Computational Systems for Sound Fields,

as Tools in Design and Diagnosis

January 2000

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（音場の設計支援ならびに診断システム）

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Preface

This dissertation is submitted for the Doctor of Philosophy Degree to The Graduate School of Science and Technology, Kobe University, Japan.

The theme of this study is using computational systems as tools in design and diagnosis. The main feature of this computational system is that the whole system is based on a subjective preference theory that is itself based on a model of the auditory-brain system. Such a system could be expected to lead to many important acoustic discoveries, because its design and diagnostic function include the objective and the result of the study of subjective preference based on the auditory-brain system model. Through the course of these studies, such discoveries were made and incorporated into the system, making them into its own features and characteristics.

Masatsugu Sakurai

Acknowledgments

I would like to express my thanks to Professor Yoichi Ando, Kobe University, and the members of Ando Laboratory, especially, Mr. Hiroyuki Sakai, Dr. Shin-ichi Sato, and Mr. Yukio Suzumura, and Mr. Shinichi Aizawa, Yoshimasa Electronic Inc., who has helped me in many respects concerning the system development for this study.

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Chapter 1

Introduction

1.1. Previous Studies Relating to Computational Systems for Sound Fields

Introduction

To realize an excellent sound field in a concert hall, it is important to identify the orthogonal factors influencing subjective evaluation. Using dissimilarity tests, Yamaguchi found two significant physical factors, the sound pressure level and the reverberation characteristics [1]. Edward also tested dissimilarity, and reported the early-echo pattern, the reverberation time RT, and the volume level to be the significant factors [2]. Schroeder, Gottlob, and Siebrasse [3] avoided using ill-defined adjectives, such as "intimate", "warm", "rich", and "clear", by conducting subjective preference tests. They found two significant factors, RT and the interaural crosscorrelation (IACC), defined by Damaske and Ando [4]. Kimura and Sekiguchi found reverberance and loudness to be significant factors [5]. Wilkens found that perception of strength and extension of sound source, perception of clarity, and tone color are significant subjective attributes [6]. Systematic investigations by Ando were made to find the orthogonal factors in subjective preference for sound Gelds [7]. Ando described four orthogonal physical factors (the listening level LL, the initial time delay gap Δt_1, the subsequent reverberation time T_{sub}, and the IACC), which determine the scale values of subjective preference for simulated sound Gelds. From subjective questionnaires, Barron found that the reverberation time, the early decay time, the early-to-late sound index C_{80}, the total sound level, and the early lateral energy fraction were significant subjective factors [8, 9]. Beranek suggested adding two more factors to Ando's four physical factors, the Bass ratio (BR) and the Surface Diffusivity Index (SDI) [10]. Problems with regard to both objective and subjective orthogonalities in above two investigations still remain. As far as subjective preference is concerned, it is worth noting that the sound fields in a real concert hall for listeners facing to the performers (IACC is obtained at $\tau = 0$) can be described by only four orthogonal physical factors [11, 12].

The theory of subjective preference

Four orthogonal physical factors have been identified from the systematic investigation of sound fields simulated by computer and listening tests (paired-comparison tests) [7]. Several reverberation free music signals were used to simulate sound fields from which fully independent physical parameters of the sound signals could be determined. The law of comparative judgments enables us to construct a linear scale value of subjective preference. The optimum design objectives can be described in terms of subjectively preferred sound qualities, which are related to comprehensive criteria consisting of temporal and spatial parameters describing the sound signals at the two ears.

(1) Listening level

The listening level is, of course, the primary criterion for listening to the sound field in concert halls. The absolute preferred listening level depends upon the musical program and the particular passage being performed. The listening level LL is given by

$$LL = 10 \log (1+A^2) - 20 \log d_0 - 11[\mathrm{dB}], \quad (1)$$

where A is the total pressure amplitude of the early reflections and subsequent reverberation, and d_0 (= | r - r_0|) is the distance between the source and the listener's position.

(2) Initial time delay gap

The relationship between the autocorrelation function (ACF) of the source signals, the total pressure amplitude of the early reflections and subsequent reverberation, and the most preferred delay time of a single reflection has been obtained from the approximate formula [7, 13]

$$[\Delta t_1]_p \qquad (1 - \log_{10} A)\,\tau_e \quad (2)$$

where the total amplitude of the reflections A is given by

$$A = \sqrt{A_1^2 + A_2^2 + \ldots + A_N^2} \quad (3)$$

A_i (i = 1, 2, ..., N) being the pressure amplitude Of each reflection relative to that of the direct sound, N is selected a large number for the convergence. The quantity τ_e is the effective duration of the ACF with a 10-percentile delay, defined as the delay at which the envelope of the normalized autocorrelation function becomes 0.1.

(3) Subsequent reverberation time

For flat frequency characteristics of reverberation, the preferred subsequent reverberation time after early reflections is described simply in terms of the effective duration of ACF of the source signals [14, 15], as given by

$$[T_{sub}]_p \quad 23\,\tau_e \tag{4}$$

(4) IACC

The IACC and subjective preference show a negative correlation for all available data. That is, listeners prefer dissimilar signals for the left and right ears. This holds only under the condition that the maximum value of the interaural crosscorrelation function at an interaural time delay equal to zero is maintained to ensure frontal localization of the source [12]. Otherwise, an image shift of the source or an unbalanced sound field may be perceived, and the value of preference decreases.

(5) A-value

The A-value is not included in the orthogonal physical factors, however, it is a factor used for calculating the most preferred delay time of a single reflection related to the autocorrelation function (ACF) of the source signals (equation 2). The A-value also is related to calculating the listening level by equation (1). It is worth noting that the "Hallradius"[16] defined by the distance between the source position and the positions in which the pressure amplitude of the early reflections and subsequent reverberation is equal to the total pressure amplitude of the direct sound (A = 1).

Independent effects of the four objective factors were examined by several experiments as shown in Table 1.1 [14, 17, 18, 19, 20]. The theory of subjective preference for each seat in a concert hall is described based on these results. The linear scale value of preference has been obtained using the law of comparative judgments [21] and reconfirmed by the goodness of fit [22]. Furthermore, the units derived from experiments with different sound sources and different subjects were almost equal [7], so the scale values may be added to give

$$S \quad s_1 + s_2 + s_3 + s_4 \tag{5}$$

where s_i (i = 1,2,3,4) is the scale value obtained in relation to each objective parameter. Equation (5) indicates four-dimensional continuity. Since the scale value is relative, it may conveniently be set equal to zero for the preferred conditions, without any loss of generality.

According to the behavior of the scale value in relation to each objective parameter, the resulting expression for s_i is given by

$$s_i \quad -\alpha_i \,|\, x_i \,|^{3/2} \tag{6}$$

where the parameter x_i and the coefficients α_i for global subjects are listed in Table 1.2. Here, P is the sound pressure at the seat, and $[P]_p$ is the most preferred sound pressure that may be assumed at a particular seat position in the room under investigation.

Cocchi, Farina, and Rocco were the first to report that the subjective preference for sound Belds in an existing hall calculated using the four orthogonal physical factors for different source signals agreed with subjective judgments [11]. Sato, Mori, and Ando found that the scale values of subjective preference obtained in changes of source location for a number of listeners at fixed seats in an existing hall agreed with the calculated scale values, when a further parameter, interaural time delay τ_{IACC} was added to the four orthogonal physical factors [12]. The interaural time delay τ_{IACC} relates to the horizontal sound localization and responds to the image shift due to strong reflections or an unbalanced sound field.

Recently, the theory of subjective preference using the four orthogonal physical factors was applied to the acoustic design of the Kirishima International Concert Hall in Kyushu [23]. During the design process, the four orthogonal physical factors at each seat in the concert hall were calculated for several different designs of the hall. After construction of the concert hall, acoustic measurements were performed. To enhance individual satisfaction, the seat selection system testing individual subjective preference of sound fields was introduced.

Table 1.1. Examinations on independent effects of each two of four objective factors on the subjective preference judgements. *Effects of Δt_1 ware examined under conditions of greatly different fixed SL.

Factors	LL	Δt_1	T_{sub}	IACC
LL	- - -	Ando & Okada	None	Ando & Morioka [17]
Δt_1(*SD*)		- - -	Ando, Okura & Yuasa [14]	Ando & Imamura [18]; Ando & Gottlob [19]
T_{sub}			- - -	Ando, Otera & Hamana [20]

Table 1.2. Objective parameters and coefficients (global subjects).

i	x_i	α_i	
		$x_i >= 0$	$x_i < 0$
1	$20 \log \dfrac{P}{[P]_p}$ [dB]	0.07	0.04
2	$\log \dfrac{\Delta t_1}{[\Delta t_1]_p}$	1.42	1.11
3	$\log \dfrac{T_{sub}}{[T_{sub}]_p}$	0.45+0.74A	2.36-0.42A
4	Interaural crosscorrelation (IACC)	1.45	—

Physiological responses to sound fields

It is quite natural to assume that subjective preference is reflected by brain activity or physiological responses.

The relationship between the slow vertex response (SVR) and subjective preference has been investigated systematically [24]. The SVR was recorded by averaging the evoked potentials responding to auditory stimuli, such as clicks, noise and speech. An adjustable test stimulus was presented alternately with a reference stimulus. The pair of stimuli was presented 50 times to integrate and average the evoked potentials, and the SVRs were obtained from the left and right temporal area (T_3, T_4: according to the International 10-20 system [25]). The results show that the latency of N_2-components, that is, the interval between the time the stimulus was presented and the time of the second negative peak of the SVR, corresponded significantly to the subjective preference for changes of the sensation level SL, the delay time of single reflection Δt_1, and the IACC, respectively [24, 26, 27]. The longest latencies are always observed for the most preferred condition, revealing that most of the brain is relaxed under the preferred condition. Furthermore, it is remarkable that hemispheric dominance appeared in the amplitude of the early stage of the SVR. In the results the amplitude of $A(P_1 - N_1)$, Which is the amplitude of the first positive peak to the first negative peak, shows that the hemispheric dominance differed as acoustic factors changed. The left hemisphere was dominant when the Δt_1 was varied and the right hemisphere was dominant when SL or IACC were varied.

The evoked-potential methods cannot be applied to changes of the reverberation time with signals longer than 0.9s, therefore a method for analyzing a continuous brain wave was developed. When a pair of stimuli are presented, the continuous brain wave can be recorded. The effective duration of ACF, τ_e, of the α-waves for the continuous brain wave was analyzed for changes in the delay time of the single reflection and the reverberation time, respectively. It is noteworthy that the τ_e of α-waves are longer only in the left hemisphere for the preferred conditions $[\Delta t_1]_p$ and $[T_{sub}]_p$ [28, 29]. This may be interpreted as being caused by a similar repetitive feature in the α-waves evoking comfortable relaxation repeatedly in the mind.

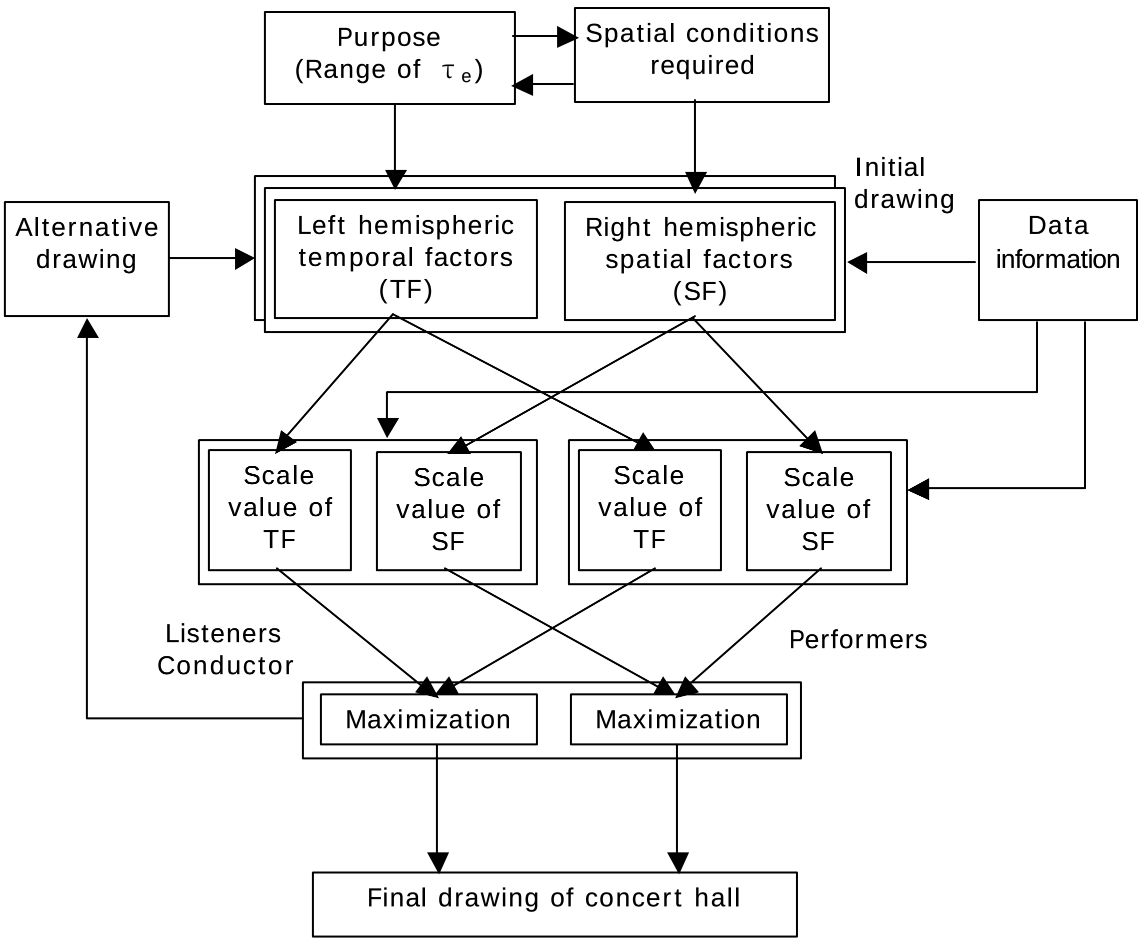
Thus, the subjective preference can be traced back to a imitive response seen as gross brain activity that corresponds well with the scale value of subjective preference. Also, the evidence indicates that the left hemisphere dominance of the temporal factors (Δt_1, T_{sub}) and the right hemisphere dominance of the spatial factors (IACC and SL) may independently influence subjective preference values [30].

Design process using the temporal and spatial factors

Design procedure

The fundamental concept for the acoustic design of a concert hall was derived from the above theory and is illustrated in Figure 1.1. The specialization of the left and right hemispheres for temporal and spatial factors should be taken into consideration for both listeners and performers. Alternative drawings, for increasing the scale values of preference, should be determined using the data information. The first step is to determine the dominant use of the concert hall under design by selecting a certain range of the τ_e for the source programs, which depends on the type of music and its tempo. The second step is to form the initial drawings of the enclosure so as to optimize the spatial factor IACC. The final goal is to maximize the scale values of preference for both the listeners and the performers, and this is reflected in the final drawing of the concert hall.

Figure 1.1. Flow chart for the design of concert hall



1.2. Purpose of This Thesis

The science technology calculation system which is the basis of measurement and psychological evaluation of the environmental noise, including sound fields in a room, is indispensable to this fields' study, so its realization have been hoped. Our study tried to make the computational systems for sound fields, as tools in design and diagnosis based on the model of auditory-brain system, and used it for the concert hall actually.

Chapter 2

Computational System of Simulating the Sound Field in a Room

2.1. Introduction

This paper describes a procedure for the acoustic design of a concert hall that applies the theory of subjective preference, which is associated with the human cerebral hemispheres. Four orthogonal physical factors can be calculated using architectural drawings.

2.2. Outline of Simulating the Sound Field in a Room

The testing room is divided into a sound field simulation room (Figure 2.1) and a control room (Figure 2.2), which are partitioned by a double soundproof door. The interior of the room is decorated and illuminated so that the user (the subject who is being tested) does not feel locked inside a special room, surprised at being in a soundless state that is not ordinarily experienced. In the control room, whose functions are described below, three workstations and a sound field synthesizer (Figure 2.3) are installed on a free-access floor. The test listening method is a multi-speaker method that allows up to four people to listen simultaneously. The speakers are arranged to emphasize the precise reproduction of the direction of reflected sound. The entire surface of the speakers is covered with glass wool to a thickness of 50 mm, minimizing the reflection of medium- and high-frequency sound.

Figure 2.1. The sound simulation room of the Miyama Conceru. Reproduced with kind permission of Mr Suiyo Sato, Suikoh-sya, Tokyo, Japan.

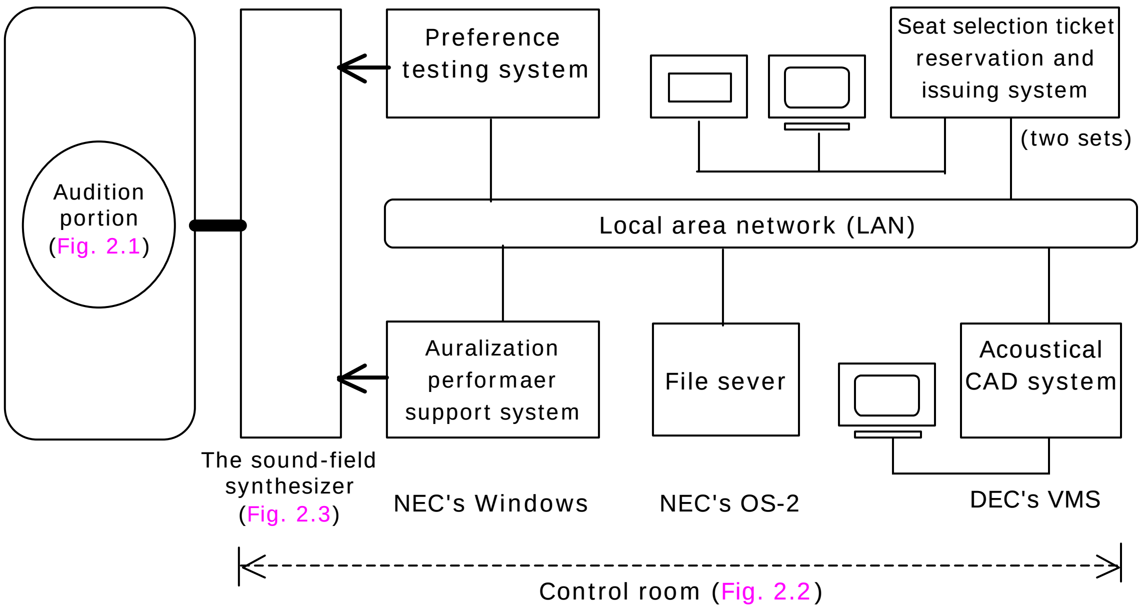
Figure 2.2. The control room for sound simulation etc. Reproduced with kind permission of Mr Suiyo Sato, Suikoh-sya, Tokyo, Japan.

Figure 2.3. The hardware for the sound simulation. Reproduced with kind permission of Mr Suiyo Sato, Suikoh-sya, Tokyo, Japan.

Composition of the overall software system

The system shown in Figure 2.4 is a computer system based on the client-server model and operate under three different operating systems, namely, DEC VMS, NEC OS-2, and Windows. The acoustical CAD system supports everything from the input of architectural drawings to the acoustical design of the interior of the room by the sound ray tracing method and mirror image method, as well as its visual presentation. The part of the routine of the effective wall is shown in Appendix A. A file server allows the sound field auralization system [47, 48] to use the results of the acoustical CAD system. The auralization system also feeds the results to the devices for sound field synthesis and equalization. The seat reservation and ticket issuing system, as an independent subsystem on the LAN, has the functions of event control, testing for sound preferences, issuance of individual cards, seat selection, ticket reservation, and ticket issuance. These functions can run simultaneously from multiple (at present, two) workstations on the LAN.

Figure 2.4. Composition of the overall software and hardware system.



2.3. An Application in Designing of the Kirishima International Concert Hall

The physical factors at each seat were calculated at the design stage with architectural drawings using the image method [31]. In the computer mode, the effect of diffusers on the walls at high frequencies, reflections and scattering caused by seats were not taken into account in the calculation. A single source was considered, located at 3 m from the stage front, 2m to the side of the centre line, and 1.5 m above the stage Boor. The direct sound and reflections up to third order, which included about 300 reflections, with their amplitudes, delay times, and directions of arrival at the listeners were calculated. In the calculation of IACC, white noise was selected as the source signal. In this calculation, the reverberation time is assumed to be 1.8 s (500Hz, occupied) throughout the hall.

The calculated results for the physical factors of the listening level, the initial time delay gap, the IACC, and the total amplitude of reflections (A-Value) are shown in Figure 2.5. The value for the listening level has a range of ±4.0 dB through b out the main audience moor. The largest initial time delay Δt_1 (27 ms) is observed near the source on the center line of the hall. The IACC at each seat is less than 0.2, except for seats close to the source.

Scale values of subjective preference for listeners are shown in Figure 2.6. For slow tempo music (Music motif A), positioning the source near to the stage front increases the total scale value in seats near the back wall of main audience moor. For fast tempo music (Music motif B), positioning the source near the back wall of the stage increases the total scale value in seats at the center of the main audience floor. As far as the type of music is concerned, the fast tempo music may be best suited for this hall.

Preference for performers is not discussed here, but it has been obtained for changes in the delay time of a single reflection [32].

Figure 2.5. Results of calculated values of physical factors: (a) Contour lines of equal relative listening level [dB]; (b) Contour lines of initial time delay gap between the direct sound and first reflection excluding the floor reflection [ms] ; (c) Contour lines of equal IACC for white noise; (d) Contour lines of equal A-value.

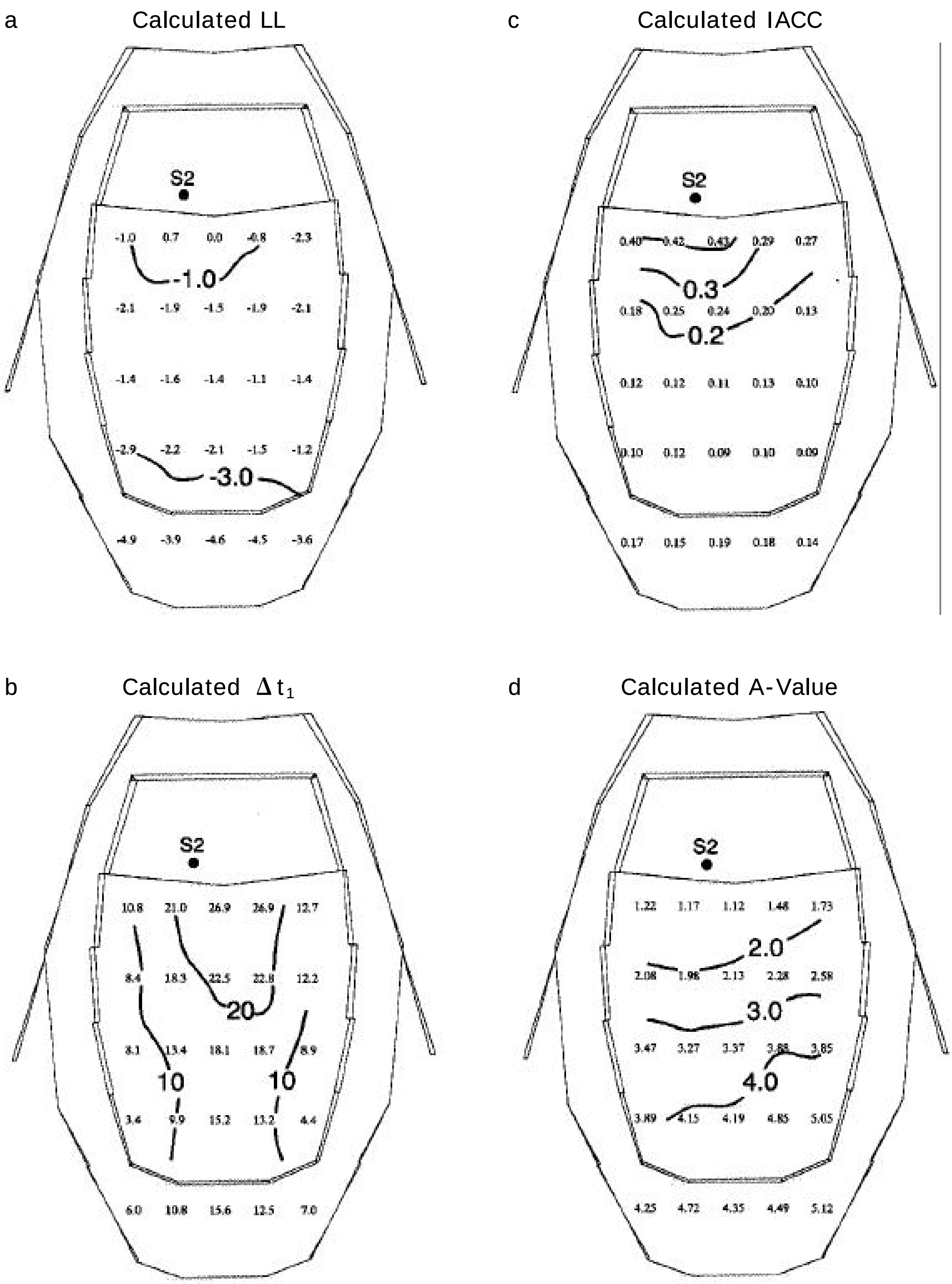
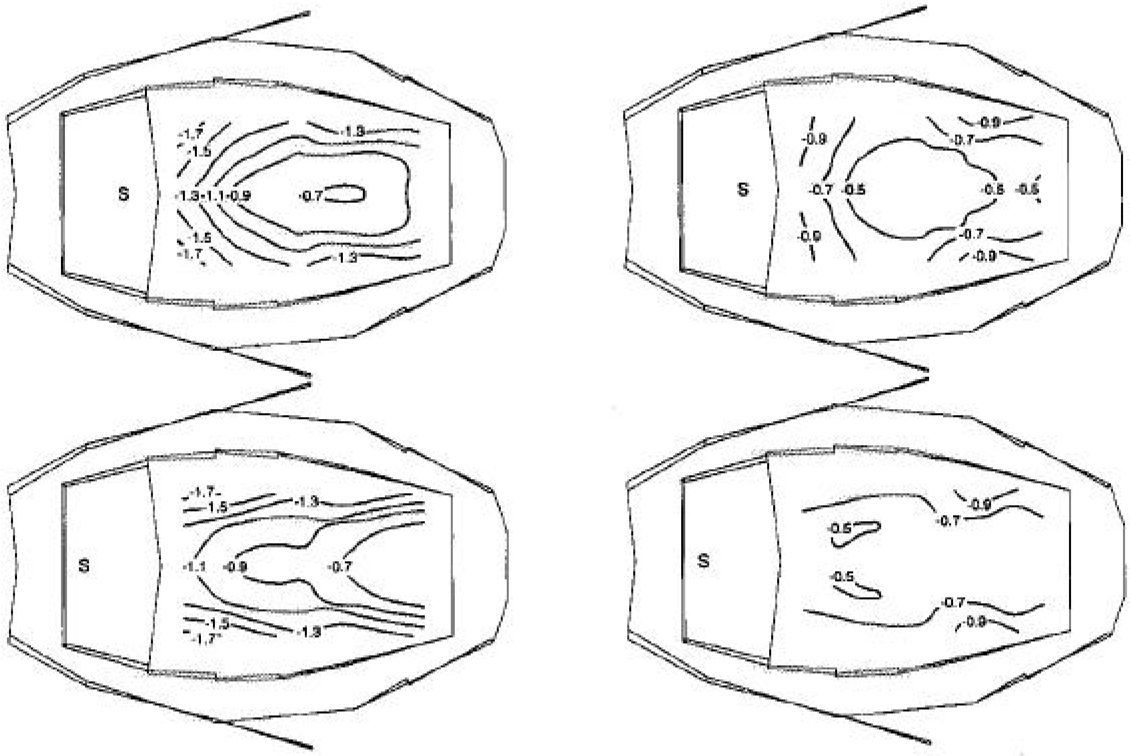


Figure 2.6. Results of calculated scale values of subjective preference for two source positions: (a) Music motif A; (b) Music Motif B.



2.4. Remarks

Based on previous study, sound fields ware simulated by the concert hall building plan in the designing stage. After obtaining the binaural impulse responses at each seat, four orthogonal factors including the sound pressure level (SPL), the initial time delay gap between the direct sound and the first reflection (Δt_1), the subsequent reverberation time (T_{sub}) and the interaural crosscorrelation (IACC) ware analyzed to evaluate subjective preference. The sound fields at each seat in the concert hall can be evaluated before construction.

Chapter 3

Diagnostic system for sound fields in a Room

3.1. Introduction

In order to measure orthogonal factors, SPL, ΔT_1, T_{sub}, IACC, τ_{IACC}, and W_{IACC} [37-42], and also the running ACF of sound field at each seat in a scale model as well as in a real auditorium, a diagnostic system is developed. Based on the model of auditory-brain system which consists of the autocorrelation mechanism, the interaural crosscorrelation mechanism between the both auditory pathways, and the specialization of human cerebral hemispheres as shown in Figure 3.1 [37], a diagnostic system was designed. The system works on PC for Windows with an AD & DA converters, there is non need for special additional devices. After obtaining the binaural impulse responses, four orthogonal factors including the SPL, the initial time delay gap between the direct sound and the first reflection, the subsequent reverberation time and the IACC are analyzed. These factors are used for the calculation of both the scale values of global and individual subjective preferences. In addition to the four factors, two more factors, τ_{IACC} and W_{IACC} as defined in Figure 3.2, extracted from the interaural crosscorrelation function can be figured out for evaluating the image shift of sound source and the apparent source width [42], respectively. Also, the averaged sound energy, $\Phi(0)$, the effective duration, τ_e, defined by the delay at which the envelope of normalized ACF becomes 0.1 (Figure 3.3), and fine structures of autocorrelation function of sound signals including the magnitude of first maximum, ϕ_1, and its delay time, τ_1, of source signals are analyzed. In order to examine effects of reflectors' array above the stage in a 1/10 scale model of auditorium, the IACC measurements are demonstrated here.

Figure 3.1. A model of auditory-brain system.

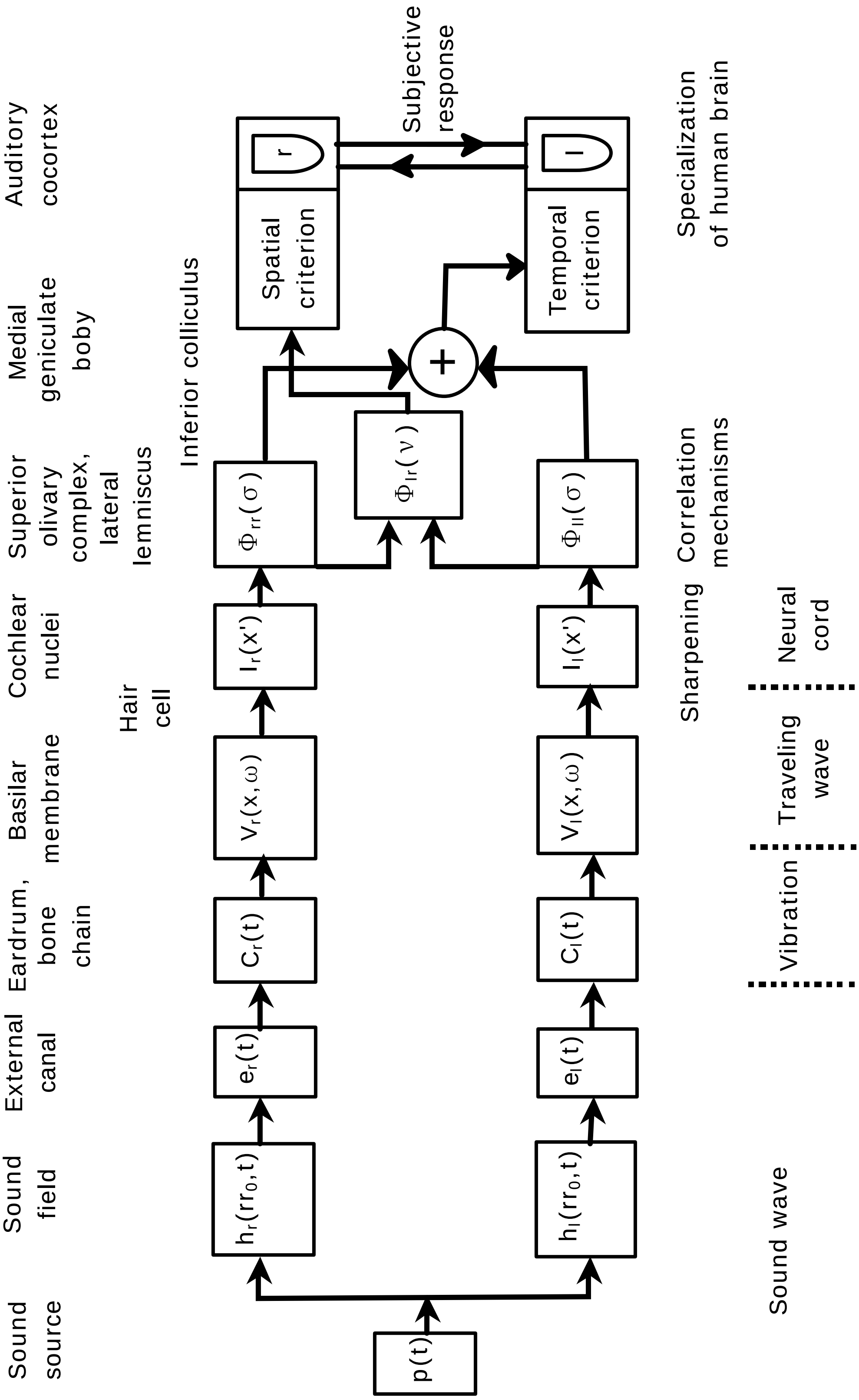


Figure 3.2. Definitions of the IACC, τ_{IACC} and W_{IACC} in the interaural crosscorrelation function.

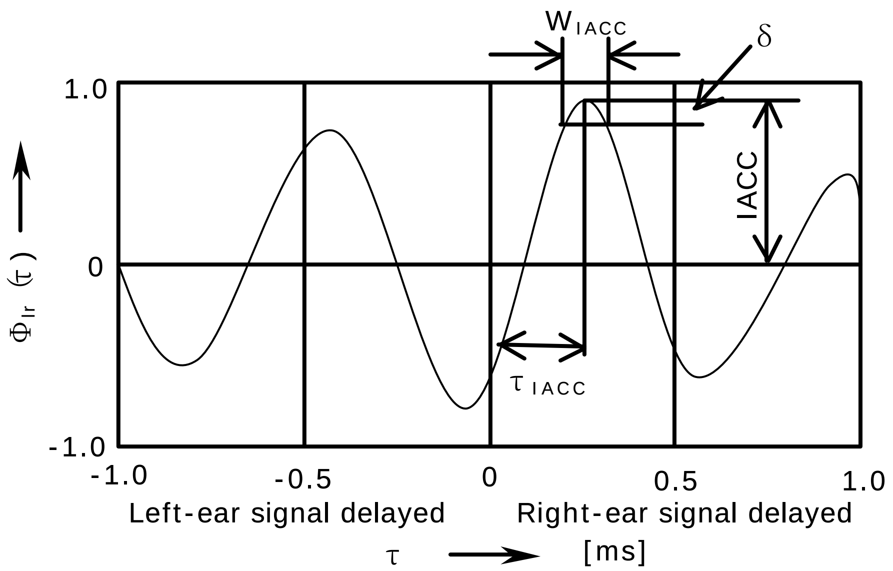
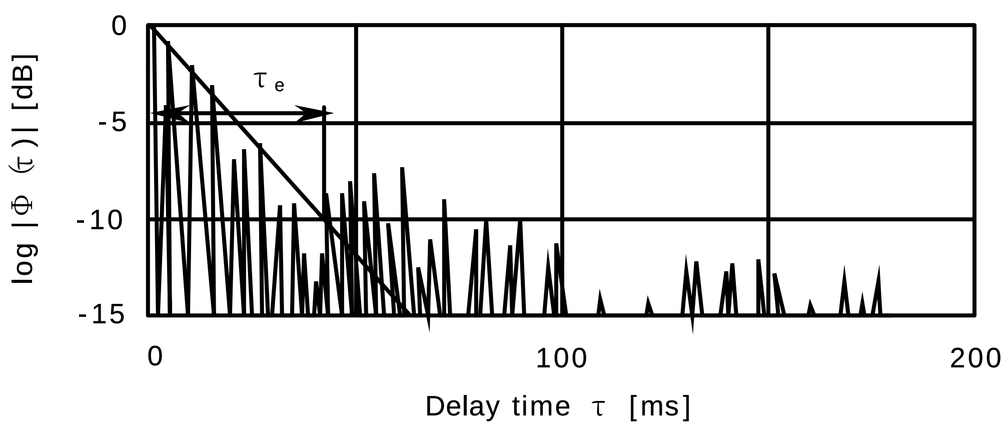


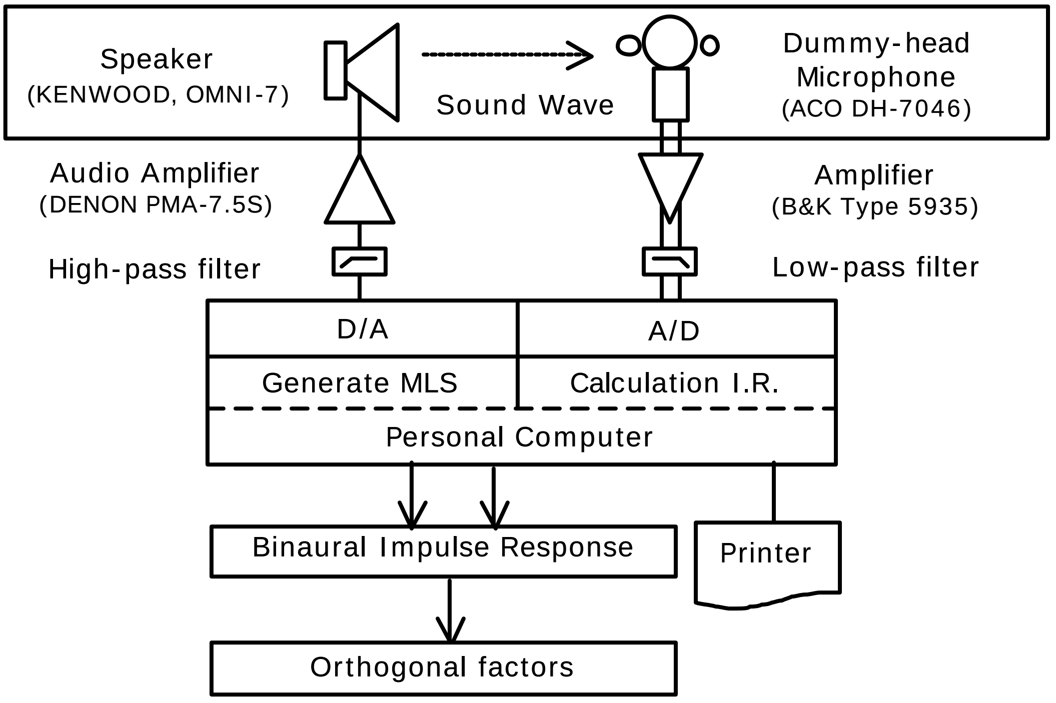
Figure 3.3. A practical example of determining effective duration of ACF defined by the ten-percentile delay, with the straight line-fitting envelope of ACF from 0 to -5 dB.



3.2. Outline of a Diagnostic system

Because the complex requirements made the system difficult to evaluate, an advanced diagnostic system and a high-power computer was used. The measuring system was utilized to obtain the binaural impulse response at each listening position. The sound was created using an omni-directional loudspeaker fed with a maximum-length signal produced by a diagnostic system in a notebook PC. The period of maximum-length signal (MLS) was between 1024 and 524288 samples, and the sampling rate can be changed between 8 kHz and 48 kHz. The acoustic signal amplified from the two microphones placed at the entrances of ears of a 1/10 scale model of dummy head (a sphere with the diameter of 25 mm) was sampled after passing through a low pass filter (Figure 3.4). The binaural-impulse-response measurement may be performed by a summation of the output data from the linea system, without any multiplication operation [43, 44]. The measurement was done automatically within only a few seconds by pushing a single button. This is realized by the logic that the Δt_1 is obtained automatically (Appendix B). It took another few seconds for the analysis of the orthogonal acoustic factors and the scale value of the subjective preference. And at the same time this program can took the result to compute the acoustic parameters and prepare the reports.

Figure 3.4. A block diagram of the measurement system.



3.3. Application of the Diagnostic System (I) for the Kirishima International Concert Hall after Construction

A. Comparison between Simulated and Measured Values of Orthogonal Factors

The Kirishima International Concert Hall contains 770 seats and has a volume of 8,475 m3. In order to increase the number of seats close to the stage, the width of the hall was increased from 17.9 m to 19.7 m (1.8 m) just before construction.

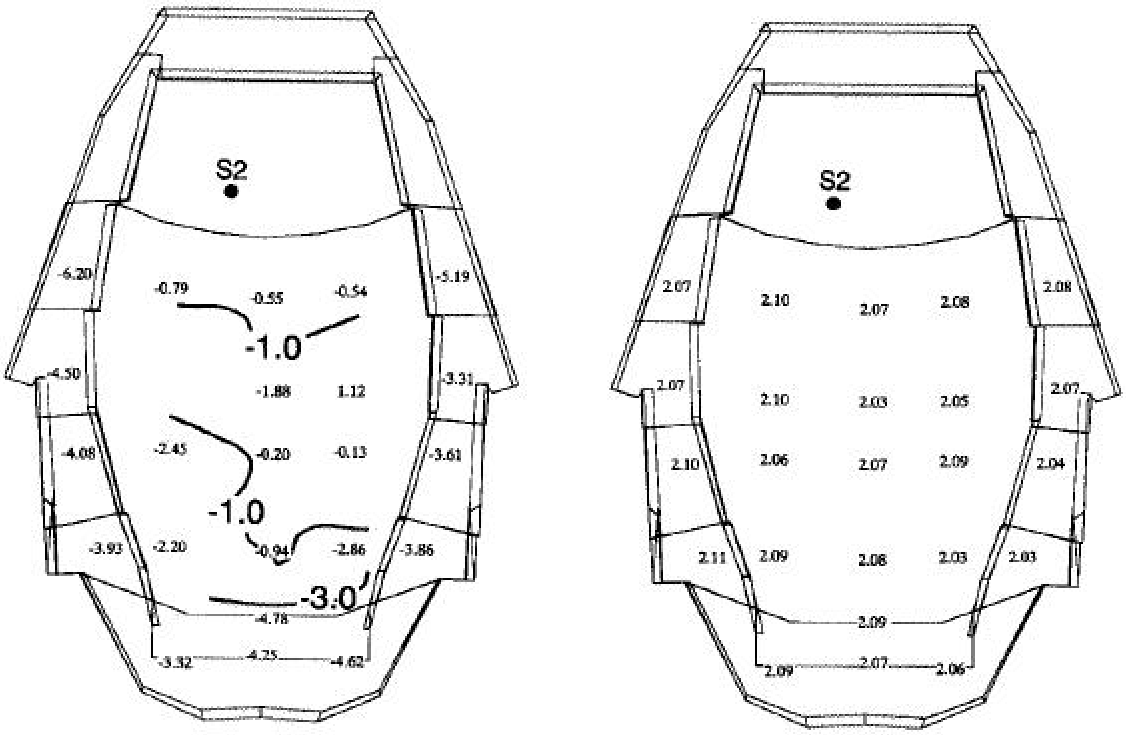
The physical factors at each seat were measured using a maximum length sequence sound signal [33]. The signal was reproduced by an omnidirectional dodecahedron loudspeaker placed 1.5m above the stage floor. The sound source was located either at the center of the stage (S1) or 2 m to the side of the center (S2). Sixty-four seats were selected to test the source location S 1 and 6fty-Gve seats for the source location S2. At each seat, the acoustic signals were recorded by two small microphones at the entrances of the ear canals of a real person. In order to obtain the impulse responses, the signals were analyzed using the Hadamard Transformation. Using the impulse responses at both ears, the physical factors LL, Δt_1 , T_{sub} , IACC, and A-Value were calculated for all octave bands with center frequencies between 125 Hz and 4 kHz.

The measurements of the physical factors (LL and T_{sub} at frequencies of 500Hz, IACC for white noise, Δt_1, and A-Value) for S2 are shown in Figure 3.5. The listening levels indicated are relative to those at the front seat near the centre line. The range of the values is similar to that of the calculated results, that is, 8.5dB for Sl and 6.5 dB for S2. The largest initial time delay is 29ms and is not much different from the calculated value (27ms), in spite of the change in hall width. The reverberation time T_{sub} Obtained by the decay rate between -5 dB and -20 dB is almost constant throughout the hall, and has a range of j=0.05 s (unoccupied). It is estimated to be about 1.75 s when the hall is occupied, which is close to the design goal (1.8 s). The IACC values for frequencies above 500 Hz are less than 0.5 throughout the hall, and the interaural time delay τ_{IACC} for S1 is constant at zero for almost all seats. Generally, the IACC at each seat is smaller for source position S2 than for S 1. With regard to the physical factors of upper level seating, the contour lines between calculated and measured values cannot be compared because there were few measuring points. The differences between calculated values and measured values throughout the seats measured (23 points) are as follows; LL: ±1.0 dB, Δt_1: ±3.0ms and IACC: ±0.05.

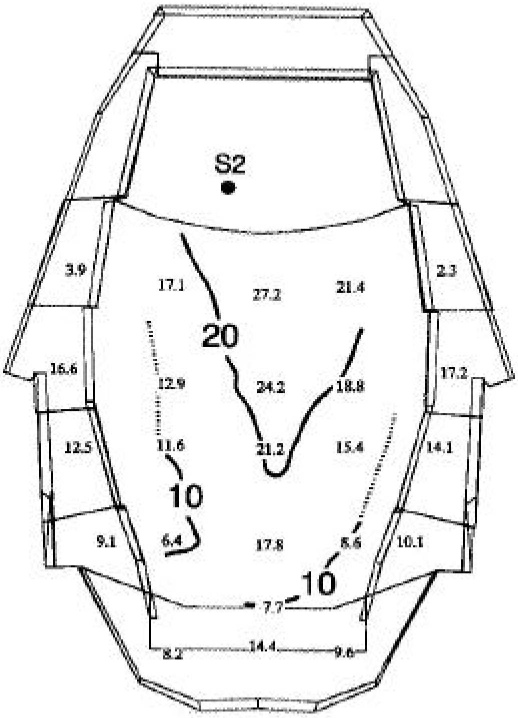
In spite of the change (about 10 percent) in the width of the hall, reasonable agreement between calculated and measured values of the four physical factors was obtained in the design Process.

Figure 3.5. Results of measured values of physical factors for source position S2: (a) Contour lines of equal relative-listening level (500Hz) [dB]; (b) Contour lines of initial time delay gap between the direct sound and first reflection excluding the floor reflection [ms]; (c) Values of reverberation time (500Hz); (d) Contour lines of equal IACC (for white noise) (e) Contour lines of equal A-value.

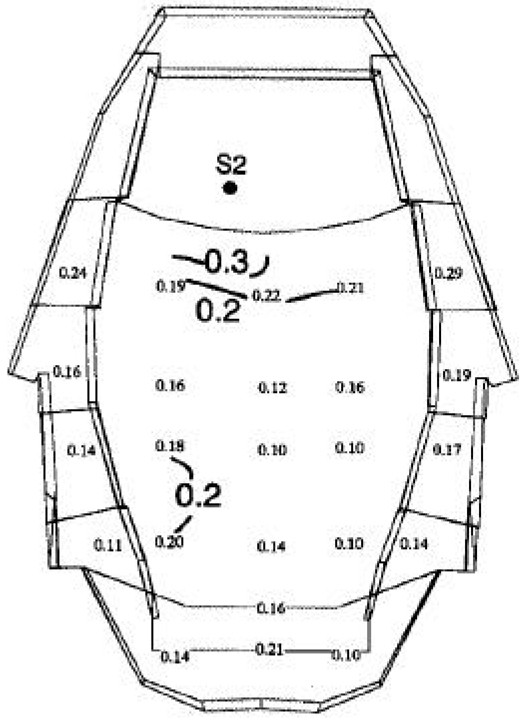
a Measured LL   c Measured T_{sub}



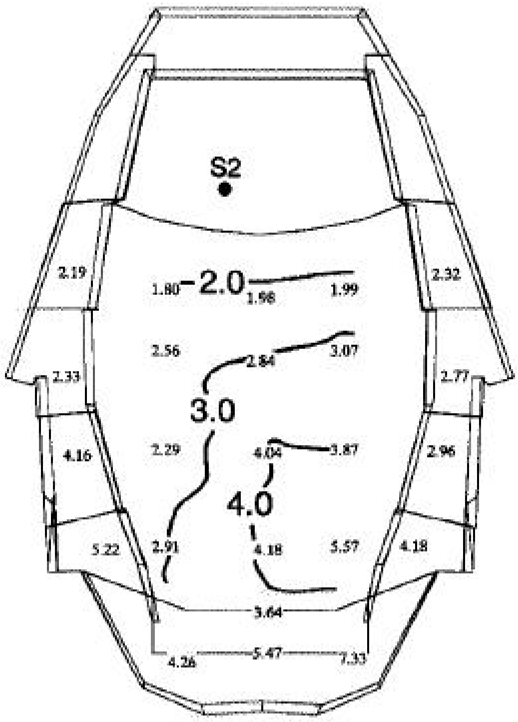
b Measured Δt_1



c Measured IACC



e Measured A-value



B. Remarks

In this study, the system had been developed, which can measure four orthogonal factors including the SPL, the initial time delay gap between the direct sound and the first reflection, the subsequent reverberation time and the IACC, in order to diagnose sound fields of each seat in the concert hall at its completion based on the model of auditory-brain system.

The usefulness has been revealed by comparison between the orthogonal factors obtained by actual measurement at Kirishima concert hall's completion and the orthogonal factors obtained by simulation at the planning stage.

3.4. Application of the Diagnostic System (II) for the 1/10-Scale Model of the Tsuyama Music Cultural Hall

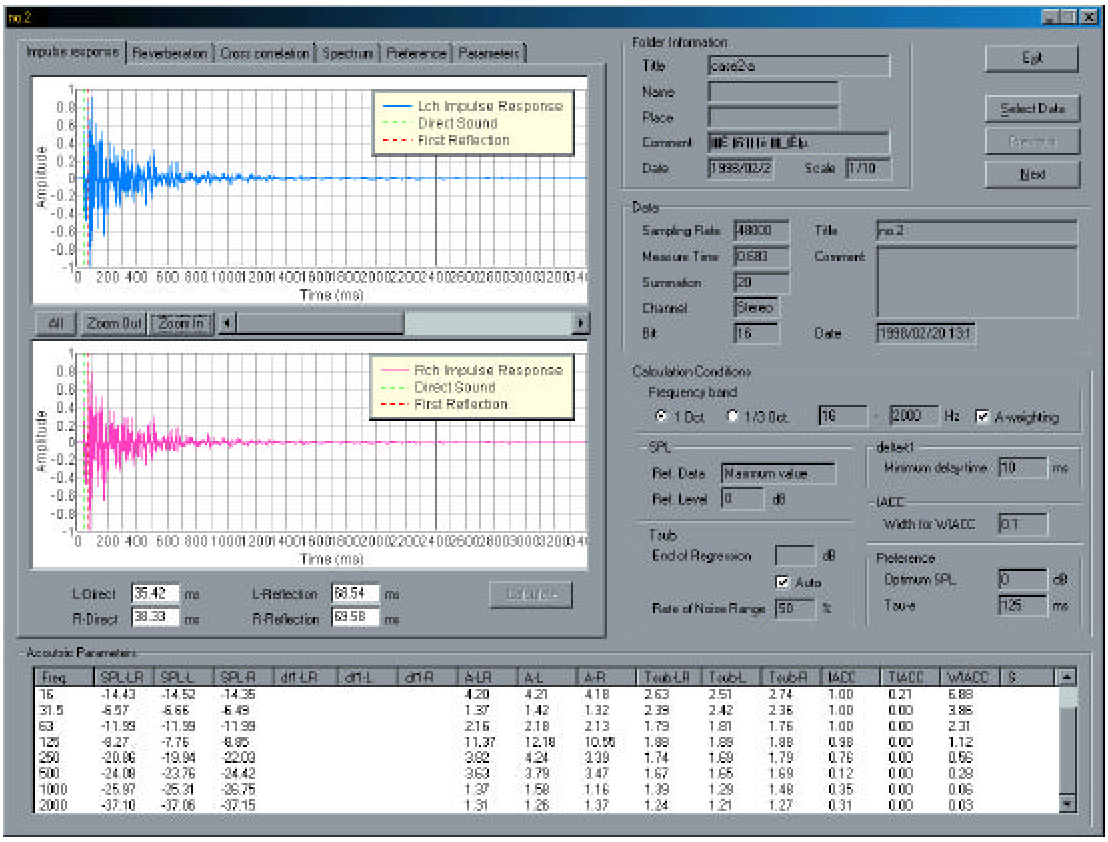
A. Effects of Reflectors Above the Stage

(1) Procedure

The diagnostic system developed may examine effects of scattered reflections of complex boundary conditions of the room. The reflectors above the stage are designed mainly for the performer obtaining the preferred reflections according to the program sources. We measured the IACC of sound field at each seat to find effects of reflectors' array above the stage [46]. The effective direction of reflections to listeners for the 2000 Hz range is centered on +- 18 degrees from the median plane, which might be realized by the reflectors above the stage [37]. Therefore, the IACC of the 2000 Hz-frequency band is selected here to be examined.

In order to obtain reliable results, measurements were repeated several times until obtaining the same results of the binaural impulse responses. The sound is produced using an omni-directional loudspeaker fed with the MLS produced by the diagnostic system in a notebook PC. Figure 3.6 shows the window on PC of actual diagnostic system with the data obtained from the impulse responses. In the measurement, a special attention should be made to maintain a suitable value of the signal to noise ratio adjusting the power level of the loudspeaker.

Figure 3.6. An example of display window of the diagnostic system, with binaural impulse responses.



(2) Results

As mentioned above, in order to examine effects of reflectors' array on the IACC for the 2000 Hz range at 15 seating positions shown in Figure 3.7, measurements were performed with and without reflectors above the stage. As indicated in this figure, the location of the sound source is marked by a star, and triangular reflectors` arrays [46] are installed above the stage. Figure 3.8 shows the measured results of the IACC without reflectors, and Figure 3.7 shows those with reflectors` array. The IACC values of the 2 kHz-frequency band for a real room were measured at the 20 kHz frequency band in the 1/10 scale model.

As shown in these figures, the reflectors decrease results of the IACC at the 9 measuring points, so that acoustic quality is much improved. Especially, the decrement of IACC values was remarkable in the frontal area close to the stage in audience floor except for the center, due to the reflections from above the stage to the listeners.

Figure 3.7 Measured IACC for the 2000 Hz-frequency band with reflectors above the stage.

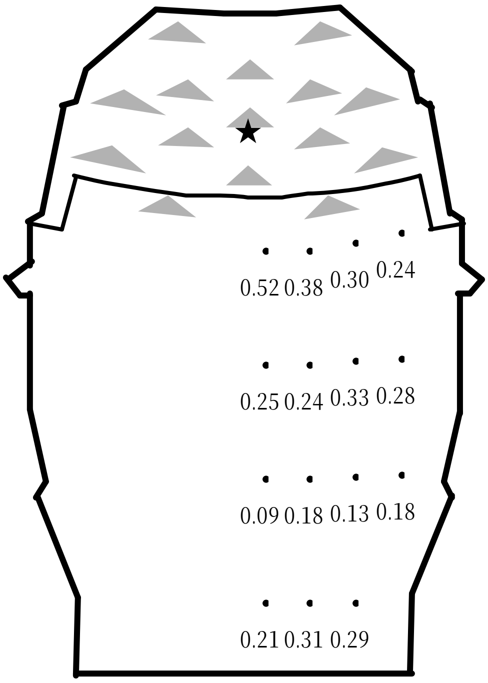
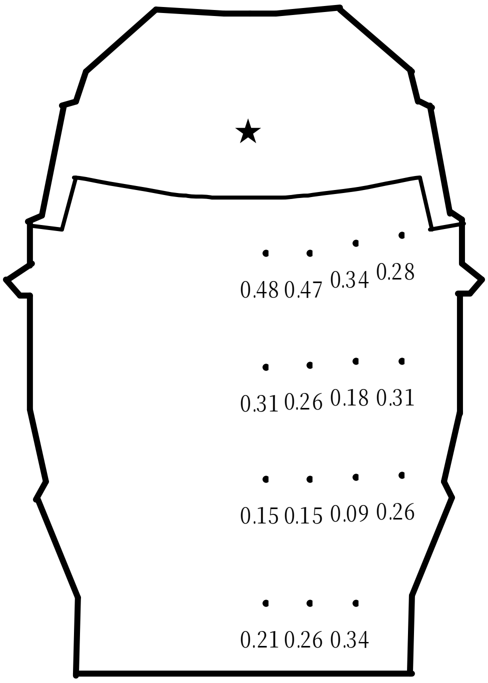


Figure 3.8. Measured IACC for the 2000 Hz-frequency band without reflectors above the stage.



B. Remarks

It is shown that measurements in the 1/10 scale model for acoustic parameters by the diagnostic system may prove effects of reflectors' array and other scattering elements which may not available by calculation at the design stage. In order to examine the sound fields after the construction of the auditorium, the diagnostic system measuring orthogonal factors may be applied. Also, keeping the subjectively optimal conditions, this system may be applied for the automatic control of sound fields by the use of electro-acoustic systems.

Chapter 4

A Seat Selection System

4.1. Introduction

As a service to those who attend the concert hall, Miyama Conceru (Kirishima Concert Hall) provides a system for dispensing tickets for seats that match the audio preferences of the patron (the seat reservation and ticket issuing system). This system is implemented by acoustical devices and computers installed in a testing room. In addition to this function, an acoustical CAD system, a performer support system and other software systems for managing the hall, including event control, customer control, booking and renting out of facilities, are also running as subsystems on a local area network (LAN).

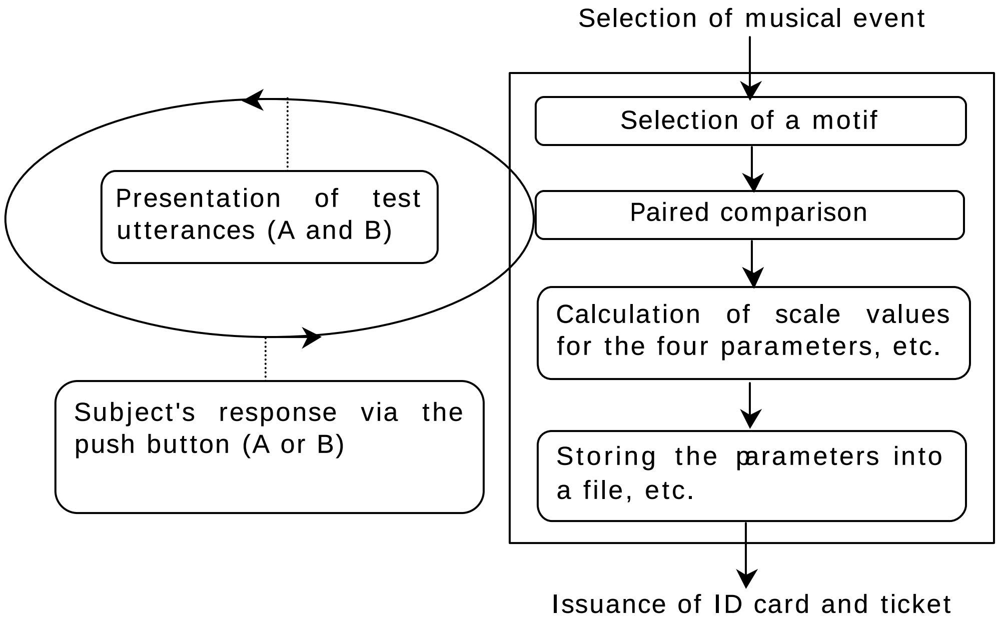
4.2. A Listening Room for Testing the Individual Subjective Preference

Preference testing system

Listeners have preferences concerning sounds, and it is known that these preferences can be described by four independent acoustic parameters: the listening level, LL, the initial delay gap' Δt_1 , the reverberation time, T_{sub}, and the interaural cross correlation, IACC. [49]

Figure 4.1 shows a block diagram of the preference test. After selecting the musical motif, those parameters related to the motif are fixed. An individua1's most preferred values, $[LL]_p$, $[\Delta t_1]_P$, $[T_{sub}]_P$, can be measured by the paired comparison test. For IACC, a smaller value for this parameter is always preferable. [50] The acoustical CAD system can calculate the above four parameters individually for all the seats in the hall. Then the seat that an individual would find the most agreeable is determined from the values of the test results of his preferences for each parameter and the predicted values at the seats.

Figure 4.1. Block diagram of the preference test, Photo reproduced with kind permission of Mr Suiyo Sato, Suikoh-sha, Tokyo, Japan.



Subjects in the listening room

(1) Motifs

Twelve musical motifs were chosen for the compositions expected to be performed more frequently at Miyama Conceru, including short string pieces, wind pieces, piano solos and violin solos. For all of these, dry source recordings were used, with permission, from commercially available CDs. Table 4.1 shows the measured parameters for 12 motifs.

Table 4.1. Music used and its parameters for the preference test of sound fields

Motif	Name of motif	Composer	Source	Track	T_s	T_d	τ_e	$[\Delta t_1]_p$	$[\Delta T_{sub}]_p$
1	Royal Pavane	Gibbons	1	1	0.0	7.9	127	30	1.75
2	Sinfonietta, Opus 48, IV	M.Arnold	1	2	0.0	6.3	54	21	1.2
3	Piano, Classical Mood	-	2	1	135.0	7.6	119	23	1.4
4	Female Chorus Standard	-	2	15	0.0	7.2	33	26	0.8
5	'Water Music' Suite VI	G.F. Handel	3	24	0.0	7.9	62	25	1.5
6	Jazz, Ensemble	-	3	35	7.0	6.0	49	38	1.1
7	Marriage of Figaro Overture	W.A. Mozart	3	25	7.0	8.4	55	22	1.2
8	Flute, Classical Mood	-	3	31	0.0	8.1	40	32	1.0
9	Cello, Solo	-	2	10	199.0	8.0	234	23	1.4
10	Violin, Solo	-	3	30	0.0	7.1	114	22	1.3
11	Clarinet, Solo	-	2	11	0.0	7.2	65	26	1.5
12	Trumpet, Solo	-	2	8	61.0	8.6	35	28	1.6

Source 1: The original tape was made by Gottingen University.
Source 2: Japan Audio Society (CD-3), 'Impact 2'.
Source 3: DENON, Vol.2 (DENON COC0-75085).
T_s (s): The elaspsed time from the beginning of the specified track to the starting time of the motif.
T_d (s): The duration time of the motif.
τ_e (ms): The effective duration time of the autocorrelation function at which the envelope of the normalized autocorrelation function (ACF) becomes just 0.1.
$[\Delta t_1]_p$ (ms).
$[\Delta T_{sub}]_p$ (s).

(2) Paired comparison test

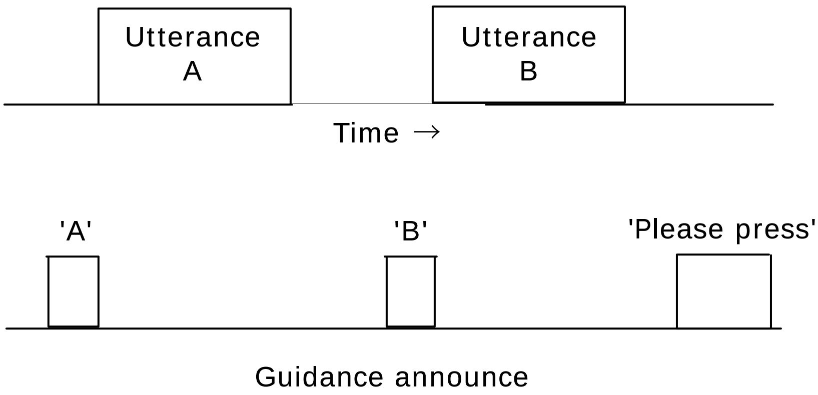
For the listening level, the peak of the motif signal was fixed at the five levels 70, 75, 80, 85, 90 dBA (time constant 100 ms, peak) and the interaural cross correlation was set at the three levels 0.4, 0.75 and 1.0. For the initial delay gap and reverberation time, it is known from previous studies and experiments that the tested subject's overall average $[\Delta t_1]_P$, $[T_{sub}]_P$, can be inferred from the effective duration time τ_e of the autocorrelation of the musical motif signal. The parameter Te refers to the time at which the envelope of the normalized autocorrelation function (ACF) becomes just 0.1 of its maximum. The time duration of the total signal used for obtaining the ACF was 2 seconds. In the actual preference test, the time delay and T_{sub} values are set around these inferred values. The value of the level of the four parameters for motif 1, for example, is as listed in Table 4.2.

A comparison pair is presented by the time schedule shown in Figure 4.2. Since it is assumed here that there is no order effect of the presentation, the number of test signals totals 33 pairs, including 10 each for LL, All, T_{sub} and three for IACC. The tested subject listens to a pair of test signals, then responds with a switch indicating which alternative he prefers. The subject is told when to respond by an audio instruction saying 'Please press'. It takes about 15 minutes to test for the 33 pairs.

Table 4.2. The levels of the parameters for the motif 1, 'Royal Pavane'

LL (dBA)	70	75	80	85	90
Δt_1 (ms)	5	10	21	42	84
T_{sub} (s)	0.3	0.6	1.2	2.4	4.8
IACC	0.4	0.75	1.0		

Figure 4.2. Sample of a paired test utterance.



4.3. Outline of the Preference Tests, Seat Selection and Reservation Systems

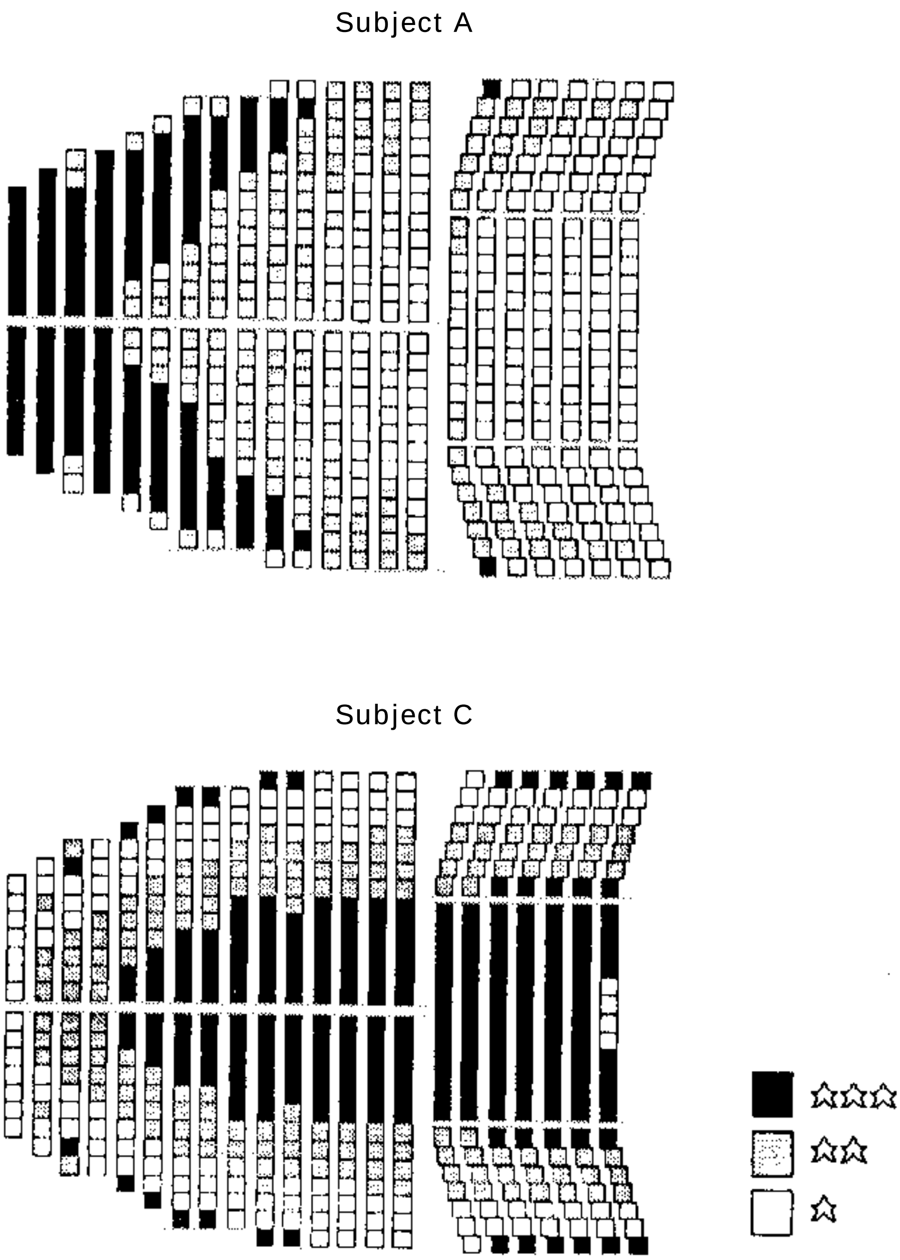
Reservation and ticket issuing system

When a person who has taken a preference test wishes to buy tickets for a concert, he hands over his individual ID card to a ticket-issuing operator, and then a view of the concert hall seats is displayed on a computer screen, classified into three grades according to the concert-goer's preferences (Figure 4.3).

The results of the display are printed out, allowing the patron to inspect the results and either choose his seat then, or to reserve a seat later by telephone. Initially the preference grades are classified into three levels (Figure 4.4). The advantage of this reservation system is that it allows each user to enjoy music with the assurance that he is hearing the concert from 'his' seat.

Figure 4.3. Ticket issuing by using customer's ID card. Reproduced with kind permission of Mr Suiyo Sato, Suikoh-sha, Tokyo, Japan.

Figure 4.4. The preference grades classified into three levels.



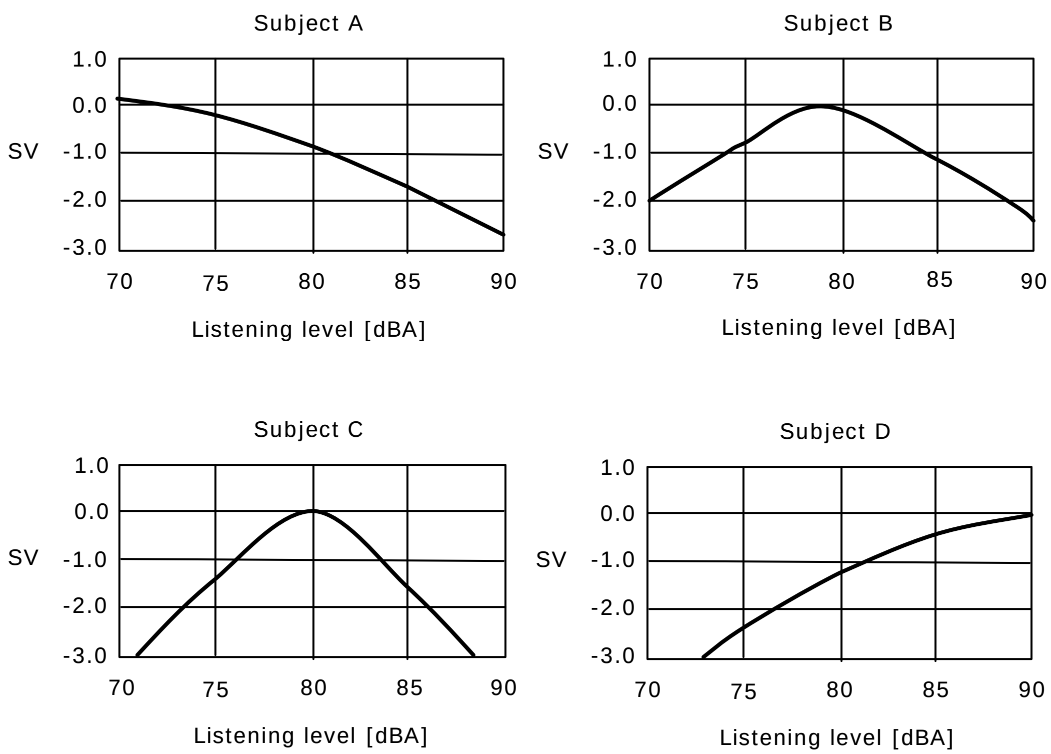
4.4. Examples of Individuals Subjective Preference by the Paired-Comparison Tests

Examples of the concert-goer's preferences

Figure 4.5 shows samples for measured preference curves of four subjects for the listening level parameter. This shows that the peak and curvature of the preference curves are individually different. This indicates that people have strong preferences concerning sound fields.

These results are stored into a file on the database for concert-goers. Then an individual ID card is issued.

Figure 4.5. Measured preference curves of four subjects for listening level.



4.5. Application of subjective preference theory for a Seat selection system

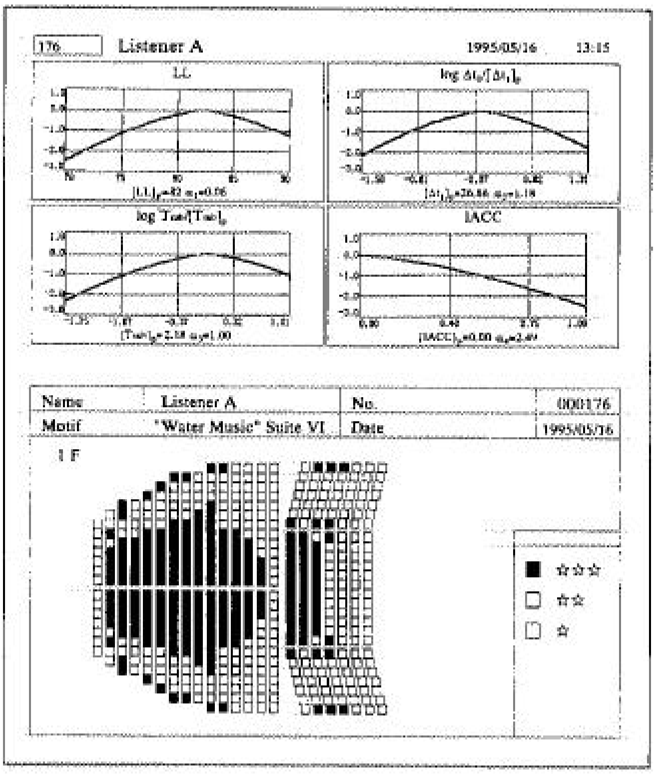
A selection system was introduced to maximize the preference of each individual with respect to the sound field as described by the four acoustic factors. Preference tests were performed in a listening room simulating the sound fields with multiple loudspeakers installed in the Kirishima international concert hall. Suitable methods can be to use either the paired-comparison test or a method of adjustment by the listener, or a combined method that obtains the most preferred combination of LL, Δt_1, and T_{sub}. To examine the preference as a function of the IACC, only the paired-comparison test is applicable, because a smaller value for the IACC is always preferable. Preference tests were performed using several music motifs. Since it is assumed here that there is no effect of the order of presentation, the test sound fields were tested using a total of thirty-three pairs, with five levels of LL, Δt_1, T_{sub}, and three levels of IACC. The duration of each stimulus is about 10s. It takes about 15 minutes for each listener.

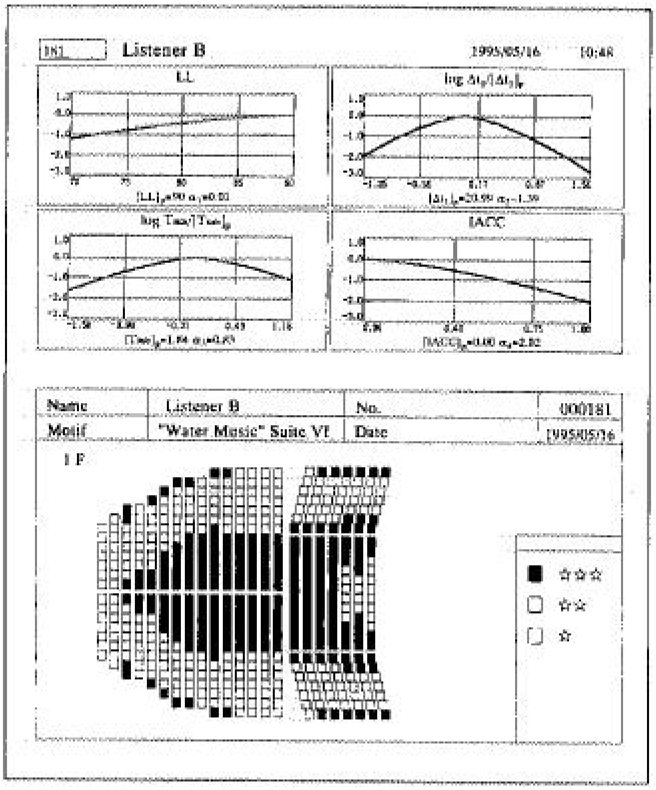
Scale values of individual preference as a function of each physical factor were obtained by the simplified method given in [34]. An area of seats where individual preference is maximized can be found. The large individual differences in the most preferred listening level are at least partly related to the individual hearing level. The preferred initial time delay and the preferred reverberation time are associated with an individual preference for "liveness". Generally, the preferred values of LL, Δt_1, and T_{sub} for each individual are quite different, but all of the subjects tested always preferred a small value of IACC.

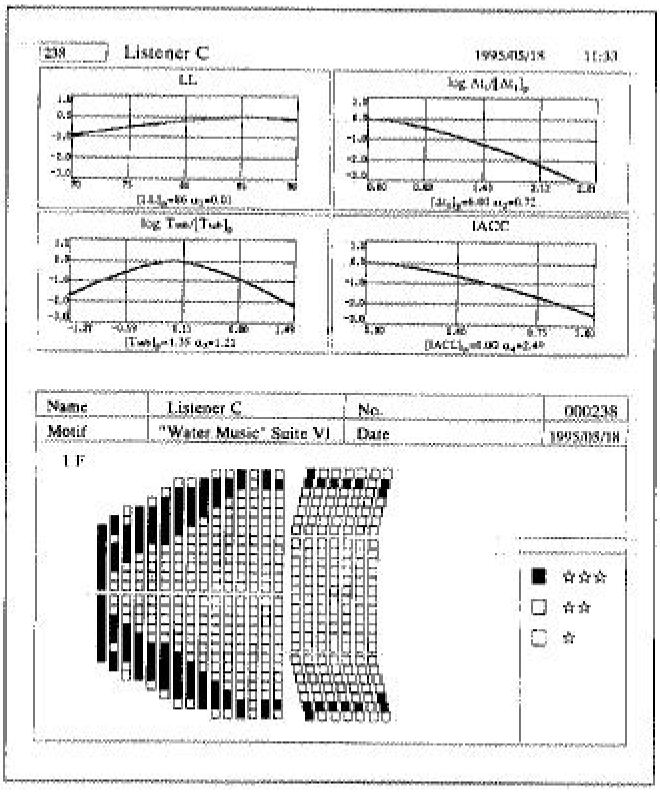
Examples of preference-test results for three individuals using Suite VI from the 'Water Music" by G. F. Handel (τ_e = 62ms) are shown in Figure 4.6. Listener A shows a preference similar to the global preference for each factor. Listener B is recommended to sit close to the stage because he prefers a high listening level. The listening level is designed to be nearly constant throughout the hall; however, a large variation in the listening level may be useful for meeting the large range of individual preferences in listening level. Listener C is recommended to sit near the side walls because he prefers a short initial time delay gap. The preferred value of each physical factor is very different. The range of preferred listening level, for example, is much greater than 20 dB (A) due to the individual difference of the hearing level. With regard to the reverberation time, the range of preferred value is 0.5-4.5 s [35]. The initial time delay gap also has a great range of preferred value. To maximize the individual preference, such facts must be considered.

The results of the preference tests for each individual have been discussed in terms of the inter-individual differences [36]. The investigation shows that subjects with small inter-individual difference have critical preferred value indicating large values of α_i in equation (6), chapter 1. On the other hand, the preferred listening level for the subjects with small values of α_i are barely determined.

Figure 4.6. The results of preference tests and preferred seats graded into three levels: Listener A, B, C.







238
Listener C
1995/05/18 11:33
LL
log Δt1/[Δt1]p
log Tsub/[Tsub]p
IACC
Name
Listener C
No.
000238
Motif
"Water Music" Suite VI
Date
1995/05/18
1 F
☆☆☆
☆☆
☆

4.6. Remarks

The individual subjective preference may be met using the seat-selection system based on the four orthogonal physical factors of the sound field for each seat. Examples of scale values obtained by preference tests and the results of seat selection to maximize individual preference are demonstrated.

Chapter 5

Diagnostic Systems for Environmental Noise

5.1. Introduction

In order to measure orthogonal factors, SPL, Δt_1, T_{sub}, IACC, τ_{IACC}, and W_{IACC} [51-56], and also the running ACF of sound field at each seat in a scale model as well as in a real auditorium, a diagnostic system is developed. Based on the model of auditory-brain system which consists of the autocorrelation mechanism, the interaural crosscorrelation mechanism between the both auditory pathways, and the specialization of human cerebral hemispheres [51], a diagnostic system was developed. The system works on PC for Windows with an AD & DA converters, thus it is no need for special additional devices. After obtaining the binaural impulse responses, four orthogonal factors including the SPL, the initial time delay gap between the direct sound and the first reflection, the subsequent reverberation time and the IACC are analyzed. These factors are used for the calculation of both the scale values of global and individual subjective preferences. In addition to the four factors, two more factors, τ_{IACC} and W_{IACC}, extracted from the interaural crosscorrelation function can be figured out for evaluating the image shift of sound source and the apparent source width [56], respectively. Also, the averaged sound energy for two ears, the effective duration, τ_e, defined by the delay at which the envelope of normalized ACF becomes 0.1, and fine structures of autocorrelation function of sound signals including the magnitude of first maximum, ϕ_1, and its delay time, τ_1, of source signals are analyzed. Also, for the internet based measurement of environmental noise, this system may be utilized for identifying source signals and spatial information.

5.2. Outline of a diagnostic system

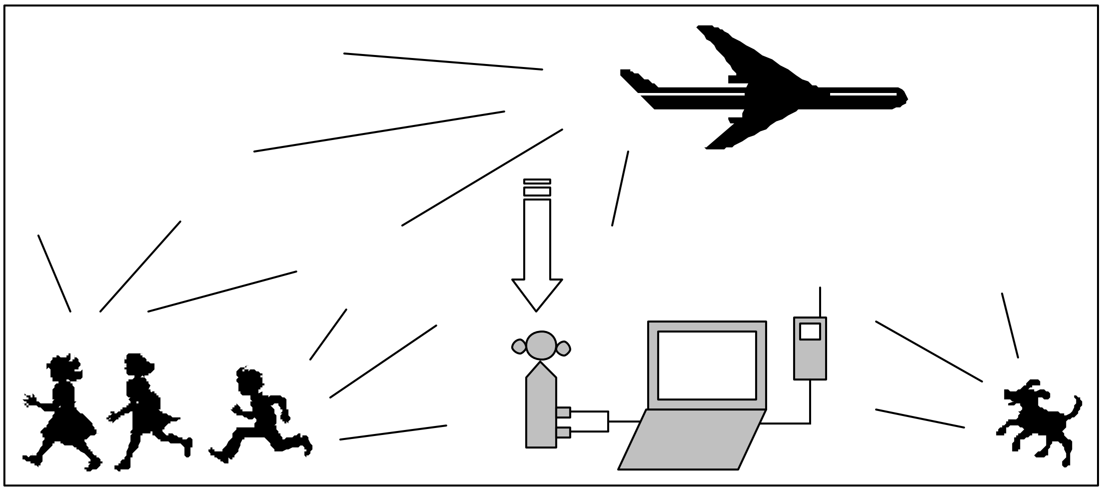
Measurement of environmental noise is illustrated in Figure 5.1. The purpose is measuring an influence which an airport have on the region. This is an example of technical application of the concert hall measurement system [59, 60] for the internet technology.

The measurement is performed by two channels on each measuring point, where the collected data are distinguished the aircraft noise from the others automatically [61, 62].

In case of the aircraft noise, it is analyzed there, and is sent to central office through internet. Primary sensations, pitch, loudness and timbre, of a given source signal and sound field are described based on a model of auditory-brain system [51]. The model consists of both autocorrelation and interaural crosscorreration mechanisms. In order to describe timbre or quality of sound fields, for example, the human cerebral hemisphere specialization for the temporal and spatial factors is taken into consideration as the similar manner to the subjective preference.

As this result, by sending only these parameters, it is not necessary to send whole sound data obtained by measurement. The diagnostic system may be applied to the internet by sampling environmental noise.

Figure 5.1. The internet-oriented system for measuring environmental noise with dummy-head microphones; .The system identifies, for example, the aircraft noise.



5.3. Identification of a noise source

The noise is identified as that of an aircraft, an automobile, a factory and so on, by using four factors extracted from ACF; $\Phi(0)$, τ_e, τ_1, and ϕ_1 (Figure 5.2, 5.3). It is the same method that is used for the speech intelligibility of single syllables [61]. Four factors, $\Phi(0)$, τ_e, τ_1 and ϕ_1 are utilized for the identification. At our test, it has been found that the system can distinguish the engine sound between Porsche and Mercedes Diesel.

A measurement is done automatically according to the condition a user sets. Measured data is analyzed in the background by MS Windows's multi-thread function, and each factor is extracted and the noise source is identified.

Figure 5.2. A practical example of determining effective duration of ACF defined by the ten-percentile delay, with the straight line-fitting envelope of ACF from 0 to -5 dB.

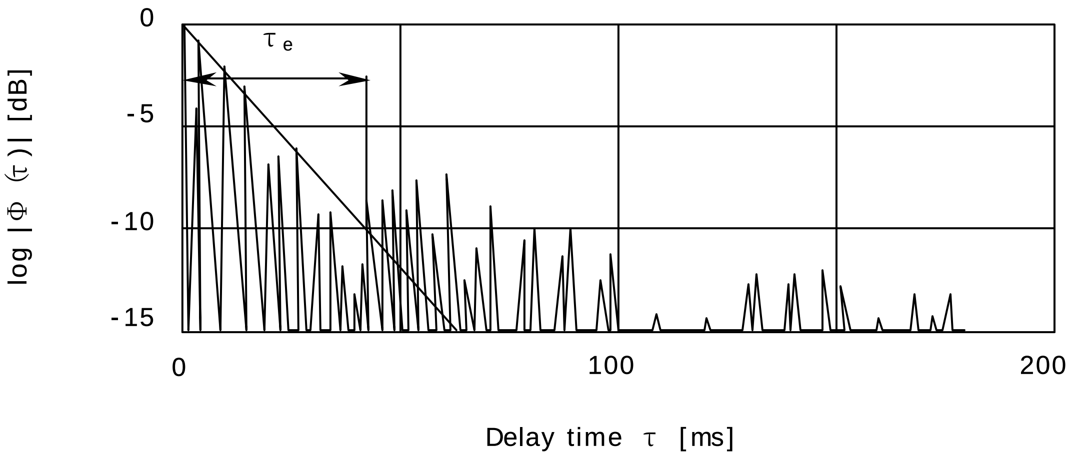
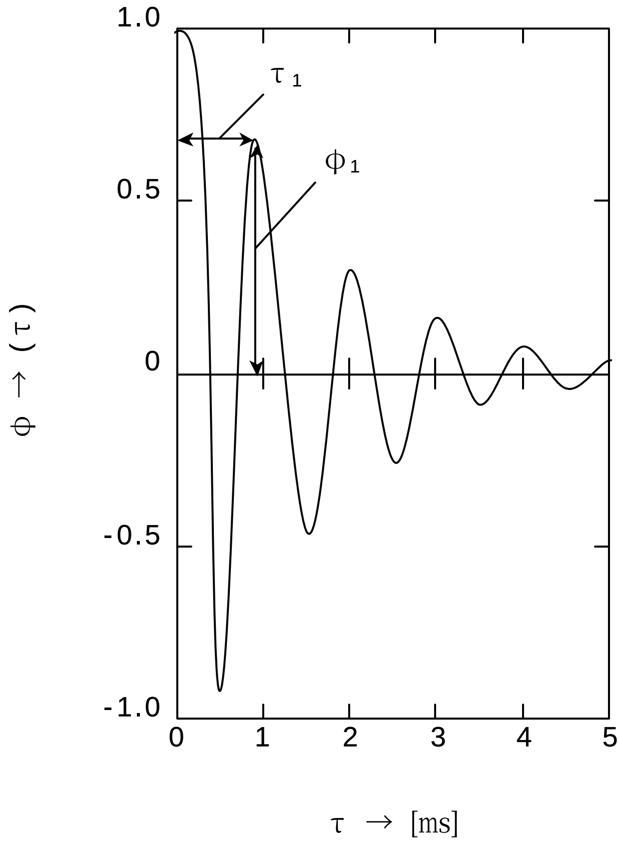


Figure 5.3. Definitions of the ϕ_1 and τ_1 for the autocorrelation function.



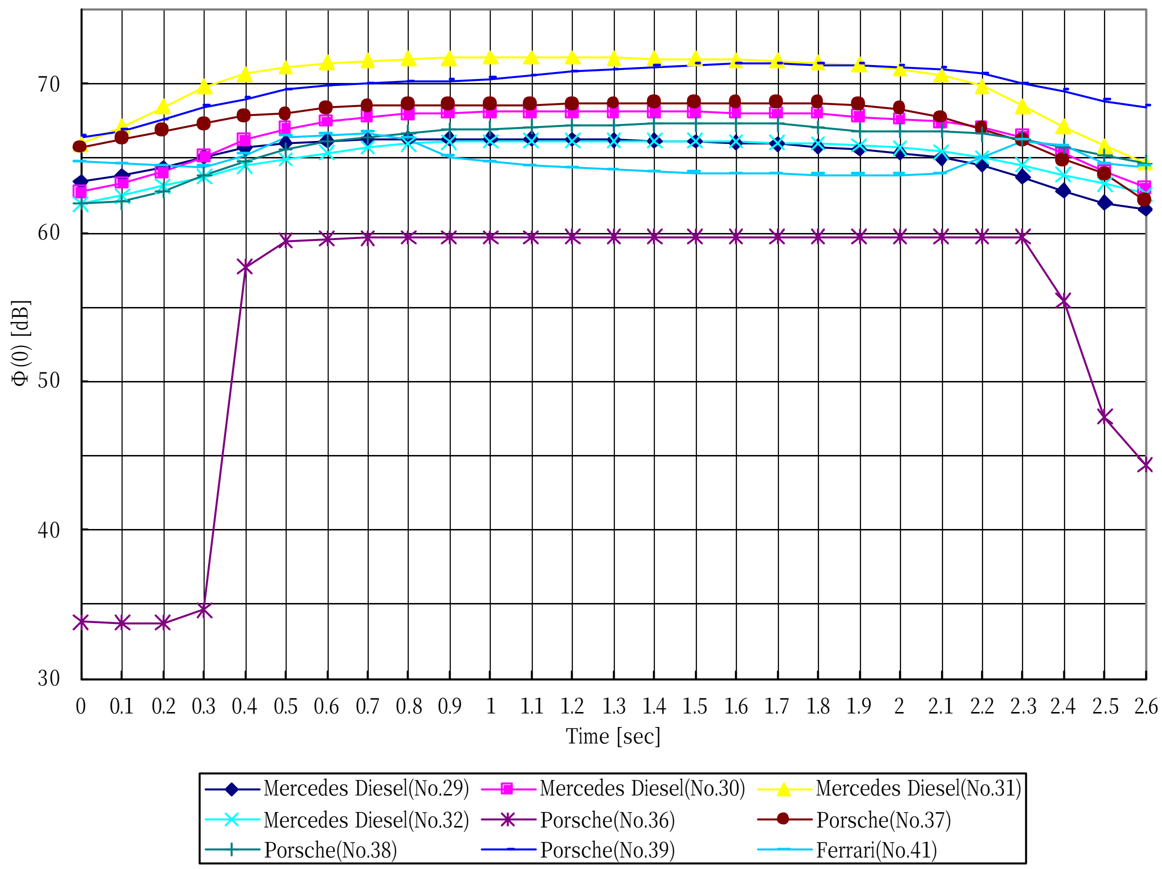
By measuring an actual noise, this program can learn to get the template for identification of noise. It helps to raise the rate of correct identification. This learning function has a manual learning mode and an automatic learning mode. Matters that demand special attention is that a user must correct the noise source if the program makes wrong identification of it. If not, the program will do wrong learning and the probability of correct identification will be getting worse after that.

The template of sound can be adjusted in "Noise Source Template" dialog. The average value, the maximum value and the minimum value about four factors, $\Phi(0)$, τ_e, τ_1 and ϕ_1 are input. The maximum value and the minimum value are valid only when its check-box is checked. The measured data that is beyond the range of setting value will be removed from its template. The more the value in "Weighting" area is large, the more its factor has great influence in identification. (Appendix C)

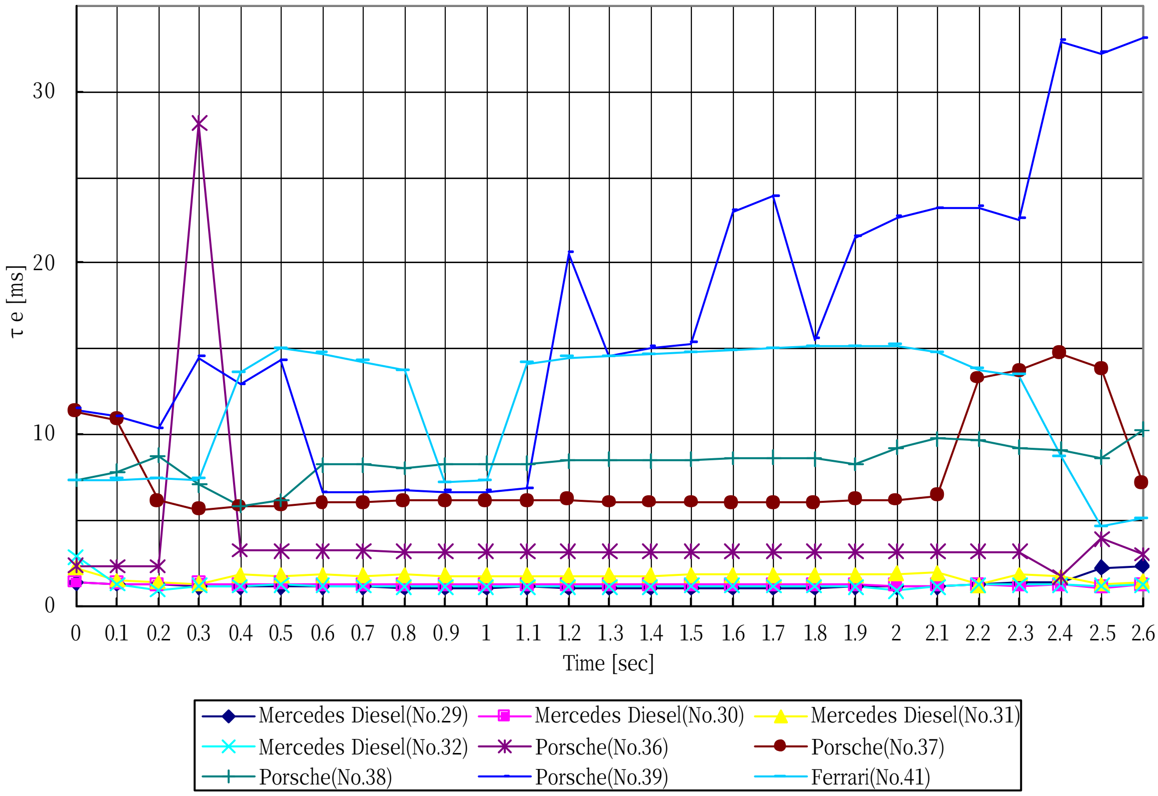
Measured data will be stored in the database, and it can be analyzed more detailer by "Acoustic Analyzing System". The data of "Acoustic Analyzing System" was shown as Figure 5.4. The data of Porsche's engine sound and Mercedes Diesel's one is difference clearly, having each characteristic. The data of template was shown as Figure 5.5.

Figure 5.4. Example of $\Phi(0)$, τ_e, τ_1 and ϕ_1 of Porsche Turbo, Mercedes 200 diesel, and Ferrari analyzed by Acoustic Analyzing System.

$\Phi(0)$:

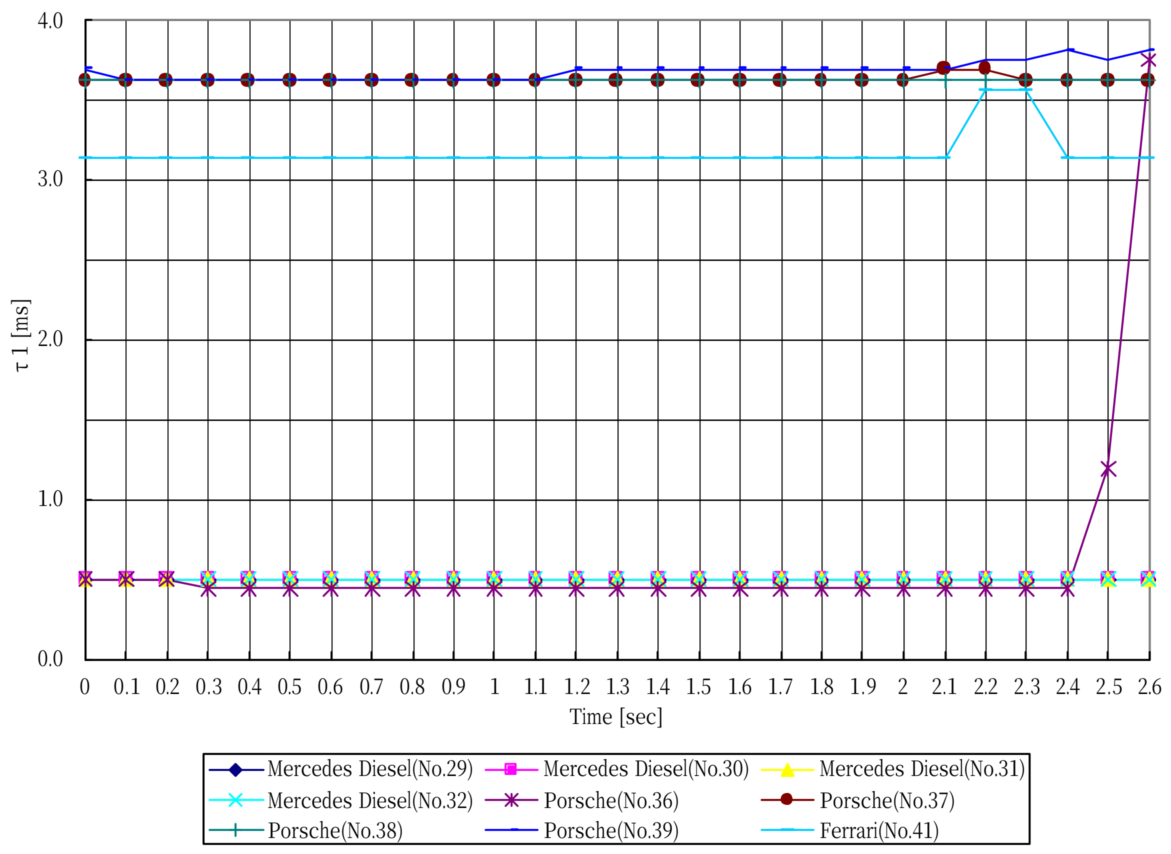


τ e:



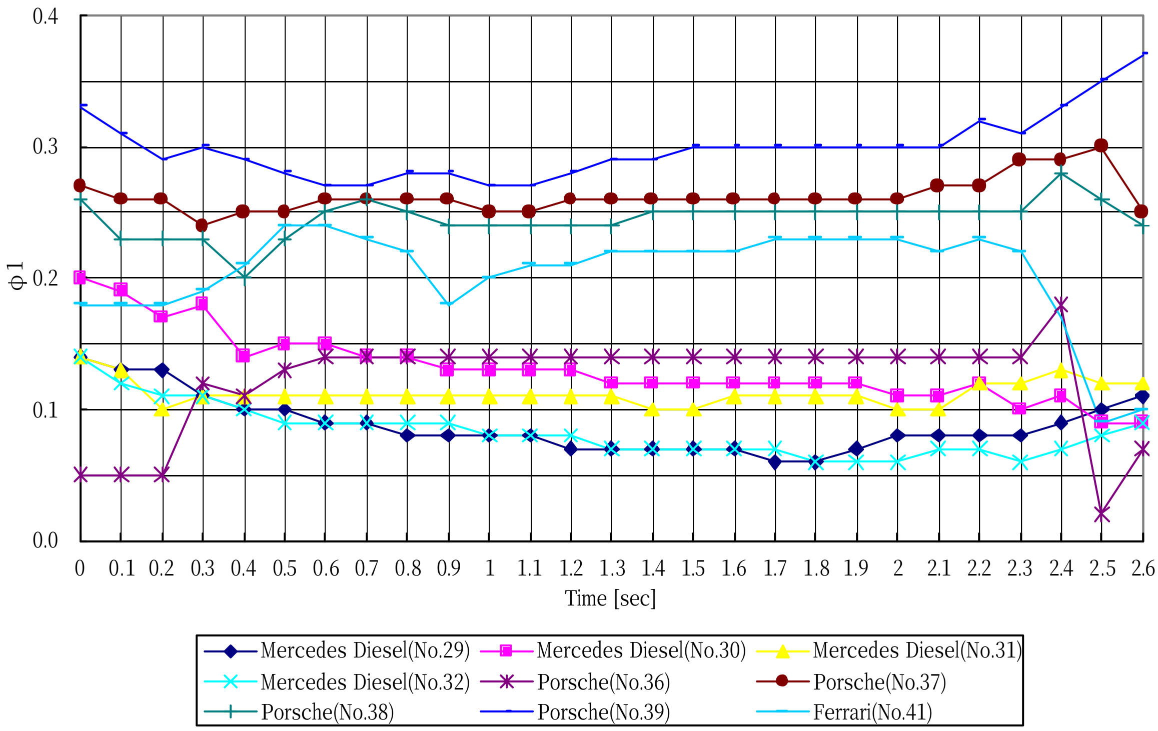
τ e [ms]
30
20
10
0
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
1.1
1.2
1.3
1.4
1.5
1.6
1.7
1.8
1.9
2
2.1
2.2
2.3
2.4
2.5
2.6
Time [sec]
Mercedes Diesel(No.29)
Mercedes Diesel(No.30)
Mercedes Diesel(No.31)
Mercedes Diesel(No.32)
Porsche(No.36)
Porsche(No.37)
Porsche(No.38)
Porsche(No.39)
Ferrari(No.41)

τ_1:



4.0
3.0
2.0
1.0
0.0
τ 1 [ms]
0 0.1 0.2 0.3 0.4 0.5 0.6 0.7 0.8 0.9 1 1.1 1.2 1.3 1.4 1.5 1.6 1.7 1.8 1.9 2 2.1 2.2 2.3 2.4 2.5 2.6
Time [sec]
Mercedes Diesel(No.29)
Mercedes Diesel(No.30)
Mercedes Diesel(No.31)
Mercedes Diesel(No.32)
Porsche(No.36)
Porsche(No.37)
Porsche(No.38)
Porsche(No.39)
Ferrari(No.41)

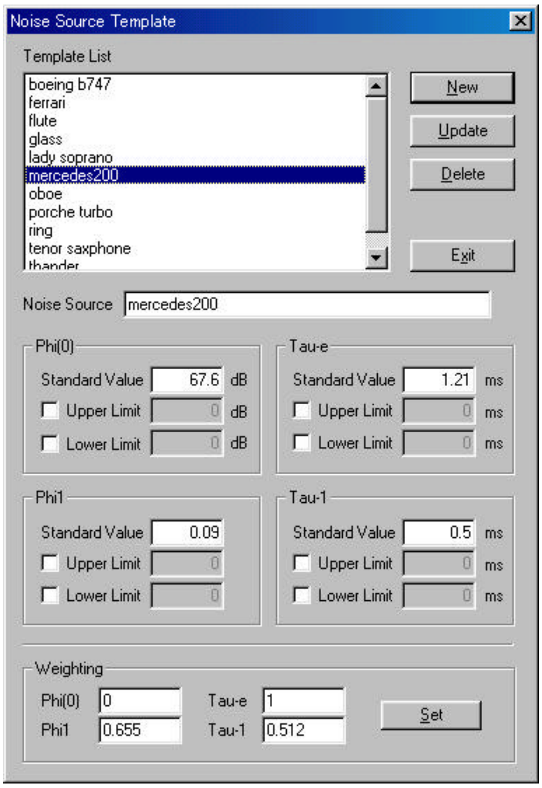
ϕ_1:



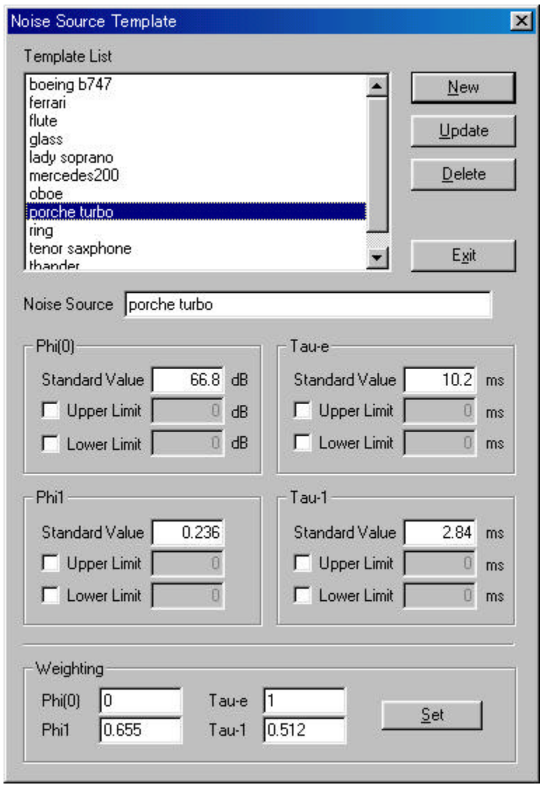
0.4
0.3
0.2
0.1
0.0
φ1
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
1.1
1.2
1.3
1.4
1.5
1.6
1.7
1.8
1.9
2
2.1
2.2
2.3
2.4
2.5
2.6
Time [sec]
Mercedes Diesel(No.29)
Mercedes Diesel(No.30)
Mercedes Diesel(No.31)
Mercedes Diesel(No.32)
Porsche(No.36)
Porsche(No.37)
Porsche(No.38)
Porsche(No.39)
Ferrari(No.41)

Figure 5.5. Noise source template of Porsche Turbo and Mercedes 200 Diesel.

Porsche Turbo



Mercedes 200 Diesel



5.4. Further study of signal of environmental noise

There are four orthogonal factors of sound fields, namely, listening level LL, initial time delay gap between the direct sound and the first reflection Δt_1, subsequent reverberation time T_{sub}, and IACC. The IACC (maximum of the IACF) is related to the subjective diffuseness. In addition to the IACC, τ_{IACC} (delay time at the IACC obtained) and W_{IACC} (width of the maximum) extracted from the IACF are deeply related to the image shift of sound source and the apparent source width (ASW), respectively. All of these factors must be considered to describe the primary sensations as well as the subjective preference [62].

1. The noise level: listening level LL at both ears. (1)

$$LL = [\Phi_{ll}(0)\Phi_{rr}(0)]^{1/2} \tag{1}$$

2. The spatial information, for example, may be expressed by

$$s_t = f_t\,(LL, IACC, W_{IACC}, \tau_{IACC})_{right} \tag{2}$$

Chapter 6

Conclusion

6.1 Simulation System

We developed a system that computes independent physical factors in a sound space from building drawings at the design stage and can actually estimate the factors at each seat.

6.2 Diagnostic System for Sound Field in the Room

We built a system to diagnose the sound field at each seat in the actual building and a model based on a functional model of the human sense of hearing (cerebral system). We clarified the usefulness of the system by comparing the results *6.1* obtained at the design stage with those from our actual diagnosis.

6.3 Seat Selection System

We built a seat selection system to find the optimum location for each auditory sense and developed a system that indicates the range of seats for which different individuals will have the highest preference.

6.4 Diagnostic System for Environmental Noise

A system for measuring temporal and spatial factors in a noise environment was developed. This provided additional information about the physical characteristics of noise sources, suggesting its usefulness for making a mental evaluation based on distinctions between noise sources.

This development should lead to a better fundamental understanding of 1) the processes for designing and evaluating indoor sound fields and 2) individual sensitivities - for example the size (loudness) and height (a pace) of the sound, three elements of the tone, unpleasantness, spatial sense, and individual sensitivity in such cases as a mental time sense.

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Appendix

Computational parts of the program lists for Each Systems

Appendix A. Simulation System

```
/*
**  system's header files
*/
#include <windows.h>
#include <string.h>
#include <stdio.h>
#include <math.h>

/*
**  user's header files
*/
#include "..¥..¥inc¥common.h"
#include "..¥..¥inc¥mimdef.h"
/*
**  prtotype functions
*/
extern void KmCxdr(double *,double *,double,double *,double *,double *);

double    zDX[3],zDY[3],zDZ[3],zAX,zBY,zCZ,zDD;

void KmClfg(double *CENT,double *XAXS,double XROT,double *A,double *B);

/*
** <Title>  Transform the equation of spherical surface by local coordinates system into generalized coordinates.
*/
void KmClfg(double *CENT,double *XAXS,double XROT,double *A,double *B)
{
```

```
/*
** double CENT[3]    (I  ) :position of spher by local coordinates system
** double XAXS[3]    (I  ) :direction of X-axis
** double XROT       (I  ) :rotation angle with X-axis
** double A[4]       (I  ) :the equation of spherical surface
** double B[4]       (  O) :the equation of spherical surface by generalized coordinates
*/

    /* Calculate the direction of axis. */
    KmCxdr( CENT, XAXS, XROT, zDX, zDY, zDZ );

    /* Calculate the equation of spherical surface. */
    zAX = zDX[X_CO] * A[KSU_A] + zDY[X_CO] * A[KSU_B] + zDZ[X_CO] * A[KSU_C];
    zBY = zDX[Y_CO] * A[KSU_A] + zDY[Y_CO] * A[KSU_B] + zDZ[Y_CO] * A[KSU_C];
    zCZ = zDX[Z_CO] * A[KSU_A] + zDY[Z_CO] * A[KSU_B] + zDZ[Z_CO] * A[KSU_C];

    zDD = -(zAX * CENT[X_CO] + zBY * CENT[Y_CO] + zCZ * CENT[Z_CO]) + A[KSU_D];

    B[0] = zAX;
    B[1] = zBY;
    B[2] = zCZ;
    B[3] = zDD;

    return;
}
```

Appendix B. Diagnostic System for Sound Fields in the Room

The point where the energy which does not include the minimum reflection time of the reflection from a floor is maximum is obtained by the sum of all frequency band.

```
////////////////////////////////////////////////////////////////////////////
//
//      function  :calculation of  Δt1
//
//      argument  :pData - impulse answer data
//                 nData  - number of impulse answer data
//                 fRate  - sampling rate
//                 minTime  - minimum delay time of reflection
//                 pT0  - delay time of the direct sound(output)
//                 pT1  - delay time of the first reflection sound(output)
//
//      return  value :no
//
////////////////////////////////////////////////////////////////////////////
void CalcParamDeltaT1(float *pData, int nData, double fRate, double minTime,
        double  *pT0,  double  *pT1)
{
    float  T0Level,  T1Level;
    int  T0Pos,  T1Pos;
    float  maxLevel;
    int  i;
    float  data;
    int  startPos;
    BOOL  bUpFlag;

    //  Ascertain  exsitence of impulse answer data.
    if (pData == NULL) {
        *pT0  =  0;
        *pT1  =  0;
        return;
    }
```

```
// Calculate the maximum peak value of impulse answer data.
 maxLevel = 0;
for (i = 1; i < nData / 2; i++) {
      data = (float)fabs(pData[i]);
      if (data > maxLevel)
            maxLevel = data;
  }

// Calculate the delay time of the diredt sound.
 T0Level = 0;
 T0Pos = 0;
for (i = 1; i < nData / 2; i++) {
      data = (float)fabs(pData[i]);
      if (data > (float)fabs(pData[i - 1]) && data > (float)fabs(pData[i + 1])) {
           if (data > T0Level) {
                  T0Level = data;
                  T0Pos = i;
           } else if (T0Level > maxLevel / 5)
                   break;
        }
  }

 // Calculate the position where it passed for minimum delay time of reflection from direct sound.
startPos = (int)(T0Pos + minTime / 1000 * fRate) + 1;
 if (startPos >= nData)
      startPos = nData - 1;

// Calculate the maximum peak value after above positon.
 maxLevel = 0;
for (i = startPos; i < nData / 2; i++) {
      data = (float)fabs(pData[i]);
      if (data > maxLevel)
            maxLevel = data;
  }

// Calculate the delay time of the first reflection sound.
```

```
    T1Level = 0;
    T1Pos = 0;
    bUpFlag = FALSE;
    for (i = startPos; i < nData / 2; i++) {
        data = (float)fabs(pData[i]);
        if (data > (float)fabs(pData[i - 1]) && data > (float)fabs(pData[i + 1])) {
            if (bUpFlag) {
                if (data > T1Level) {
                    T1Level = data;
                    T1Pos = i;
                } else if (T1Level > maxLevel / 2)
                    break;
            } else {
                if (T1Level != 0 && data > T1Level && data > maxLevel / 10)
                    bUpFlag = TRUE;
                else {
                    T1Level = data;
                    T1Pos = i;
                }
            }
        }
    }

    // Calculate the time direct sound and first reflection sound in milli-second unit.
    *pT0 = (double)T0Pos / fRate * 1000;
    *pT1 = (double)T1Pos / fRate * 1000;
}
```

Appendix C. Diagnostic System for Environmental Noise

Algorithm for the identification of the sound source

```
//////////////////////////////////////////////////////////////////////////
//
//          function    :identification with noise source
//
//          argument    :pNmsFactorData - AFC factor data
//                        nNmsFactorData  - number of AFC factor data
//                        pNoiseSrcData  - data of noise source(output)
//
//          return  value  :ID-number of noise source template
//
//////////////////////////////////////////////////////////////////////////
long Identification(NmsFactorData *pNmsFactorData, int nNmsFactorData, NoiseSrcData *pNoiseSrcData)
{
     CDbNsTmp  dbNsTmp;
     DbNsTmpRec  dbNsTmpRec;
     NsTmpData  nsTmpData;
     AcfFactorData  acfFactorData;
     float  fdPhi0;
     float  fdTaue;
     float  fdTau1;
     float  fdPhi1;
     float  fdTotal;
     float  fdMin;
     CString  str;
     NsWeightData  nsWeightData;
      long nNoiseTmpID = -1;

    // Open the database about noise source template.
     if  (!dbNsTmp.Open())
          return  nNoiseTmpID;

    // When AFC factor data is zero, noise source can't identify.
```

```
if (nNmsFactorData == 0)
    return nNoiseTmpID;

 // Acquisition of weight coefficient.
 ReadNsWeightData(&nsWeightData);

// Acquire the factor whose Φ(0) made the target of identification is the biggest.
acfFactorData = GetNoiseAcfFactor(pNmsFactorData, nNmsFactorData);

 // The initialization of the name of noise source.(initial name is unknown)
 str.LoadString(IDS_UNKNOWN);
 strcpy(pNoiseSrcData->name, str);

// The initialization of distance variable.
fdMin = 1000;

// Look up the template that distance becomes the smallest.
for (;;) {
    // Acquisition of noise source template.
    if (!dbNsTmp.ReadRecNext(&dbNsTmpRec, &nsTmpData))
        break;

    // Check limit value, and calculate a distance inside the range.
    if ((!nsTmpData.bPhi0LowerCheck || acfFactorData.fPhi0 >= nsTmpData.fPhi0LowerLimit)
            && (!nsTmpData.bPhi0UpperCheck || acfFactorData.fPhi0 <= nsTmpData.fPhi0UpperLimit)
            && (!nsTmpData.bTaueLowerCheck || acfFactorData.fTauE >= nsTmpData.fTaueLowerLimit)
            && (!nsTmpData.bTaueUpperCheck || acfFactorData.fTauE <= nsTmpData.fTaueUpperLimit)
            && (!nsTmpData.bTau1LowerCheck || acfFactorData.fTau1 >= nsTmpData.fTau1LowerLimit)
            && (!nsTmpData.bTau1UpperCheck || acfFactorData.fTau1 <= nsTmpData.fTau1UpperLimit)
            && (!nsTmpData.bPhi1LowerCheck || acfFactorData.fPhi1 >= nsTmpData.fPhi1LowerLimit)
            && (!nsTmpData.bPhi1UpperCheck || acfFactorData.fPhi1 <= nsTmpData.fPhi1UpperLimit)) {

        // Calculate the distance between template with factor of data that was measured.
        fdPhi0 = (float)fabs(acfFactorData.fPhi0 - nsTmpData.fPhi0Standard) / 10 * nsWeightData.fPhi0;
        fdTaue = (float)fabs(log(acfFactorData.fTauE) - log(nsTmpData.fTaueStandard)) * nsWeightData.fTaue;
        fdTau1 = (float)fabs(log(acfFactorData.fTau1) - log(nsTmpData.fTau1Standard)) * nsWeightData.fTau1;
        fdPhi1 = (float)fabs(log(acfFactorData.fPhi1) - log(nsTmpData.fPhi1Standard)) * nsWeightData.fPhi1;
```

```
            fdTotal = fdPhi0 + fdTaue + fdTau1 + fdPhi1;

            // Output the template --factor and name of noise source-- of the minimum distance.
            if (fdTotal < fdMin) {
                fdMin = fdTotal;
                pNoiseSrcData->fdPhi0 = fdPhi0;
                pNoiseSrcData->fdTauE = fdTaue;
                pNoiseSrcData->fdTau1 = fdTau1;
                pNoiseSrcData->fdPhi1 = fdPhi1;
                strcpy(pNoiseSrcData->name, dbNsTmpRec.name);
                nNoiseTmpID = dbNsTmpRec.nsTmpID;
            }
        }
    }

    return nNoiseTmpID;
}
```