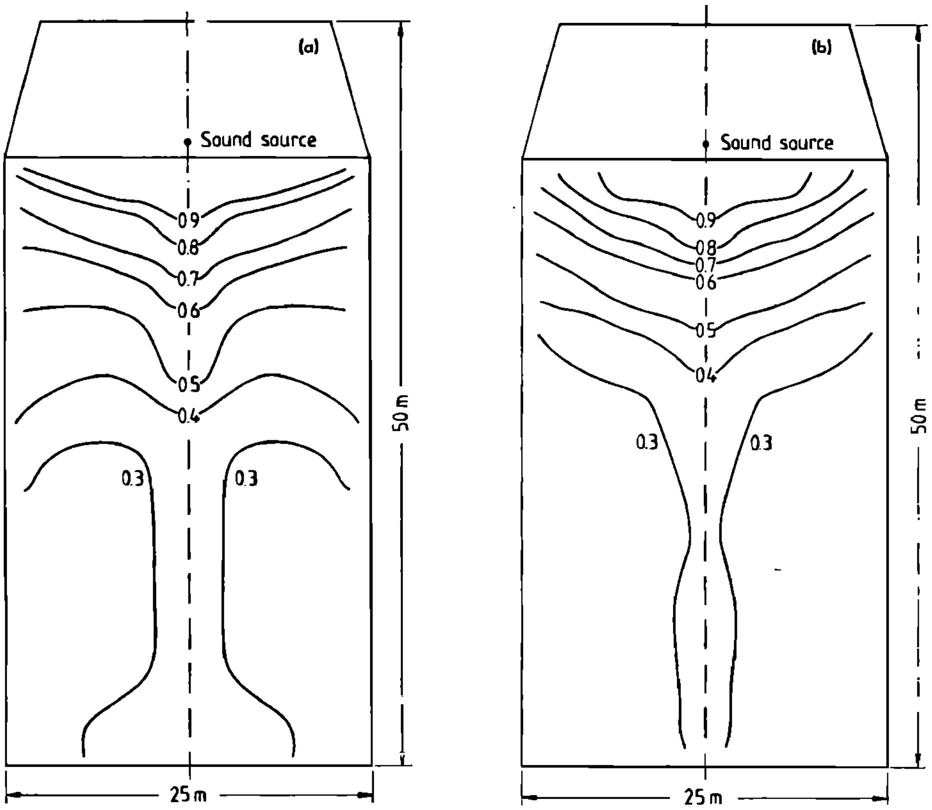


FIG. 15. Contour lines of equal IACC in the concert hall calculated by Eq. (7) with correlation values listed in Table II. (a) Music motif A; (b) music motif B.

SPL, is given by

$$\mathrm{SPL} = \mathrm{PWL} + L,$$

$$L = 10 \log(1 + A^2) - 20 \log d_0 - 11(\mathrm{dB}), \tag{23}$$

where PWL is the power level of the source [$= 10 \log(W/10^{-12})$], W being the acoustic power of the sound source; A is the amplitude as defined by Eq. (16) and d_0 ($= |r - r_0|$) is the distance between the source and the positions of the listeners. Since the power level itself is unknown at the design stage of concert halls, only the relative term L can be calculated at each seat. Contour lines of equal L values are shown in Fig. 12 and those of equal amplitude A are shown in Fig. 13. As is shown in Fig. 12, the rapid attenuation near the source for $d_0 < 12$ m is due to the inverse square law for the direct sound; for $d_0 > 12$ m, the value of L is almost independent of the distance and it is in the range of $\pm$ 1.5 dB.

The initial time delay gap between the direct sound and the first reflection is simply obtained by

$$\Delta t_1 = (d_1 - d_0)/c, \tag{24}$$

where d_1 is the pathlength of the first reflection and c is the wave velocity. Contour lines of equal time delay are shown in Fig. 14. It is quite natural that the largest time delay occurs on the center line near the source in concert halls.

Calculations of the IACC are carried out by means of Eqs. (7) and (8) at each seat. The values of correlations needed for the calculations are listed in Table II. In order to obtain the values of correlations for each single reflection with arbitrary angles to the listener (ξ,η) the equivalent angle measured from the median plane is given by

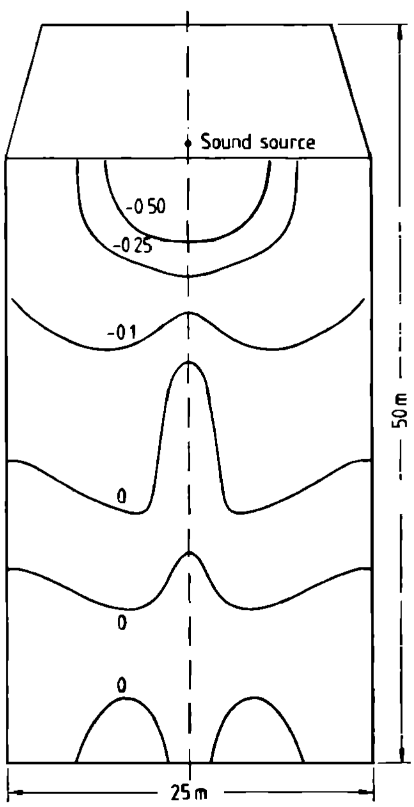


FIG. 16. Contour lines of equal preference for listening level (S_1) in the concert hall, assuming that the most preferred listening level is obtained on the center line at 20 m from the sound source. The value approaching to zero is preferred.

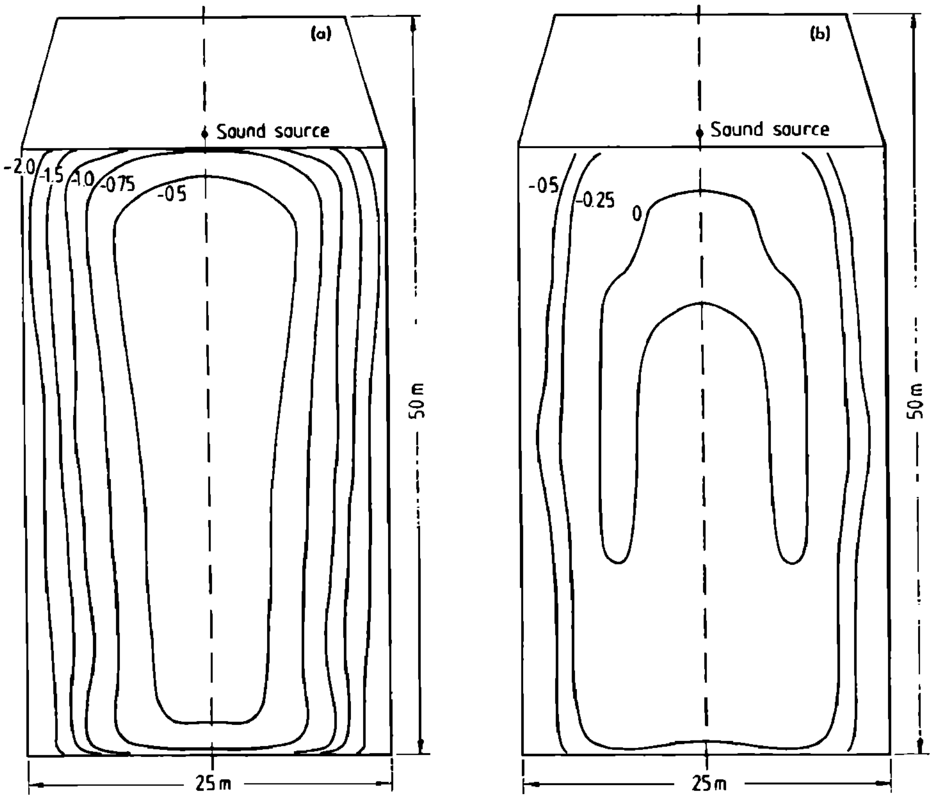


FIG. 17. Contour lines of equal preference for the initial time delay gap (S_2) in the concert hall. (a) Music motif A; (b) music motif B.

$\sin^{-1}$ ($\sin\xi \cos\eta$) and is substituted for the horizontal angle ξ. This approximation is useful, because the interaural cross-correlation is mainly determined by the angle measured from the median plane. Contour lines of the equal IACC value for music motif A and B are shown in Fig. 15. Obviously, values do not greatly differ according to the kind of music. Another interesting fact is that values on the center line in the hall become larger than those of other parts. This tenden-

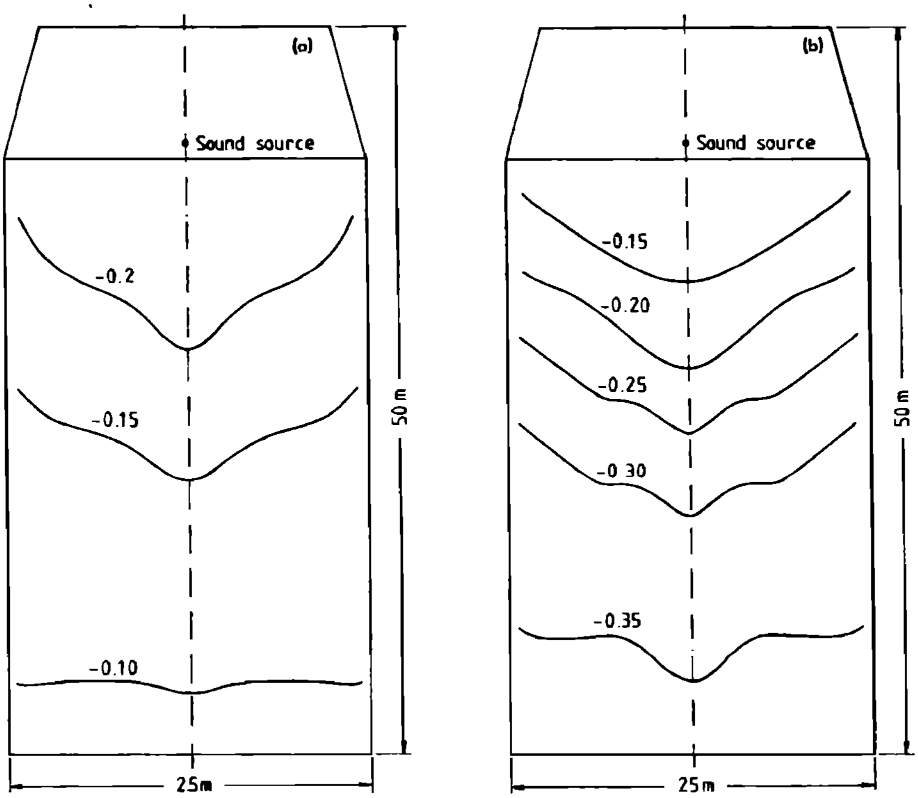


FIG. 18. Contour lines of equal preference for the subsequent reverberation time (S_3) in the concert hall. The values depend upon the total amplitude of reflections A (see Fig. 13). (a) Music motif A; (b) music motif B.

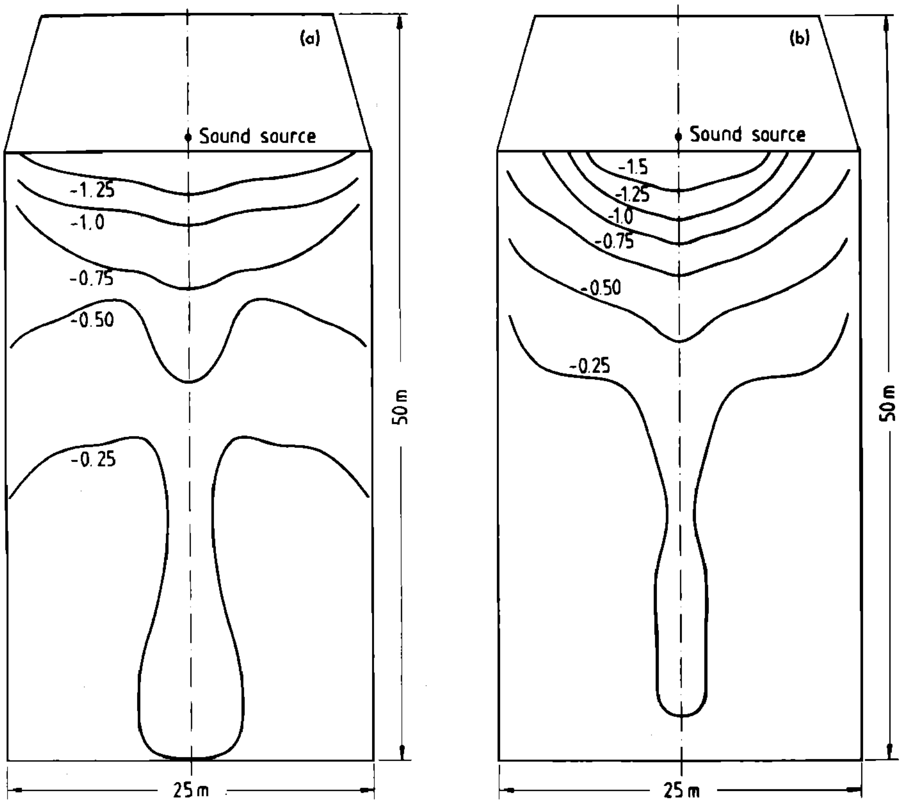


FIG. 19. Contour lines of equal preference for the IACC (S_4) in the concert hall. (a) Music motif A; (b) music motif B.

cy may be emphasized in existing concert halls with symmetrical acoustic properties, because coherent reflections arrive simultaneously at the listeners on the center line (see Fig. 5 in Ref. 1).

B. Scale values of preference

According to approximate formulae given by Eqs. (19)–(22), we can obtain scale values of preference for each objec-

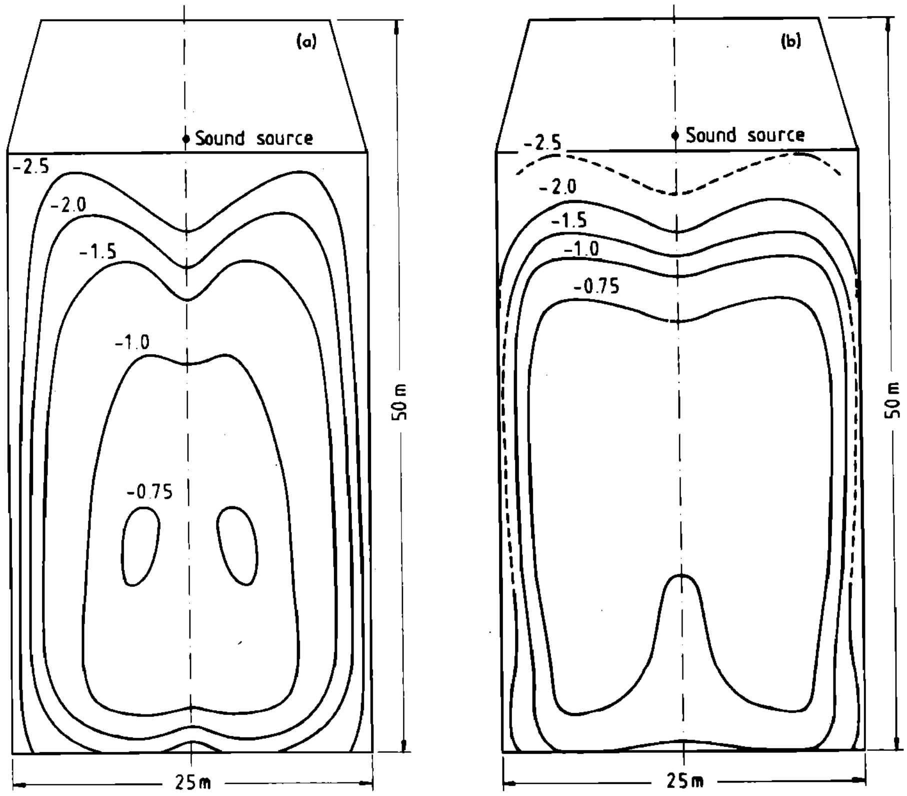


FIG. 20. Contour lines of equal subjective preference in total for four objective parameters (S) in the concert hall. Greater preference values can be found for music motif B at each seat in the concert hall than those for music motif A, because of the different effective duration of autocorrelation function of the sources. (a) Music motif A; (b) music motif B.

tive parameter at each seat.

Assuming that the preferred listening level is obtained on the center line 20 m distant from the source in the hall ($L = -30$ dB in Fig. 12), the scale values of preference at other seat positions can be derived by use of Eq. (19). Results of contour lines in the hall are illustrated in Fig. 16. This procedure may be applied for any kind of music source, because the distribution of the listening level is not different among the sound sources.

After getting the most preferred delay time at each seat by means of Eqs. (14) and (15), the scale values of preference for the initial time delay are calculated by Eq. (20). Contour lines of equal preference for music motifs A and B are shown in Fig. 17. For motif A, no optimum conditions with zero scale value are appeared in this hall, because of the long effective autocorrelation function.

On the other hand, the effective duration of the autocorrelation function of motif B is slightly short for the hall, so that the optimum area with zero scale value is limited in a tuning-fork-shape as shown in Fig. 17(b). The total amplitude of reflections shown in Fig. 13 is increased with increasing distance from the stage, and this causes the decline of preference in the rear seats for motif B.

The scale values of preference for the subsequent reverberation time are calculated by Eq. (21) with the value of A. The resulting preference distribution for the two motifs is shown in Fig. 18. The slight distribution of preference is caused by the effects of the amplitude A. In this calculation, the "actual reverberation time" is supposed to be 1.8 s throughout the hall. The preferred reverberation time $[T_{sub}]_p$, however, is 2.9 s for motif A and 0.99 s for motif B, thus values of $T_{sub}/[T_{sub}]_p$ are 0.62 and 1.82, respectively.

The scale values of preference for the IACC at each seat (Fig. 15) can be obtained by Eq. (22). The contour lines of equal preference in the hall for the two motifs are shown in Fig. 19. A similar spatial distribution in the hall can be found for the both motifs.

Based on the principle of superposition which has been described in Sec. V, the total scale value of preference for the sound fields in concert halls can be obtained by

$$S = \sum_{i=1}^{4} S_i \approx -\sum_{i=1}^{4} \alpha_i |X_i|^{3/2}. \quad (25)$$

Thus we can supply a negative value of preference counted from the possible optimum condition at each seat prior to the final architectural design of concert halls.

For example, contour lines of the total scale value of preference for the two motifs in the hall are shown in Fig. 20. Obviously, if different music is performed in a given concert hall, then the total scale value changes according to its autocorrelation function. Comparing the results for motif A and B shown in Fig. 20, motif B has more advantage for this concert hall. Neither music, however, is optimum for the hall. The most preferred and suitable music motif may be selected by means of the effective duration of the autocorrelation function. For instance, music motif D (Symphony No. 102 by Haydn) with $\tau_e = 65$ ms would seem to be one of the most suitable pieces of music to be performed in the concert hall.

Due to the coloration effects,[1,16] the preference value appear to be greatly decreased at the seats near the boundary walls (see Fig. 8, $\Delta t_1 \rightarrow 0$).

VII. DISCUSSION

So far, the preference for a single sound source located on the stage has been discussed, but preference may be calculated for every sound source location.

There are, of course, other minor objective parameters that may affect the preference. For example, spectral effects of reflections and reverberation have not been taken into account. As far as speech signals are concerned, however, the most preferred condition was obtained for the perfectly reflecting wall.[17] Also, a recent result of preference judgments with motif A shows that the effects of reverberation time as a function of frequency seem to be insignificant.[18]

VIII. CONCLUSIONS

The results of experiments described in this paper demonstrate that the four significant objective parameters affect, almost independently, the subjective preference judgments of simulated concert hall sound fields. The linear scale values of preference which are obtained by applying the law of comparative judgment indicate sufficient consistency among the different series of preference judgments with different music sources.

Consequently, after computing each objective value, the total preference value at each seat in the concert hall under design can be calculated.

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