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The Highly Preferred Sound Levels of Ambience Microphone Arrays to Front Microphone Arrays with the Fixed Levels for Surround Recordings

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ABSTRACT

Recording sessions of orchestra in a concert hall was held simultaneously using eight frontal microphone arrays, seven rear microphone arrays and thirty-six spot microphones. The comparison of these microphone techniques to find the highly preferred scale in subjective listening test by seven experienced sound engineers was attempted. The result of the experiments led to make a following hypothesis: the most preferred - values of the ambience microphone arrays to the front microphone arrays with the fixed levels will be possibly decided according to common preferences of sound engineers. The estimated scores were around 1dB in dispersion and -6±2dB in average. Furthermore, the related study of the C - value which means an energy ratio of direct sounds vs reverberation to each microphone techniques was examined. The hypothetical C - value was the index for the most preferred_mixings on the ambience microphone arrays to the frontal microphone arrays.

1. INTRODCUTION

The AES Surround Study Group was founded by AES Japan in late 2005 to evaluate the surround sound microphone techniques. The recording session of the orchestra in a concert hall was conducted on September 26th and 27th, 2006. The performance of musical selections by Osaka Philharmonic Orchestra in the Symphony Hall, Osaka had been simultaneously recorded in multi-channels employing various kinds of microphone techniques. As a result, the precise comparisons of the microphone arrangements could be easily made thanks to the abundant sound sources under the same condition of recordings. As the following subjective listening tests were planned to hold in various places for further studies of the microphone arrangements, it is indispensable to seek the highly preferred levels for the ambience microphone arrays to the front ambience microphone arrays.

2. RECORDINGS

2.1. The Hall

All orchestra pieces were recorded at the Symphony Hall in Osaka, Japan, a hall exclusively designed for classical music with a proud of two seconds of the reverberation time in fully occupied seats. The Symphony Hall was built by Asahi Broadcasting Corporation in 1982 as a commemorative business of the 30th anniversary foundation of the company; constructed by Taisei Corporation and sound-designed by Kiyoteru Ishii.

Photo1.1 Evening View of the Symphony Hall

The ground plan of the hall in Figure 1.1 and the cross section in Figure 1.2 are shown respectively. The average height of the hall from the first floor to the ceiling is 20.7 m; the distance from the rear wall of the stage to the back wall on the first floor is 35m. The average dimension from the tip of the stage to the rear wall of each floor is 28.3 m long. The average width between left and right sides of the wall on the first floor is 31.7 m. The dimension of the stage is approximate 24 m long and 12 wide. The seat capacity is 1854.

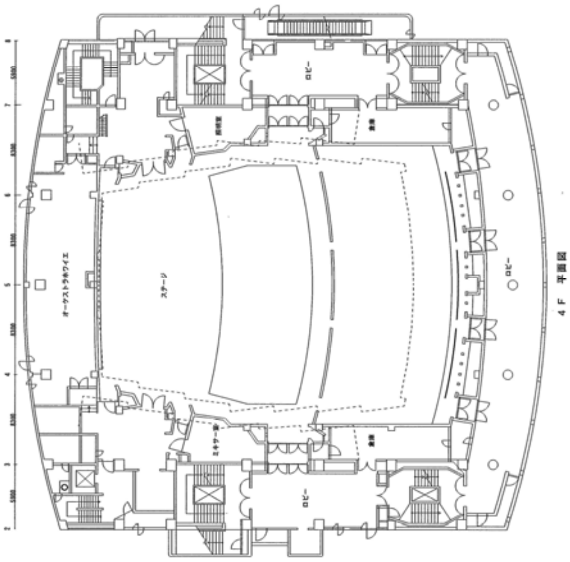

Fig.1.1 Floor plan of The Symphony Hall

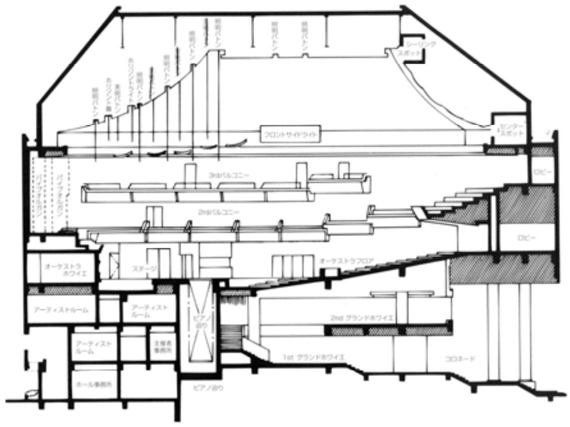

Fig.1.2 Cross Section of the Symphony Hall

Photo.1.2 Stage of the Symphony Hall

2.2. Performance

The recording session was held with the performance of the Osaka Philharmonic Orchestra, conducted by Shigeo Genda on September 26th and 27th, 2006.

2.3. The Music Selections

Three compositions for the recordings were selected.

1) Ottorino Respighi, a Symphonic Poem "Pines of Rome"

2) Ludwig van Beethoven, "Wellington's Victory"

3) Wolfgang Amadeus Mozart, an Overture to "The Marriage of Figaro"

1) "Pines of Rome" including a variety of orchestra elements is the ideal piece for subjective listening tests. As the first movement does not have bass instruments such as double basses, it can provide people a chance to evaluate sound quality without low pitch tone, which is hardly affected by the timbral differences between omni-directional microphones and cardioid microphones. The organ's deep bass sounds in the second movement are suitable for the LFE channel test. The third movement is ideal for sound localization because of numerous solo instrumental parts. The fourth movement with the "banda (a fanfare group)" sounds from the back of the audience represents dynamism. It is a suitable piece for evaluating dynamic surround audio.

2) The placement of the instruments in "Wellington's Victory" is also suited to surround sound performance. Following the Beethoven's notes on the placement of the orchestra, two marching bands as British and French armies were set up at the different locations of an orchestra. In consideration of surrounding listeners, the British army band played on the left at the rear of the audience on the first floor and the French army band performed in the upper audience on the second floor at the rear of the orchestra.

3) The overture to "the Marriage of Figaro has full of well-balanced sounds for a unit orchestra. It is a suitable number for sound evaluation test.

2.4. Microphone Techniques for Recordings

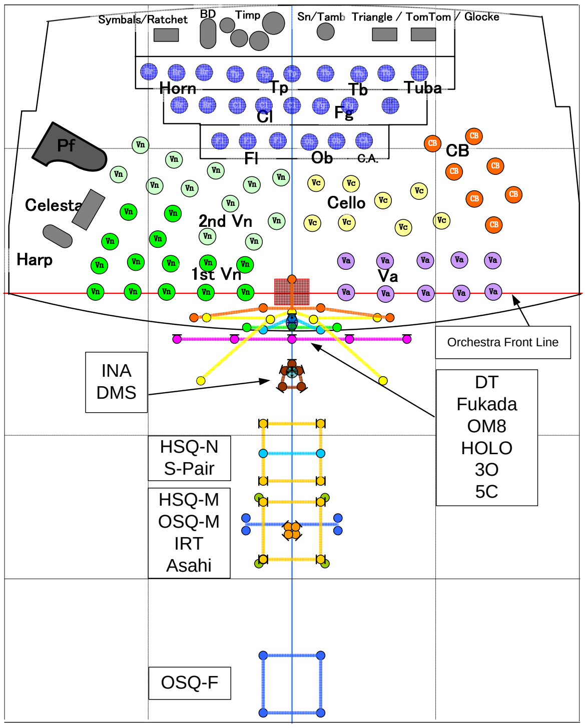

Fig.2.1 Layout of the Microphone Arrays

Figure 2.1 shows the layout of the microphone arrays for surround audio recording in 5 m/grid. Five types of exclusive surround sound microphone arrays (Chapter 2.4.1.); three types of front microphone arrays modified from the two channel stereo arrangement (Chapter 2.4.2); and seven types of ambience microphone arrays including thirty-six spot microphones (Chapter 2.4.3.), were used. Ninety-eight microphones in total were connected to the microphone amplifier and recorded on the DAW (ProTools HD3). Please note that the "Orchestra Front Lines" in the following configurations was set along the first row of the first violins and the violas of the orchestra through the center of a podium. The center of the podium was used as a datum point for the distance measurement.

2.4.1. The Surround Sound Microphone Techniques

- 1) Fukada Tree (to be called Fukada hereafter)

The original Fukada consists of five cardioid microphones (L, C, R, Ls and Rs) and two omni-directional microphones on the sides (LL and RR). In this recording session, however, Mr.Fukada, an inventor of the Fukada, suggested to use omni-directional microphones for L, C, R, LS and RS to add richness to the low range of sounds. The horizontal angle of these front omni-directional microphones was decided according to its directivity in high frequency. On the other hand, the rear omni-directional microphones made its directivity narrower in mid and high frequencies, using the Kugels, an acoustic equalizer. The separation of the sounds from the rear and the front microphones could be clearly identified.

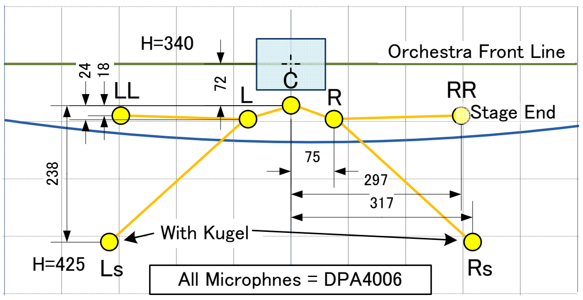

Fig. 2.2 Configuration of “Fukada Tree”

- 2) INA5 “Ideale Nierenanordnung 5” (to be called INA hereafter)

The INA shown in Figure 2.3 is also known as the “Ideal Cardioid-microphone Arrangement”. The INA optimizes the placement of each microphone to record in 360°angles concerning a possibility of reproducing the sounds from all directions. The original recording angle for three front channels is at 180°. However, in this recording session, as the INA was placed far from the orchestra, the recording angle for the three was set at approximately 150°. The dimension of the microphones was slightly enlarged, as referred to the information on the Williams [8]. Figure 2.4 shows the enlarged configuration of the INA.

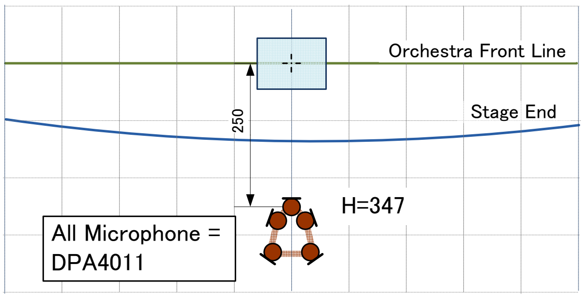

Fig. 2.3 Configuration of “INA 5”

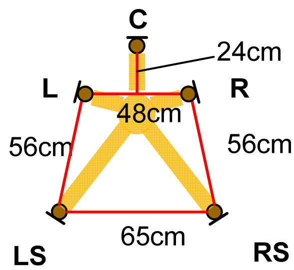

Fig. 2.4 Enlarged Configuration of “INA 5”

- 3) Omni 8 “Omni-directional Microphones & “Figure of 8” Microphone” (to be called OM8 hereafter)

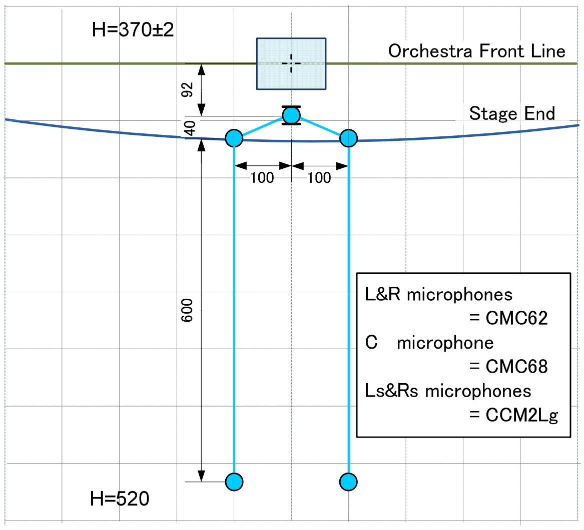

Fig. 2.5 Configuration of “Omni8”

This technique called “Omni 8” in Figure 2.5 was devised by Mr. Kawekawa at Tokyo National University of Fine Arts and Music. Two omni-directional microphones and one “Figure of 8” microphone are arranged in the front array; and a pair of ambience microphones of either omni-directional or cardioid is set at the rear. In this recording, the combination of the front array and rear with a pair of omni-directionals as a rear pair was used in the distance of 6 m between two arrays. Please note that only the front microphones of the OM8 were used when the experiment for the preferable mixing levels of the ambience microphone arrays was made.

- 4) Double MS (to be called DMS hereafter)

Fig.2.6 shows a scheme that the MS method is applied to the rear likewise. Two cardioid microphones, each one is directed to front and rear (Mfront, Mrear hereafter referred to Mf and Mr), and a "Figure of 8" microphone directed toward left and right (S) are MS decoded according to the following scheme to obtain 4-channel signals (L, R, Ls, Rs). (In the equations, a minus sign signifies the opposite phase.)

* L = Mf +S;
* R = Mf - S;
* LS = Mr +S;
* RS = Mr - S;

Though we used the above 4ch decoding in our experiment, Schopes GmBH is recommending 5ch decoding. Because that the Schopes' recommendation was not yet made public at the planning stage and that the said decoding scheme was only brought to us just before our experiment, we had to pass on adopting it considering the practical accomplishment yet experienced by us. For that matter, please take it into consideration that our subjective study might have led different conclusions, if the said 5ch decoding were used. (See the reference [9])

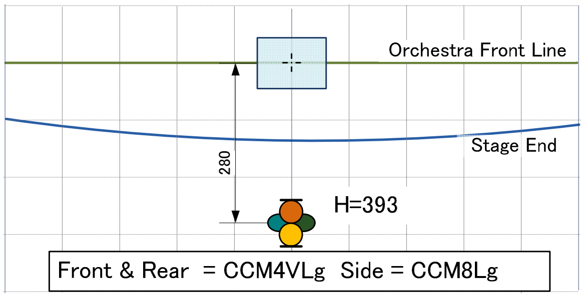

Fig. 2.6 Configuration of "Double MS"

- 5) Holophone H2-PRO

The Holophone H2-PRO is a surround sound microphone of Rising Sun Production Ltd., Canada. The elliptical shaped microphone system is designed for capturing up to 7.1 channels of surround sound.

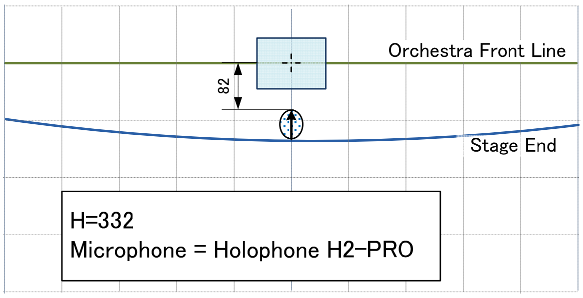

Fig. 2.7 Configuration of "Holophone H2-PRO"

2.4.2. The Front Microphone Techniques

It is the front microphone techniques that were modified from the traditional stereo microphone arrangements for this surround sound recording. Three types were concerned in this recording.

- 1) Decca Tree (to be called DT hereafter)

The technique of the DT, originally developed by British engineers in 1950s, is a setup of three omni-directional microphones on a triangular metal support and of a pair of sub-microphones for the left and right. According to the original DT technique, the center microphone is placed for using as phantom sound sources for stereo sound imaging. In the surround sound recording, it was directly connected to a C channel.

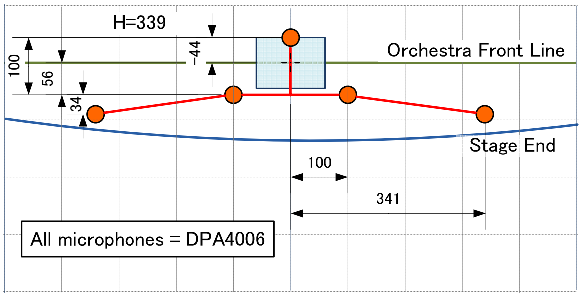

Fig. 2.8 Configuration of "Decca Tree"

- 2) Three Omni-directional Microphones (to be called 3O hereafter)

Three omni-directional microphones were placed in a straight line shown in Figure 2.9. As well as the DT technique, the center microphone was connected to a C channel.

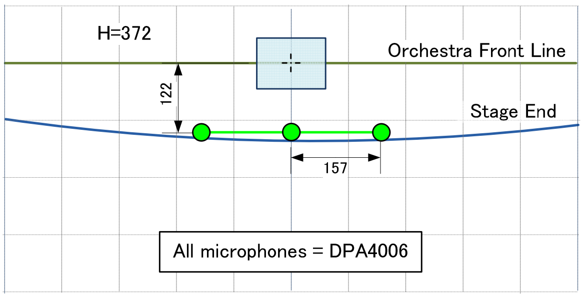

Fig. 2.9 Configuration of "Three Omni-directional Microphones"

- 3) Five Cardioids

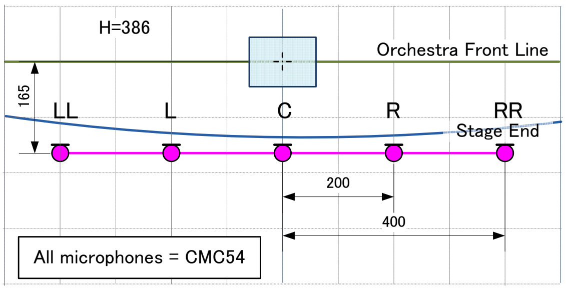

Fig. 2.10 Configuration of "Five Cardioid Microphones"

Five cardioid microphones were placed every two meters in a straight line. Signals from L and R microphones are added to the LL and C channels and the C and the RR channels, respectively, with 3dB attenuation..

2.4.3. The Ambience Microphone Techniques

The ambience microphones techniques are generally applied for the purpose of recording of reverberation in a hall. Especially, a square of four ambience microphones without the center is called a "Square Array". Only two microphones are sometimes used at the rear. Seven types of ambience microphone arrays were prepared for this recording.

- 1) Hamasaki-Square in Zone 1 (to be called HSQ-N "near" hereafter)

The HSQ-N in Figure 2.11 shows that each null point of four "Figure of 8" microphone was facing toward the stage with positive lobe of the microphone directivity pattern toward the outside. The HSQ-N was arranged in square (spacing D: 2 m). The distance between the center of the square and the center of a podium was 5.6 m.

- 2) Hamasaki-Square in Zone 2 (to be called HSQ-M "middle" hereafter)

The same configuration as that of the HSQ-N is seen in Figure 2.11. The distance from the center of a podium was 8.3 m to set.

- 3) IRT cross (to be called IRT hereafter)

The square arrangement of four cardioid microphones (D: 25 cm) is seen as IRT in Figure 2.11. Each microphone of the IRT was facing toward outside at an angle of 90°. The distance from the center of a podium was 8.3m.

- 4) Omni-Square in Zone 2 (to be called OSQ-M "middle" hereafter)

The square arrangement of four omni-directional microphones (spacing D: 2 m) and its heads at the downward is seen as OSQ-M in Figure 2.11. The distance from the center of a podium was 8.3m.

- 5) Omni-Square in Zone 3 (to be called OSQ-F "far" hereafter)

The same configuration as that of the OSQ-M is seen in Figure 2.11. The distance from the center of a podium was 13.7m.

- 6) Asahi-Method (to be called Asahi hereafter)

Asahi Broadcasting Corporation devised the modified version of the A-B Method shown in Figure 2.11. Two sets of front and rear omni-directional microphones were positioned on the left and the right sides in 3 m of the line. The distance between the front and the rear was 44 cm. The distance from the center of a podium was 8.1m.

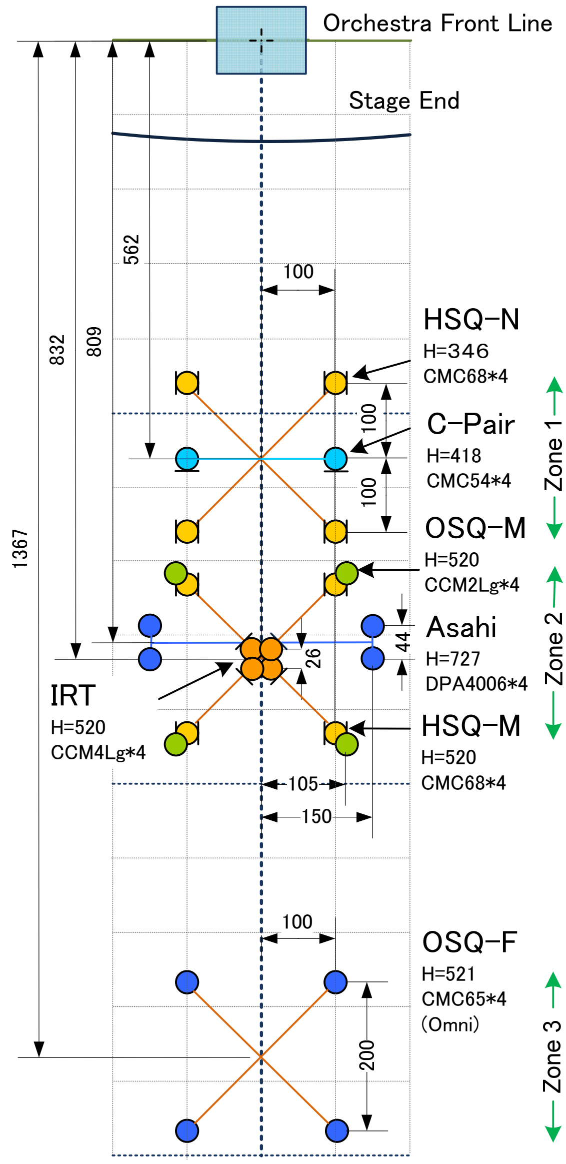

Fig. 2.11 Configuration of "7 Ambience Microphone Arrays"

- 7) Cardioid Pair (to be called C-Pair hereafter)

Two cardioid microphones facing toward the audience in the A-B method were placed as shown in Figure 2.11. The distance between the left and the right microphones was 2m and the one from the center of a podium to the C-pair was 5.6 m. The C-Pair was only used in for the recording of "Wellington's Victory".

3. SUBJECTIVE LISTENING TEST 1

3.1. Introduction

Two types of microphone techniques, as mentioned earlier, were generally used for the surround sound recordings: the microphone arrays designed exclusively only for the surround sounds; and the combination of the front microphone arrays which have been modified from the traditional stereo recordings and the ambience microphone arrays specialized for recordings of reverberation. In this test, the latter was applied.

The subjective listening test was examined with the sound sources that had been colleted from the simultaneous recording under the fortunate circumstance in order to find the highly preferred levels for mixing. The test was conducted in a participation of seven leading professional sound engineers as a subjects at Sound Production Studio of Tokyo National University Fine Arts and Music, which is built with accordance to the recommendation of ITU-RBS.775-1 on December 14^{th} and 15^{th}, 2006. The subjects who were mixing the ambience microphone levels according to their professional decisions, had been following the method of adjustment.

3.2. Loudness Matching Test

The loudness matching test was done by three sound engineers in prior to the subjective listening test. First, the pink noise was reproduced on the dodecahedron speaker at the back position of the stage in 6 m from the center of a podium, and then the sound signals over all microphones were recorded in multi-channels simultaneously. Then, the loudness matching of each channel was tested with reference to the 3O center channel. To avoid influence of sound characteristics by type of microphone, a frequency range was limited to 250Hz-4kHz by a band-pass filter. The level differences among three subjects was within ±0.5dB range, which could be said as a quite small value. Therefore, the final value of loudness matching adapted the average of these three levels.

3.3. Balance of Levels of Microphones in Each Array

The adjustments of the microphone level balances were decided in the following three methods upon the agreement of seven sound engineers who had been simultaneously listening to the sound sources. Because there were not much unevenness of the preferred levels within seven engineers including inventors of the Fukada and the OM8, so that the most preferred levels were decided.

1) The balances of the DT, the 3O and the 5C and also the front microphones of the OM8 were adjusted by sound engineers as well as that of the Fukada although it was not used for this experiment. Having followed the sound levels of the L/R microphones as references, the preferred levels of the center microphone, the rear microphones and the LL/RR sub-microphones were decided as shown in Table 3.1.

2) Concerning the highly preferred balance of four microphones in the ambience microphone arrays, each individual microphone was set in the equal sound levels.

Table.3.1 Preferred Sound Levels(dB) in Microphones versus the Compositions

Array	CH	Roma	Wellington	Figaro	Toccata
DeccaTree	LL	-6	-6	-6	-6
	L	0	0	0	0
	C	-3	-3	-3	-3
	R	0	0	0	0
	RR	-6	-6	-6	-6
FukadaTree	LL	-6	-6	-6	-6
	L	0	0	0	0
	C	-2.5	-2.5	-2.5	-2.5
	R	0	0	0	0
	RR	-6	-6	-6	-6
	SL	-4	-6	-4	-2
	SR	-4	-6	-4	-2
3 Omni	LL	0	0	0	0
	C	-1	-1	-1	-1
	RR	0	0	0	0
Omni & 8	L	0	0	0	0
	C	-0.5	-0.5	-0.5	-0.5
	R	0	0	0	0

3) The loudness matching of all the arrays was applied on the base of the sound level of the DT in the same method as 3.2.

*The C-Pair was reproduced by using only two rear loudspeakers. Therefore the reproduced level per channel will be increased 3dB theoretically. For the 5C, all microphones were set same levels. For convenience of the reproduction level settings, the faders were fixed at the 0dB position on the mixing console and the gain of each audio file was adjusted. .

3.4. Music Selection for Listening Test

The music pieces for the listening test were selected as follows: the first movement of “Pines of Rome”, the second movement of “Willington’s Victory” the overture to "the Marriage of Figaro" and “Toccata and the Fugue". Please note that “Toccata and the Fugue" was excluded because its evaluation test was made only on the DT due to a limited time and then there was no sufficient number of data to be analyzed.

3.5. Reproduction Levels for Listening Test

It is hard to prescribe the reproduction levels of orchestra pieces whose dynamic range is quite broad. When the last movement of “Pines of Rome” was peaked at the highest range of loudness, the largest level on the DT center channel showed -3dBFS or so. During listening to the part of this piece whose sound level appeared about -3dBFS, seven sound engineers chose the preferred loudness in monitor volume. The fixed preferred level was being kept in the monitor volume and then 0VU of the pink noise was reproduced. As a result, the reproduced sounds were 85dBSPL/ch in the sound pressure level meter.

3.6. Procedures of Listening Test

In this experiment, the level of “0” was first fixed on all faders for the front microphone arrays. Then, seven sound engineers one after another adjusted the faders of the ambience microphone arrays to their preferred levels, with seating position at the sweet spot described in ITU-R BS775-1. The faders of both arrays were prepared on the “D-Command”, the ProTools mixing console. The fader levels could be seen down to the first decimal value in dB on the display of the ProTools mixing console.

The preference choice of the levels of six ambience microphone arrays (seven in "Willington's Victory) to four front microphone arrays in the orchestral selection (described in 3.4) was first done individually by seven sound engineers. Then, the average of these preferences was calculated. It is highly possible that the average is "the most preferred level" of the ambience microphones.

3.7. Results and Analyses

As there were only seven preference choice data in the experiment described in Sections 3.5 and 3.6, means and confidence intervals of the five data (excluding the maximum and the minimum from the seven obtained) were calculated under the assumption of *t*-distribution. As a result, the score from five preference choice data was narrow enough to be regarded as reliable. It suggests that sound engineers might have selected the preferred sounds according to their common sense of the values.

3.7.1. Analysis of Front Microphone Arrays

Figures 3.1 to 3.3 show the ambience microphone array levels versus the front microphone arrays on each orchestra piece. The scores are indicated according to each ambience microphone array.

The distances from the "Orchestra Front Line" to the L/R microphones of three arrays of the 3O (3 Omni-directional microphones), the OM8 (Omni-directional microphone & figure of 8 microphone) and the 5C (5 cardioids) is between 1.2m and 1.6m each. The reverberation energies seem to be low in the 3O > the OM8 > the 5C because of the directivity of their microphones. Concerning the DT, its distance from the "Orchestra Front Line" is 0.5 m, where is closer to an orchestra than the other three arrays. The reverberation energy of the DT is lower than the 3O because of the different distance although both the DT and the 3O equally have the omni-directional microphones.

The scores of the ambience microphone arrays in Figures 3.1 to 3.3 indicate that the preferred sound levels are the 3O < the OM8 < the 5C and the 3O < the DT. It can be concluded that the lower the reverberation energy of the front microphone array is, the higher the preferred sound level of the ambience microphone reaches.

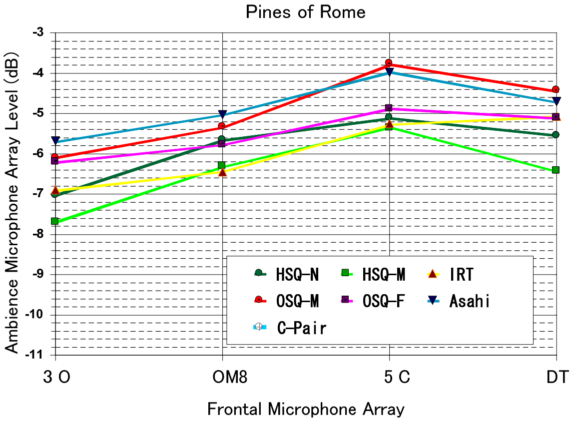

Fig.3.1 Preferred Sound Levels of Ambience Arrays on "Pines of Roma"

Figure 3.2 shows that the preferred sound level of the C-pair decreases to 3dB. The phenomena may be related to the surmise that the reproduction sounds from the rear speakers of the C-pair are theoretically 3dB louder than those of the other arrays as described in the footnote of Section 3.3.. The preferred sound levels of the ambience microphone arrays except the C-pair is within a deviation of 2dB.

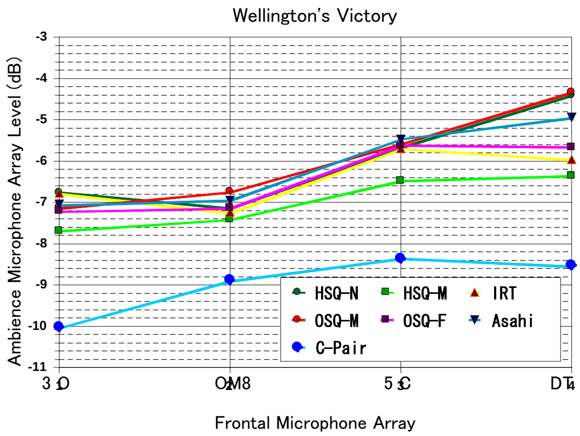

Fig.3.2 Preferred Sound Levels of Ambience Array on "Wellington's Victory "

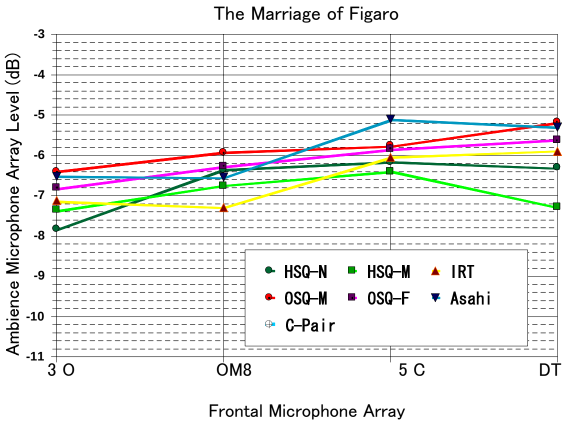

Fig.3.3 Preferred Sound Levels of Ambience Array on "The Marriage of Figaro"

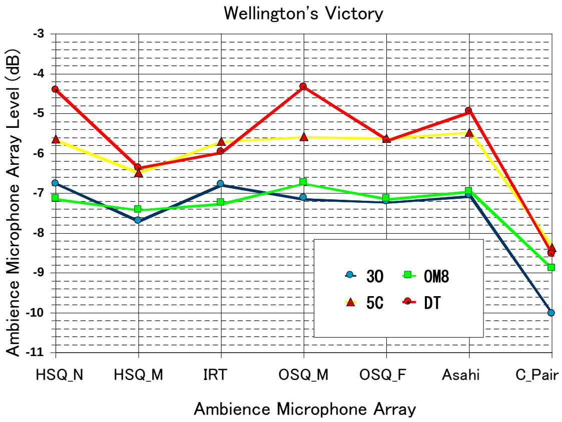

Fig.3.5 Preferred Sound Levels of Ambience Arrays on "Wellington's Victory"

3.7.2. Data of Ambience Microphone Arrays According to Front Microphone Arrays

Figures 3.4, to 3.7 show the ambience microphone array levels against the ambience microphone arrays on each orchestra piece. The scores are indicated according to each front microphone array. The preferred sound levels tend to be classified into two groups: the 5C and the DT; and the 3O and the OM8.

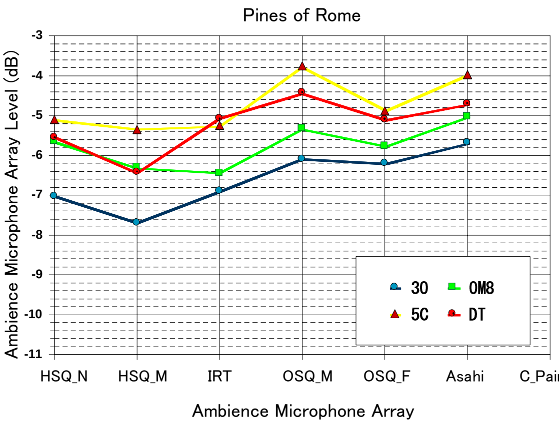

Fig.3.4 Preferred Sound Levels of Ambience Arrays on "Pine of Roma"

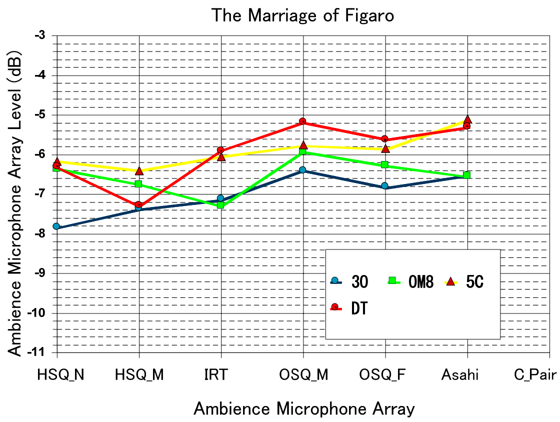

Fig.3.6 Preferred Sound Levels of Ambience Arrays on "The Marriage of Figaro"

3.7.3. Data of Ambience Microphone Arrays According to Orchestra Pieces

Figures 3.8 to 3.10 show the ambience microphone array levels against the ambience microphone arrays on the front microphone arrays. The scores are indicated according to each orchestra piece.

The deviation is relatively narrow. It is around 1.5dB except the scores of the Asahi on the OM8, the OSQ-M on the 5C and the HSQ-N on the DT. Therefore, it may be thought that the preferred sound levels will hardly be affected by any types of music pieces. In comparison among the HSQ-M, the IRT and the OSQ-M whose distances are same from the "Orchestra Front Line", the preferred sound levels tend to present the HSQ-M < the IRT < the OSQ-M.

The cover areas of the microphone array are dependent on the microphone directivities and are in the order of magnitude: HSQ-M < the IRT < the OSQ-M. The future study will be needed to find the related matter to the cover areas.

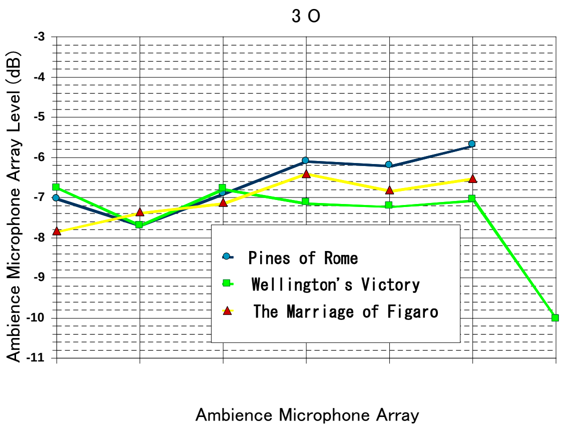

Fig.3.7 Preferred Sound Levels of Ambience Arrays on "3 O"

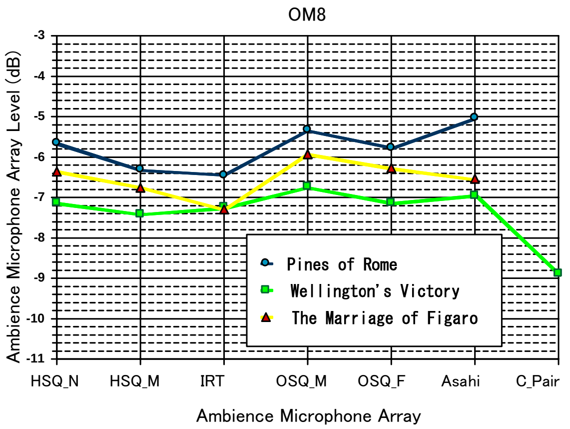

Fig.3.8 Preferred Sound Levels of Ambience Arrays on "OM8"

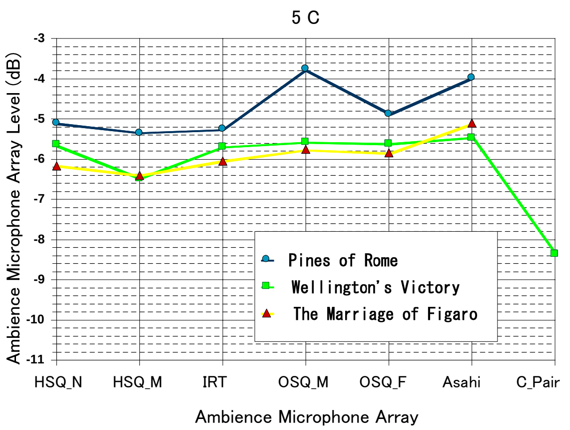

Fig.3.9 Preferred Sound Levels of Ambience Arrays on "5 C"

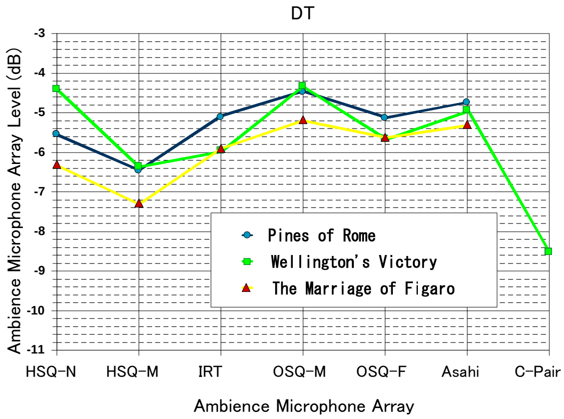

Fig.3.10 Preferred Sound Levels of Ambience Arrays on "DT"

3.8. Conclusions

As the deviation of the preferred sound levels per composition, as explained in 3.7.3, was relatively small, these orchestral elements such as abundant instrument solos, various tempos, dynamic developments, etc seem to have less effect to the mixings on the front microphone arrays and the ambience microphone arrays. Therefore, the preferred choice data among these three compositions could be calculated to find the average and the standard deviation. Please note that all the collected preferred scores of seven subjects in this experiment were processed using in this calculation formula. The excluded two data of the maximum and the minimum out of seven samples in 3.7 were not calculated; the fifteen samples in total were applied for the calculation. The Figure 3.11 shows that each yardstick is between the 95% of confidence intervals using t-distribution.

According to the preferred sound levels of the ambience microphone arrays in Figure 3.11, the front microphone arrays tend to be classified into two groups: the DT and the 5C; the 3O and the OM8. In the listening comparison of the sounds of the front microphone array, we could perceive the reverberation from the 5C array almost as equal as that of the DT, although the 5C was placed farther than the DT from the orchestra. This phenomenon of the 5C might have been caused by the attribute of the cardioid microphone array. As well, the reverberations of the 3O and the OM8 sounded almost equally.

Concerning these similar tendencies of reverberation, each sound engineer must have adjusted the levels of each ambience microphone array to gain equivalent reverberations "unconsciously".

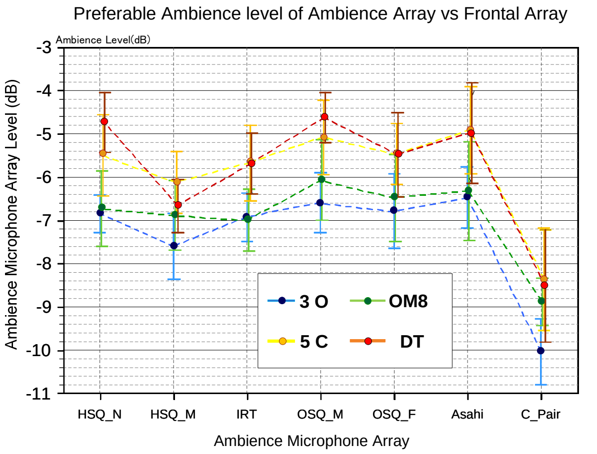

Fig.3.11 Preferred Sound Levels of Ambience Microphone Arrays vs Frontal Microphone Arrays

It is presumed that the most preferred sound level of the ambience microphone arrays except the C-pair to the front microphone arrays is -6 ±2dB. This value could be used as a starting level of the ambience microphone arrays upon surround sound recordings.

Table 3.2 Average of the Preferred Sound Levels between 95% Confidence Intervals of the Ambience Microphone Arrays via Front Microphone Arrays

Ambience \ Frontal		3O	DT	5C	OM8
HSQ-N	Ave.	-6.84	-4.73	-5.49	-6.71
	Con.int	0.43	0.69	0.95	0.87
HSQ-M	Ave.	-7.59	-6.66	-6.11	-6.88
	Con.int	0.77	0.62	0.72	0.81
IRT	Ave.	-6.92	-5.69	-5.67	-6.99
	Con.int	0.56	0.71	0.87	0.72
OSQ-M	Ave.	-6.58	-4.62	-5.07	-6.07
	Con.int	0.69	0.58	0.87	0.93
OSQ-F	Ave.	-6.77	-5.48	-5.46	-6.47
	Con.int	0.86	0.97	0.70	1.00
Asahi	Ave.	-6.46	-4.98	-4.90	-6.32
	Con.int	0.70	1.16	1.01	1.14
C.Pair	Ave.	-10.02	-8.52	-8.36	-8.88
	Con.int	0.76	1.30	1.18	0.55
Front of OSQ-M		—	—	—	-4.00

* Please note that the values in orange of the Table 3.2 were applied for the following evaluation tests. At the suggestion by the inventor of the OM8 about the combination between the ambience microphone arrays and the OM8, two front microphones of the OSQ-M as a pair of ambience microphones on the rare channels were used and -4dB of the preferred sound level were applied.

4. THE IMPULSE RESPONSE MEASUREMENT AND THE *C* - VALUE

4.1. Impulse Response Measurement

In the experiments of the recordings in the concert hall, the impulse response was also measured by a team at Faculty of Design, Kyushu University. The *C* - value acquired from impulse response measurement is an energy ratio of direct sounds to reverberation in logarithmic scale. The larger *C* - value shows the higher energy of the direct sounds. Here, the study was engaged to find whether the *C* - value would be related to the preferred sound levels. The signals of sound sources, called "Swept Sine Signals" were reproduced on the dodecahedron speakers in the concert hall, and then recorded over all microphones that had been used for the recordings of the orchestra. The *C* - values were computed with the collected data. The impulse response was measured at fifteen points where the dodecahedron speakers were placed as shown in Fiture4.1. The sound sources from the Point 9 on the center of the strings, 3 m from the center of a podium were used for this study.

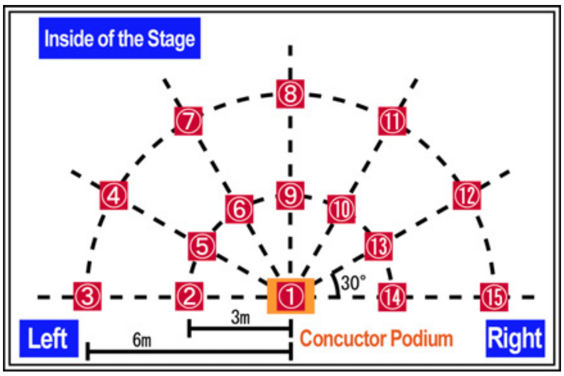

Fig.4.1 Placement of Dodecahedron Speakers in Concert Hall

4.2. The relations between the *C*- - value and the Preferred Mixing Levels of Ambience Microphone Arrays

The *C* - value can be calculated using equation (1) from the impulse response signal *p*.

$$\mathrm{C}_{te} = 10\log_{10}\frac{\int_0^{te} p^2(t)dt}{\int_{te}^{\infty} p^2(t)dt} \quad \text{...... (1)}$$

The down mixing of 5.0 surround sound signals to the mono signal provided for the *C* - values. According to the recommendation of ITU-BS.1770, it suggests that sound signal from each channel should be equally added up in the measurement of the loudness values of multi-channels. In this study, the impulse response per channel was multiplied by the coefficient = 1 and then down-mixed to make compounded impulse responses. The first sound wave form the Position 9 always reached to the center microphone of every array. Therefore, the arrival time of the first sound wave at the center channel was fixed to $t = 0$.

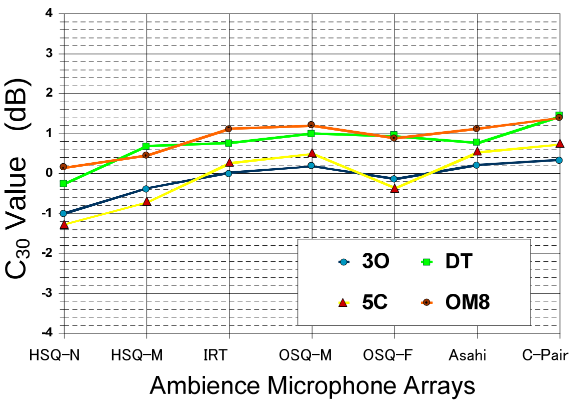

Fig.4.2. Score of C_{30} - value of Position 9

It is generally known that if it takes more than 30 ms for pulsive sound to be made one after another, then they sounds as two distinctive sounds. The t_e, following this principle, was fixed as 30 ms, and accordingly the *C* was described as the C_{30}. Next, the common method of dividing the frequency band by frequency for the calculation of the *C* - value was used in this study. The C_{30} - value in octave frequency according to the center frequency of 500Hz, 1kHz and 2kHz and the average of these three C_{30} - values were computed. The average of the C_{30} - values was defined as "the hypothetical *C - value*" for the surround mixings of this study.

4.3. Analysis

Figure 4.2 shows the average score of the C_{30} - value against the ambience microphone arrays. The hypothetical C_{30} - value is between -1.5dB and +1.5dB in all arrays. The average of 28 hypothetical C_{30} - values is 0.39dB within 95 % of the confidence intervals at a *t*-distribution is 0.27dB. Each graph of the front microphone arrays seems to be classified into of two groups: the DT and the OM8; and the 3O and the 5C

According to the graphs in Figure 4.2, the statistical dispersion of the hypothetical C_{30} - value will be concluded around 3dB at the result of a series of calculations. There is another interesting result. The most suitable distance from sound sources for microphones in hall-recordings is a "critical distance". The *C* - value = 0 is the critical distance. The hypothetical C_{30} - value in this study was close to 0.

At this point, we cannot draw a conclusion that the C_{30} - value ought be the constant. However, it is highly possible that the C_{30} - value will become the important index not only to evaluate surround sound mixings but also to find the preferred mixing levels.

5. SUMMARY

It is highly possible that the preferred sound levels of the ambience microphone arrays to the front microphone arrays in surround sound recordings are determined according to their common preferences of sound engineers. Additionally, it was shown that the C_{30} - values should be one of the most suitable indexes for mixings on the front and the ambience microphone arrays. However, the other elements such as the sound level balances between the front channels and the rear channels that have not been concerned in this experiment, might affect to the preferred sound levels. It is necessary to conduct the experiment to change the sound balances between the front and the rear microphones on the square arrays.

6. FUTURE EXPERIMENTS

In the next experiments, the psychological evaluation of each surround sound microphone technique will be examined, and the characteristics of each microphone array will be studied. Surround sound sources for these experiments had been mixed down with the preferred sound levels in Table 3.1 and the average of the preferred sound levels in Table 3.2.

The combination of the front microphone arrays and the ambience microphone arrays to be used for the future sound evaluation tests, on the bases of the results of several listening test, are as follows: the 3O and the IRT; the 5C and the HSQ-N; and the DT and the OSQ-F. Concerning the combination of the OM8 the two front microphones of the OSQ-M will be modified to an ambience pair as rear microphones. Furthermore, the future listening tests will be implemented with seven microphone arrays including additional five surround microphone arrays such as the Fukada Tree, the Double MS and the INA 5.

7. ADDRESS OF GRATITUDES

We greatly appreciate the time and effort of the following companies and people. This project has not been successfully performed without their generous cooperation.

We also thank to all sound engineers who had participated in our experiments.

○ Subventions

- Hoso Bunka Foundation, Inc.
- Panasonic AVC Networks Company
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- DVD Audio PROMOTION Conference
- Pioneer Corporation
- Dolby Laboratories Inc. Japan Branch
- TC Electronic A/S Japan Branch

○ Cooperation

- The Symphony Hall, Osaka
- Faculty of Design, Kyushu University
- Tokyo National University of Fine Arts and Music
- HEAVYMOON Inc.
- Miki Musical Instruments Co., Ltd.
- digidesign
- dts Japan Inc.
- MIXER'S LAB Co., Ltd.

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