Surround Sound Reference Disc

Surround Study Group of AES Japan Section

For all engineers and music fans

who love Surround Sound

Surround Study Group of AES Japan Section

Index

The sound sources included in this DVD are not intended to determine relative merits of the individual microphone array.

You are therefore requested to listen to features of each microphone array since the recording has been carried out under the limited conditions of a combination of an orchestra and the 'Symphony Hall'. By the time and volume constraints, some of the microphone arrays were left out in this disc. Your understanding shall be highly appreciated on these regards.

Surround Microphone Arrays Name List

English Name	Abbreviation
Decca Tree	DT
Fukada Tree	Fukada
Omni8	OM8
3Omni	3O
5Cardioid	5C
INA5	INA5(INA)
Double MS	DMS
Holophone H2Pro	HOLO(Holo)
Cardioid Pair	C-Pair
IRT Cross	IRT
Asahi Method	Asahi
Hamasaki Square Near	HSQ-N
Hamasaki Square Mid	HSQ-M
Omni Square Mid	OSQ-M
Omni Square Far	OSQ-F

Please refer to the '*Subjective Evaluation Test in Surround Sound Microphone Techniques: Part 1*' for details about the microphone trees and the microphone level balance of each microphone tree.

As far as 'Double MS' is concerned, the balance may differ from that of the recommended setting by Schoeps GmbH, because it was determined by the principal engineers using a simple 4-channel decoder, Schoeps recommends 5-channel decoding with its specific parameter values. For more details, please refer to the fore mentioned report.

On ‘Surround Sound Reference Disc’ release

Takeo Yamamoto,

In the world of acoustics, surround sound technologies are becoming a hot issue today. We are entering into a surround sound era, by which one can reproduce sound field, from mono/stereo sound era.

Since monophonic technology is what one records and plays back music by one ear, one can not enjoy it very much even with good sound quality. As one could record and playback music by both ears by the introduction of stereophonic technology, one has been able to enjoy music so that it has evolved rapidly.

On the other hand, though some trials about three or four channels were studied in view of post-stereophonic, so-called 5.1 multi-channel has been recognized as a next generation sound system.

However its evolution seems relatively slow to the author by such reasons as it needs development on multi-channel recording methods that can express well its effect in order to popularize multi-channel, and that a listening environment might require meticulous preparations to effectively reproduce multi-channel recordings.

Let me think about the cause in comparison with recording/reproduction of images. Monophonic is in a way a pinpoint acoustics, so we can not think of similar image counterpart. Stereophonic made it possible to record/reproduce music allocated on a straight line, but again we can not imagine what it’s like in the world of picture. In other words, it is because that it only can represent color or brightness change on a straight line.

With the advent of multi-channel technology, one has been able to at last record/playback music that would correspond to a two-dimensional image, but it is not like a two-dimensional upright picture; it can at best represent a quasi-two-dimensional lying, or fallen backward one.

It is because microphones as well as loudspeaker systems for playback are placed on a horizontal plane.

Therefore it isn't easy to record and/or playback acoustics or music closer to the original sound field using this scheme.

Development in video technology has made it possible to record/reproduce three-dimensional images.

To represent impressions under which we fell listening to music in a live sound field on site using multi-channel, there’s no other methods than to correctly record and playback acoustics of the music.

In this sense, multi-channel recording as well as reproduction methods have to be investigated sufficiently. I hope this DVD set will help advance the relevant studies and surround sound technology.

AES Japan Section Shinji Koyano

AES (Audio Engineering Society), founded in 1948, is an international organization whose prominent contribution to audio technologies is widely appreciated. It has over 12,000 specialists ranging from audio engineers and researchers to contents production engineers in a variety of fields and occupation from all over the world.

AES Japan Section was established in 1952. Since then it has been contributing to Japanese audio technologies.

It is holding every two years the Tokyo Convention since 1985 and it held the first Conference in 2006.

Surround Sound has been developed in concert with film audio since 'FuntaSound' of the early.

Today by the help of development in both high capacity media such as DVD and digital signal processing technology, its introduction in broadcast or music industries has been accelerating. Surround sound technology however is under development except for film audio. New technologies or methods relating to its recording and reproduction have been proposed.

Many discussions among audio engineers have been accumulating.

AES also has organized three international conferences on surround sound, including the 21st International Conference held in Germany in 2001.

Also in Japan, in the midst of situation where more and more digital broadcasting and/or package media start to include surround effects for their productions, hot discussions about its recording strategies have been under way in such occasions as Tokyo AES Convention, Conference, or independent study group meetings.

AES Japan Section started organizing a group in 2006 named 'AES Surround Study Group' proposed by Mr. Irimajiri of Mainichi Broadcasting Systems Inc.

Its objective is to provide sound sources to be served as references for all relevant engineers.

The project, headed by Mr. Irimajiri, is supported by such members from Kansai-based broadcasters, manufacturers, distributors, universities in terms of human, material and financial bases.

Osaka Philharmonic Orchestra greatly helped the project in sound source recordings taken place in September 2006. Since then, the project has been energetically working over a year on producing sound files and proceeding audio-visual tests.

This DVD is a comprehensive work on the subject by the AES Surround Study Group and we hope it can serve not only as 'reference sound sources' but also as 'guidelines for surround sound production' for engineers engaged in surround sound and for audiophiles as well.

We hope it can contribute to further development on surround technology.

The author would like to offer sincere appreciation for all people who have collaborated in the project.

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	Mixed by A.Fukada	/ Fukada Tree	
	Osaka Philharmonic Orchestra	/ Conduct: Shigeo Genda	
	1-The pine-trees of the villa Borghese		
	2-The pine-trees near the Catacomb		
Track 2:	Pines of Rome for symphonic-poem	/ Ottorino Respighi	(12:37)
	Mixed by A.Fukada	/ Fukada Tree	
	Osaka Philharmonic Orchestra	/ Conduct: Shigeo Genda	
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	Mixed by H.Irimajiri	/ Decca Tree & Omni Square Mid	
	Osaka Philharmonic Orchestra	/ Conduct: Shigeo Genda	
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	Mixed by H.Irimajiri	/ Decca Tree & Omni Square Mid	
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	Mixed by A.Fukada	/ Fukada Tree	
	Osaka Philharmonic Orchestra	/ Conduct: Shigeo Genda	
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	Mixed by H.Irimajiri	/ INA5	
	Osaka Philharmonic Orchestra	/ Conduct: Shigeo Genda	
	1- Battle		
	2- Victory Symphony		

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	Mixed by H.Nishida	/ 3Omni & Asahi Method
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	Mixed by T.Kamekawa	/ Omni8
	Osaka Philharmonic Orchestra	/ Conduct: Shigeo Genda
Track 9:	“The marriage of Figaro” Overture	/ Wolfgang Amadeus Mozart (4:25)
	Mixed by A.Fukada	/ Fukada Tree
	Osaka Philharmonic Orchestra	/ Conduct: Shigeo Genda
Track 10	“The marriage of Figaro” Overture	/ Wolfgang Amadeus Mozart (4:25)
	Mixed by H.Irimajiri	/ 3Omni & Omni Square Far
	Osaka Philharmonic Orchestra	/ Conduct: Shigeo Genda
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	Mixed by H.Nishida	/ Decca Tree & Asahi Method
	Organ: Seiko Katagiri	
Track 12:	Toccata and fugue in D Minor	/ Johann Sebastian Bach (9:10)
	Mixed by T.Kamekawa	/ 5Cardioid & Hamasaki Square Near
	Organ: Seiko Katagiri	
Track 13:	Heidenröslein D257, Op.3-3	/ Franz Peter Schubert (1:50)
	Mixed by H.Nishida	/ Decca Tree & Asahi Method
	Tenor : Hiroyuki Yoshida	/Piano : Toshiko Urabe
Track 14:	Heidenröslein D257, Op.3-3	/ Franz Peter Schubert (1:50)
	Mixed by T.Kamekawa	/ Omni8
	Tenor : Hiroyuki Yoshida	/Piano : Toshiko Urabe

Please note that the recording of “Pines of Rome" is divided into two tracks due to a limited MLP file capacity.

Profile -- Osaka Philharmonic Orchestra

Osaka Philharmonic Orchestra was founded by ASAHINA Takashi as 'Kansai Symphony Orchestra' in 1947. In 1950 it has become an incorporated body and in 1960 its name has changed to the current one.

ASAHINA Takashi had conducted Osaka Philharmonic Orchestra for 55 years since its establishment until 2001, and it had been recognized as a unique and fascinating orchestra.

In April 2003, OUE Eiji was inaugurated as a music director. The orchestra has also contracted with various famous conductors such as TOYAMA Shinji, TOYAMA Yuzo, WAKASUGI Hiroshi, AKIYAMA Kazuyoshi, TEZUKA Yukinori, and OTOMO Naoto.

The subscription concert series is held at The Symphony Hall twice each time, in total 20 a year. The orchestra is also contributing to the culture promotions not only in Osaka but in various areas in Japan.

Concert tours to Europe, North America, Korea, and Taiwan have been held several times, gaining high reputation at each country. Recordings have been done quite actively and this orchestra has released the most records and CDs in Japan. It was the 60th Anniversary for the orchestra in April 2007.

(June 2007)

Shigeo Genda (Conductor)

Graduated from the Tokyo Music College, Mr. Genda started conducting operas in 1986. He became a conductor of Shinsei Nikkyo Orchestra. He got a scholarship and studied in Vienna National Opera in 1990. In 1992 he conducted Plague National Opera during its Japan tour and made his debut in Plague Philharmonic Orchestra for its subscription concerts. In '93, 'Shinobu SATO (soprano) Recital' taken place on occasion of 'Plague Spring' was broadcasted in Europe.

He is now a permanent conductor for Kanagawa Philharmonic Orchestra and also a guest conductor for numerous orchestras in Japan and abroad.

Hiroyuki YOSHIDA, tenor

Graduated from Kunitachi College of Music and Opera postgraduate course in Tokyo College of Arts.

As a lilico leggero tenor, Mr.Yoshida has been receiving high reputation for his beautiful voice and lyrical and expressive techniques. He stayed in Rome as a trainee artist from the Agency for Cultural Affairs in 1991 and performed for many concerts in Italy.

Seiko KATAGIRI, organ

Graduated from Kobe College (organ major at the faculty of music). She received Hanna Gürick Suehiro Prize. She is an organist for both Kobe College and Kobe Catholic Church.

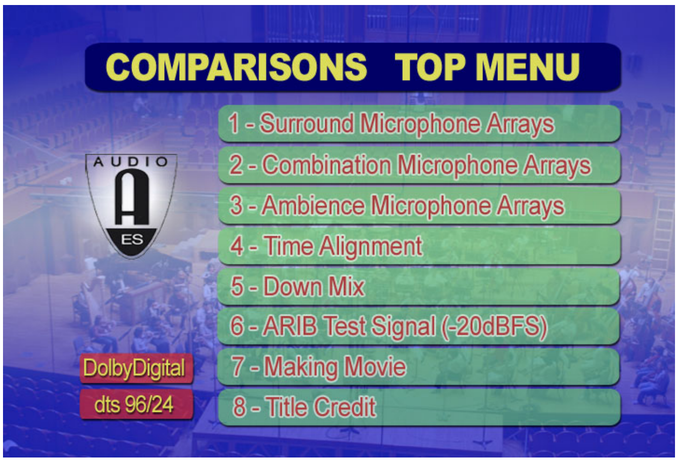
AES Surround Study Group of AES Japan Section

§1 Operation Manual

Thank you for purchasing this DVD set. It consists of three discs, namely Disc-1(DVD-Video: by which you may be able to compare different surround sound microphone arrangements), Disc-2(DVD-Audio: Demonstration disc suitable for you to ear-whiteness those surround systems) and Disc-3(CD-ROM: Documents).

Disc1 Comparison of Microphone Arrangements

You may be able to ear-whiteness differences in a variety of surround microphone arrangements. The following top menu will be shown upon activating playback; move the cursor and push return key to decide continue.



Menu 1） Comparison of surround microphone arrangements

1-1 Directions for use

This track contains eight different surround microphone arrangements for comparison purpose. Soundtrack for each microphone arrangement is recorded eight “angles”, therefore you can listen to the specific angle by switching its angle point during playback. What you are listening to as to the angle is shown on display.

Sound sources are digests edited from the recorded chunks by the AES Surround Study Group. There are 12 chapters. You may jump back and forth to any chapter by the chapter button. You can also switch angle even during repeating the identical chapter.

You may hear characteristics for each one of the microphone arrangements.

You may return to main menu by moving cursor on tom menu button.

1-2 About the microphone arrays

Eight types of microphone array are used. You may refer to each one of them for details in the '*Subjective Evaluation Test in Surround Sound Microphone Techniques: Part 1*'

i - Fukada Tree
ii - INA5
iii - Omni8
iv - Decca Tree + Omni Square Mid
v - 3Omni + Omni Square Mid
vi - 5Cardioids + Omni Square Mid
vii - Double MS
viii - Holophone H2-Pro

1-3 Comparative listening points in each chapter

①Pines of Rome 1st Movement

There appear no bass string instruments like contrabass. Therefore the recorded sound source may not well differentiate lower frequency characteristics for both omni-directional and unidirectional (cardioid) microphones for psychological assessment dependant to low frequency characters such as powerfulness. However you can compare breadth and depth impression and so on for each microphone array.

②Pines of Rome 2nd Movement ~off-stage trumpet

A far away trumpet is playing a melody. Sounds are coming somewhere from front side on the stage but they can not be identified exactly from where as they are in fact leaking. Compare the trumpet sound dispersed in the space.

③Pines of Rome　2nd Movement

The base of organ is played where music is showing its climax. The organ is situated on the wall in the deep end of the stage. Compare distance in the lows. As sample sound sources in Disc-2 use also LFE channel, you may assess how it sounds.

④Pines of Rome　3rd Movement

You may hear different solo musical instruments. Compare localization, blur and distant impression etc for such instruments in the front stage image.

⑤Pines of Rome　4th Movement

In the last part of this movement, banda is joining in the audience seats. Its localization as well as other instruments' separation, depth and width may be compared in the climax passage.

⑥Wellington's Victory British Side

A brass band playing a role of British troop is playing in the left backside of the stage. A snare drum is marching into the hall in the left and rear side of the seats from the outside hall. Compare how the sound differs at the very instant when the drum is entering the hall as well as localization and the breadth of the brass band behind the seats in the rear. In the latter part of this movement, string instruments on the stage are joining to play overlapping the brass: front- rear distance image as well as surrounding impression shall be compared.

⑦Wellington's Victory French Side

A brass band playing a role of French troop is coming to play in the right side on the second floor in the back side of the stage. Firstly, a snare drum is marching in the hall on the right side of the second floor from outside the hall. As in the same way in the previous chapter, compare how the sound differs at the very instant when the drum is entering the hall as well as localization and the distance impression of the brass band in the back on the stage. In the latter part of this movement, string instruments on the stage are joining to play overlapping the brass: front- rear distance image as well as separation shall be compared.

⑧Wellington's Victory Battle scene

Percussion instruments imitating British troop's firearms and a trumpet sounding the march in the left rear side in the audience seats are playing against the 2nd floor right based percussion instruments imitating French troop's firearms and a trumpet sounding the march. Sound, localization, front-rear distance impressions etc of the percussion instruments may be compared.

⑨Wellington's Victory 2nd movement

Music instruments are only the ones on the stage in this movement. Localization or breadth assessment for the string instruments may not be facile as the composition is too big in "Pines of Rome"; contrary to which, the 2nd movement illustrates well such details. Compare breadth in orchestra as well as localization and separation of the string instruments.

⑩The Marriage of Figaro

As the composition of the orchestra is made compact, this is a suitable piece to compare listening to the relative positions of each one of the instruments or sounds from different microphone arrays.

⑪Toccata and fugue in D Minor for organ

Organ placed on the stage rear wall is played. Because microphone arrays are positioned to best pick up orchestra sound, one may have such impression that organ sounds a little distant. Keeping these points in mind, compare listening to the difference in sound of each one of the microphone arrays as well as acoustics of the hall.

⑫Heidenröslein D257, Op.3-3

A singer and a piano are positioned so that Fukada Tree can pick up globally their perspective. Individual microphone arrays being placed so that the orchestra sounds best, we admit that not all of the arrays are placed at the most appropriate position. Because for such a composition relative distant impression greatly differs according to the real distance between an array and a singer with /piano position.

However the composition is so simple that one may easily hear the difference regarding to hall reverberation, breadth and tone timbre with each microphone array.

Menu 2) Comparison of combination arrays with front and ambience microphones

By this track recorded surround sounds are assessed for Decca Tree combined with one of the seven ambience microphone arrays.

2-1 Directions for use

Soundtrack for each microphone arrangement is recorded in seven "angles", therefore you can listen to the specific angle by pressing the 'angle button' during playback.

What you are listening to as to the angle is shown on display. Sound sources are digests edited from the recorded chunks. They consist of five chapters. You may jump back and forth to any chapter by the chapter button. You can also switch angle even during repeating the identical chapter.

You may be able to check characteristics for each one of the microphone arrangements.

You can return to 'main menu' by moving cursor on 'top menu' button.

2-2 Microphone array

Seven types of microphone array are the used. You may refer to each one of them for details in the *'Subjective Evaluation Test in Surround Sound Microphone Techniques: Part 1'.*

i - DT + HSQ-N
ii - DT + HSQ-M
iii - DT + IRT
iv - DT + OSQ-M
v - DT + OSQ-F
vi - DT + Asahi
vii - DT + OSQ-M Rear only

2-3 Comparative listening points in each chapter

①Pines of Rome 1st Movement

As in the previous comment, there appear no bass string instruments like contrabass. Therefore the recorded sound source may not well differentiate lower frequency characteristics for both omni-directional and unidirectional microphones. However you can compare acoustics for different ambiance microphone arrays.

②Pines of Rome4th Movement

In the last part of this movement, a banda (brass band) is joining in the audience seats. You may compare how an ambiance microphone array affects its localization.

③Wellington's Victory Battle scene

In this chapter, localization, tone quality (or timbre) and depth impression may be compared as to dialoguing (counterpoint) passage between percussions instruments based on the 1st floor rear seats' left (British troop) and the ones based on the 2nd floor right above the rear stage.

④The Marriage of Figaro

This track is suited to listening comparison relative to hall reverberation timbre.

⑤Heidenröslein D257, Op.3-3

This track also is good for comparing hall reverberation, breadth and timbre.。

Menu 3） Comparison of ambiance microphone arrays

3-1 Directions for use

Audio from each microphone arrangement being recorded on six angles, you can select one of these angles by pressing the 'Angle' button during playback. Display will show you what you are listening to. Sound sources are the digests edited from the recorded chunks. They consist of five chapters. You may jump back and forth to any chapter by the 'chapter' button. You can also switch angle even during repeating the identical chapter.

You may be able to check characteristics for each one of the microphone arrangements.

You can return to 'main menu' by moving cursor on 'top menu' button.

3-2 Microphone array

Six types of microphone array are the used. You may refer to each one of them for details in the *'Subjective Evaluation Test in Surround Sound Microphone Techniques: Part 1'.*

i - HSQ-N
ii - HSQ-M
iii - IRT
iv - OSQ-M
v - OSQ-F
vi - Asahi

3-3 Comparative listening points in each chapter

①Pines of Rome 1st Movement

Again, there appear no bass string instruments. Therefore the recorded sound source may mal differentiate lower frequency characteristics for both omni-directional and unidirectional microphones. However this movement is suited for you to compare acoustics (breadth or depth impressions etc.) for different ambiance microphone arrays.

②Pines of Rome4th Movement

In this movement, a banda is joining in the audience seats. You may compare how an ambiance microphone array affects its localization.

③Wellington's Victory Battle scene

Also in this chapter, localization, especially timbre impression may be heard differently as to dialoguing passage between percussions based on the 1st floor rear seats' left (British troop) and the ones based on the 2nd floor right above the rear stage.

④The Marriage of Figaro

⑤Heidenröslein D257, Op.3-3

These tracks may give you a pure example how timbre impressions differ on different ambiance microphone.

Menu 4) Comparison of cases where time alignment is applied to the microphone arrangements

Normally in the case where plural microphones are used to record an orchestra, some delays may arise between main microphones and spot microphones depending on their relative distances. In other words, sound recorded by the spot microphones is coming out from a loudspeaker earlier in playback before the identical sound reaches the main microphones.

By this fact, complex early reflection sounds to be recorded shall be disturbed by the sounds captured by the spot microphones. It could result in the loss of presence of the sounds to be recorded by the main microphones.

The idea that a delay inserted for the spot microphones can eventually cancel delays due to the said distance difference is called a 'time alignment' method.

This track compares those recordings without time alignment (later on referred to as TA) at all, TA applied to spot microphones alone, and TA applied to all microphones including ambiance ones. However relative positions of the ambiance microphones are identical to those of other recordings.

The way we adjust TA is this: we beat clappers for a spot microphone at the instrument's sound producing position and record the sounds beforehand.

Then we move waveform data looking on a DAW screen so that a spot microphone delay becomes zero at the Decca tree center microphone. In the same fashion, a set of Omni Square 4 channel data were moved so that the Omni Square Front side coincide with the Decca Tree L-R line.

(In reality, however, since too close TA might cause some coloration due to relevant waveforms' interference, appropriate TA should be decided by the careful listening. Although theoretical values were put in use, coloration was hardly heard due to this operation.)

4-1 Directions for use

By selecting 'Time Alignment' on 'Main Menu', the following 'Sub Menu' will show.

By highlighting your target sample using arrow keys (←→↑↓) on your remote controller, then push 'ENTER' to play the sample. If you leave the key operation as is, it will continue to play from the current sample to the bottom sample, then consecutively from the top one in the right column to the bottom.

It will return to 'Sub Menu' after playing the last sample.

Pressing 'Return to TOP MENU' button will bring you back to 'Top Menu'.

4-2 Comments on the contents

Sound sources are recorded using a Decca Tree for front array combined with a mid-distance Omni Square for ambiance array. Compare listening to the 1st and 4th movements from the 'Pines of Rome'.

i	-	Pines of Rome 1st Mov. ; Without Time Alignment (TA)
ii	-	Pines of Rome 1st Mov. ; Recording with spot microphones only
iii	-	Pines of Rome 1st Mov. ; With full TA
iv	-	Pines of Rome 4th Mov. ; Without TA
v	-	Pines of Rome 4th Mov. ; Recording with spot microphones only
vi	-	Pines of Rome 4th Mov. ; With full TA

You may be able to compare differences as to clearness, localization and depth of the individual instruments as well as front-rear spatial impression.

Menu 5) DownMix Compatibility

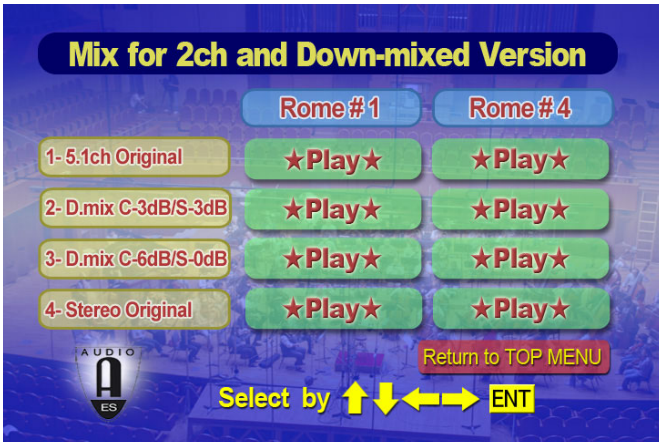
Any surround recordings, if played on a conventional stereo playback system, activate automatically a downmix circuit for stereo compatibility. The same is true for stereo broadcasting.

By the way, we have experienced a compatibility problem at transition from monophonic to stereophonic production.

This track assesses compatibility between surround and stereo recordings.

5-1 Directions for use

By selecting 'Down Mix' on 'Main Menu', the following 'Sub Menu' will show.



You may compare appropriately worked 'Original Stereo Mix' sources on this track.

The same part from the sound source is rendered by the following operations:

-Surround original mix

-Standard Down Mix, i.e. Center, Rear Mix Coefficients are both –3dB for Stereo Space Mixing

-Mix Coefficient values are set to Center –6dB, Rear 0dB for Stereo Space Mixing

and finally

-2-ch Stereo

By highlighting your target sample using arrow keys (←→↑↓) on your remote controller, then push 'ENTER' to play the sample. If you leave the key operation as is, it will continue to play from the left top sample to the bottom sample, then consecutively from the top one in the right column to the bottom.

Pressing 'Return to TOP MENU' button will bring you back to 'Top Menu'.

5-2 Comments on the contents

In the same fashion as 4), sound sources were made by combining recordings of Decca Tree for front and those of mid-distance Omni Square for ambiance.

Compare listening to the 1st and 4th movements from the 'Pines of Rome'.

i - Pines of Rome 1st Mov. ; 5.1ch Surround original mix
ii - Pines of Rome 1st Mov. ; Down Mix Center –3dB, Rear –3dB
iii - Pines of Rome 1st Mov. ; Down Mix Center –6dB, Rear 0dB
iv - Pines of Rome 1st Mov. ; 2ch Stereo original mix
v - Pines of Rome 4th Mov. ; 5.1ch Surround original mix
vi - Pines of Rome 4th Mov. ; Down Mix Center –3dB, Rear –3dB
vii - Pines of Rome 4th Mov. ; Down Mix Center –6dB, Rear 0dB
viii - Pines of Rome 4th Mov. ; 2ch Stereo original mix Pines of Rome

Generally applied Down Mix Coefficients (–3dB for both Center and Rear) were selected to keep the compatibility with movies. However since these coefficients loose Rear gain, unbalance may result (Rear becomes quieter) if meaningful sounds exist in Rear channel. Or, if sound components in L and R channels are same as the center ones, center may become louder. This is the reason why we tentatively applied Center -6dB, Rear 0dB coefficients. However since for on-the-spot broadcast quite often only the center channels were assigned at sports relay broadcasting etc. and that theoretically speaking –3dB gain equals the panpot center case, we sometimes hear a little less louder. Sound such as a shout of joy assigned to Rear channel, if reproduced in Front channels, it is in effect covered by the Front channel on-the spot commentary.

For terrestrial digital TV broadcasting, therefore, Rear coefficients have options of –6dB, -9dB, -∞ to reduce this effect.

We have to admit however that maintaining surround-stereo compatibility is tougher than mono-stereo compatibility. Compare how the recordings sound different keeping those points in mind.

Menu 6) Surround Test Signal

Test signals are generated according to the 'Surround Production Guideline (compiled by ARIB)'. Disc-3 contains those signals in 'Wave files': you may import them into your DAW for eventual use. (ARIB = Association of Radio Industries and Businesses)

Since the Test Signals on Disc-3 are encoded by 'Dolby Digital' or by 'dts', remind that lack of precision of about 0.2dB might result.

Upon selection of Menu6, the following 'Sub Menu' will show. By a click on 'Play', a sine wave is being outputted, please watch its level. It will return to 'Sub Menu' after playback.

Pressing 'Return to TOP MENU' button will bring you back to 'Top Menu'.

6-1 Directions for use

Selecting this item, the following 'sub Menu' will show. Please refer to commentary on test signals in Chapter 6 for the details.

Selecting this item, the following ‘sub Menu’ will show. Please refer to commentary on test signals in Chapter 6 for the details.

Features of test signals on Disc-3are:

-Reference sinewave signal (FFT sampling points could capture peaks)

-50Hz for LFE reference signal

-Signal level of -20dBFSrms for pink noise

To be exact, frequencies for 1kHz (Sinewave) and 50Hz are 1002.0Hz and 49.8Hz respectively.

The order of presentation: L→R→C→LFE→Ls→Rs→L&R→LRC&LFE→ALL

Moreover, you are advised to use pink noise for setting the reference level (adjust 0VU on a VU meter, refer to ‘Chapter 6’ for more details). If you can set a playback level of 79dB(C) for each channel by a Noise Meter (C curve, SLOW), your level fits exactly to the ARIB recommendations. As correct monitoring is the most important issue for surround production environment, we highly recommend using this signal.

Menu 7） Making Movie

You may witness some images for the experimental surround sound recordings.

Menu 8） Credit Titles

A list of surround sound recording project members and its staff will be shown.

Disc2 Samples for Reproduction Systems

This is a DVD-Audio disc. Please use any system that can play DVD-Audio discs.

The disc contains 12 demonstration versions (compressed by MLP from linear PCM 96kHz, 24bit) mixed down, by four engineers, from the recordings with main surround microphone arrays and the recordings with spot microphones.

There are cases where LFE channel is used. Since DVD-A specifies that reference level is set equal for all channels, we keep it for the present DVD-Audio disc. However for Dolby, dts, and terrestrial digital broadcast, a recording level for LFE is set lower by 10dB than that of other channels according to the norm of the loudspeaker setting. Recently, since professional mixing consoles are often provided with a LFE gain switch at monitor section, please let that level be 0dB. Please watch the playback level so as not to let it too loud, because LFE level is set to be its default +10dB for everyday mixing environment in TV studios. (Please note that the above level re-arrangement is not necessary for consumer AV amplifiers, as they internally process the level adjustment.)

These mix-down tracks are planned by such a specific request that one desires earnestly some surround sound sources for the reference of his or her playback system.

In Chapter 3, we are showing comments as 'Mixers Notes' written by the mix-down engineers themselves, so you are invited to refer to them while you are listening to their products.

If you could get a sense of mixer's intention as described in the relevant comments from your playback system, your system should work properly as a surround system.

We feel very flattered if you could enjoy hearing works and find different flavors depending on the engineer and think eventually of the music and the act of its 'recording'.

List of Music

Composer	Title	Time	Engineer	Microphone Array
1. O.Respighi	Pines of Rome -first part	9:58	Fukada	Fukada
2. O.Respighi	Pines of Rome -last part	12:37	Fukada	Fukada
3. O.Respighi	Pines of Rome -first part	9:58	Irimajiri	DT+OSQ-M
4. O.Respighi	Pines of Rome -last part	12:37	Irimajiri	DT+OSQ-M
5. L.Beethoven	Wellington's Victory	16:40	Fukada	Fukada
6. L.Beethoven	Wellington's Victory	16:40	Irimajiri	INA5
7. W.A.Mozart	Overture to 'The Marriage of Figaro'	4:25	Nishida	3O+Asahi
8. W.A.Mozart	Overture to 'The Marriage of Figaro'	4:25	Kamekawa	OM8
9. W.A.Mozart	Overture to 'The Marriage of Figaro'	4:25	Fukada	Fukada
10. W.A.Mozart	Overture to 'The Marriage of Figaro'	4:25	Irimajiri	3O+OSQ-F
11. J.S.Bach	Toccata and Fugue	9:10	Nishida	DT+Asahi
12. J.S.Bach	Toccata and Fugue	9:10	Kamekawa	5C+HSQ-N
13. F.Schubert	Heidenröslein	1:50	Nishida	DT+Asahi
14. F.Schubert	Heidenröslein	1:50	Kamekawa	OM8

Note: The 'Pines of Rome' is split into two tracks due to the file capacity's limitation. If it's continuously played, it fades out then fades in on the following track.

Disc-3 Documents and Surround Test Signal Audio files

This Disc (Disc-3) is a CD-ROM. WARNING: if it's played on a CD/DVD player, you may give damage to your loudspeaker systems. Therefore you are requested to use a PC for browse.

Disc-3 contains the following "pdf" files:

1) Surround Sound Reference Disc Operation Manual (This Manual)
2) AES Surround Recording Report (Japanese only)
3) Subjective Evaluation Test in Surround Sound Microphone Techniques:
   (Part 1. to 3. Japanese and English)

On this Disc, there are also audio signal files according to the 'standard surround test signals' defined by the '5.1ch Surround Program Production Technique Guideline' (ARIB).

You may find the following folders on the CD-ROM:

(1) 18dBFS_TEST_FILE_FOLDER
(2) 20dBFS_TEST_FILE_FOLDER
(3) PINKNOISE_FILE_FOLDER

WAVE files are stored in the folders. You may import and use them on your DAW. Please refer to the details in 'Descriptions on Test Signals' in Chapter 6.

§2 Comments on Titles and Interpretations

Hideo Irimajiri

1）Symphonic Poem ‘Pines of Rome’　Ottorino Respighi

Composed by an Italian composer Ottorino Respighi in December, 1924. Together with ‘Fountain of Rome’ and ‘Festivals of Rome’, they are called ‘Trilogy of Rome’

The ‘Pines of Rome’ is composed of four parts (i.e. movements) titled each. Each of them is represented by his favorite colorful orchestration.

Instrumentation

Three Flute (the 3rd flute or piccolo), two Oboes, English Horn, two Clarinets, two Bass Clarinets, two Fagots, Contra-Fagot, Horn, three Trumpets, Tuba, Timpani, Triangle, two Small Cymbals, Tambourine, Ratchet, Bass Drum, Tam-Tam, Harp, Glockenspiel, Celesta, Piano, Organ, Trumpet (behind the stage, used in the 2nd part), “Buccina” and Bugle (soprano-tenor-bass in B flat) two each (used in the 4th part. “Buccina” is a brass ancient Roman soldiers played, considered to be an ancestor of trombone. Replaced by four trumpets and two trombones in this recording.)

String instrument group (1stviolin, 2nd violin, viola, cello, contrabass), Voice recording for Nightingale (used in the 3rd part, on the original score the use of a Deutsche Gramophone record is assigned; this part is our original make.)

The First Movement　‘Pines of the Villa Borghese’

A scene where children play on a pine-lined street in Borghese park in Rome was depicted on a swift melody with loudly played horns and cheerful and spectacular orchestration.

This movement features no bass instruments such as contrabass. For this reason, we get light but lively impression out of it. To conduct a subjective test, our objective also, ‘no bass instruments’ means that one can reduce the eventual influence of microphone and/or loudspeaker characteristics which may influence evaluation results by their ‘low frequency range’. During listening, delicate movements of percussion can be heard clearly without the influence of low frequencies.

The Second Movement 'Pines near a catacomb'

A Horn starts singing a piece from Gregorian Chant in concert with bass instruments. A trumpet outside the stage is joining to play a hymnal melody sounded from far away.

Check point :How an out-of-stage trumpet can be heard from the bottom. LFE channel, seldom used for classical music, being used for the organ's lowest notes, it can also help check its usefulness.

The Third Movement 'Pines of the Janiculum'

The Janiculum hill is situated in southwestern part of Rome. Pines behind a full moon and a dreamy moonlight are depicted. Followed by a cadenza-like piano solo, sentimental clarinet solo, string solo, then sensual? oboe solo are performed one after another and finally nightingale' twittering is being heard.

Since this movement offers a consecutive solo instruments' performance, you may assess their localization.

Localization is not always felt alike by a listener and it is also true as it depends on the system's tuning and the recording method.

Therefore many cases are that other person's impression is not the one of your own. If you know the relative positions of the instruments by an attached allocation figure, you may be able to sense a slight difference in impression of separation depending on a playback system, especially its loudspeaker placement.

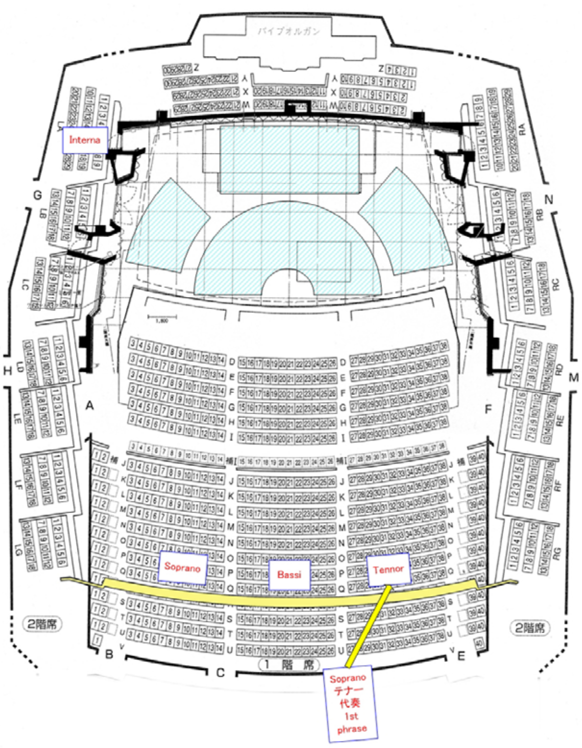
Moreover, a nightingale part was specially prepared for this recording and it is more or less TV program conscious. Therefore its twittering may sound too loud for those who enjoy music alone. This is because nightingale recording is not a real one. It is based on 4ch recordings of birds in Germany mixed with stereo recordings of a nightingale offered by an ORF sound engineer. It may be that they do not twit simultaneously.

Our intention is to make you imagine forest daybreak and feel such a scene or forest dimensions evenly spreading in the depth direction.

'Pines of the Appian Way'

Via Appia still exists today via which ancient Roman battalions marched. Clarinet plays a melody of military march, its loudness is escalating, then English Horn, representing troubadour's gossip, starts to play. An out-of-the-stage fanfare (brass) band, called 'banda', representing Roman battalion's fanfare, is approaching. While it's reaching a fortessimo, the orchestra and the banda are roaring toward the climax then energetically closing the piece.

In this movement, a brass (fanfare) band is joining behind the rear seats. Four trumpets and two trombones are divided into three groups. They are separately allocated at left, center and right behind the (audience) seats. Two trumpets (the first banda) on the right behind the seats are played first, then moved about 5 m towards the stage until the next performance. The second banda plays on the left behind the seats.



Check points:

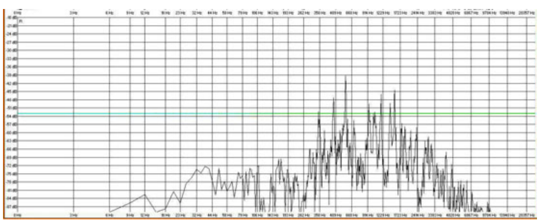
Distance impressions as well as localization of the rear brass may be confirmed.

At climax, brass on the stage and the banda are playing one another at the same loudness. You may experience that spectacle.

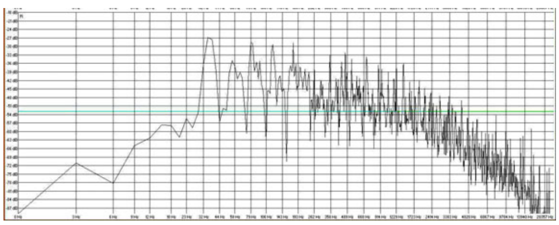
(In many cases, as we loose separation between the two brasses if this part is down-mixed, a prescription to this is to place the banda to the left and the on-stage brass to the right in case of 2-channel stereo.)

-1) Frequency characteristics of the musical titles and the microphones

Earlier, we have mentioned that there are no bass instruments in the 1st movement. We analyzed by FFT to see the frequency characteristics (spectra). The following figures show those of the 1st and the 4th movements.

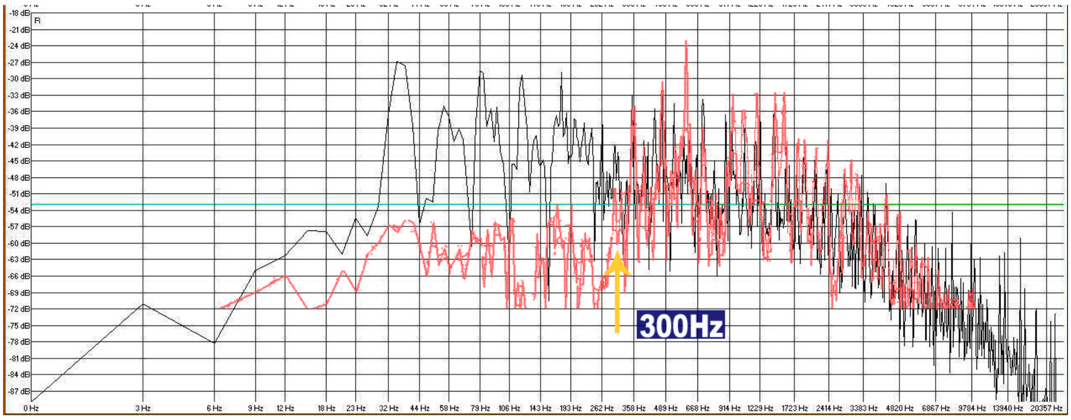


Spectrum on the beginning of the 1st movement



Spectrum on the last part of the 4th movement

Since an overall level of the 1st movement differs from that of the 4th movement, if we level high frequency spectra for two figures and overlap them, we get the following figure.

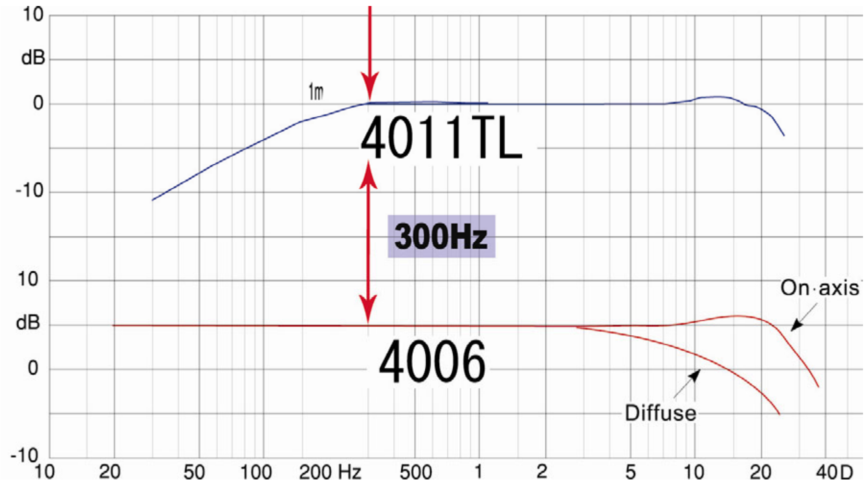


Spectra analysis on the 1st and the 4th movements from ‘Pines of Rome’

(Pink : 1st movement　White: 4th movement)

As seen on this figure, the energy below 300Hz is obviously low for the 1st movement. Therefore the 1st movement is suitable for the listening test in which one wants to exclude influence of lower frequencies as mentioned earlier.

For example, Decca Tree is composed of omni-directional microphones (DPA 4006). While INA5 array uses uni-directional (cardioid) microphones (DPA 4011).　Frequency response catalog data are shown in the following figures.



Frequency responses for DPA 4011TL and DPA 4006 microphones,

where DPA4011 response is taken at 1m on axis

As seen on the figure, DPA4011 response rolls off at 300Hz or lower. A music piece that has rich energy below 300Hz tells frequency domain difference between these two microphones. Therefore in a subjective test in which frequency characteristics may influence its result, the test results might vary depending on a type of microphone used.

In a case where we conduct a comparative listening between Decca Tree and INA5 array, a criterion word like 'richness', sensitive to low frequency components, evaluation results may be more influenced by the types of microphones than the allocation of microphone arrays. Even for such a case, if an original music piece, such as the 1st movement, contains less low frequency energy, a test is less sensitive to a type of microphone, therefore you may be able to assess the influence by a microphone array arrangement.

2) Symphony 'Wellington's Victory' Ludwig van Beethoven

Composed by Ludwig van Beethoven to praise the victory of the British troop led by duke Arthur Wellesley Willington who defeated French troop on June 21, 1913. It is also called a War Symphony.

The first public performance was a concert held in 1914. The program listed his two symphonies No.7 and 8 anticipating the 'War Symphony', but the main program was this last one which had been very popular at that time. Though it is labeled a symphony, this piece is not included in Beethoven's symphony category that has a sonata form by orchestra.

he musical title was composed of two parts: the first part is a revival of the Battle of Victoria, and the latter part is a spectacular triumphal song celebrating the victory of the British troop.

As Beethoven wrote himself, the manner of performance is

Together with an ordinary orchestra, two groups of 'Bass Drum as a canon, Ratchet as a rifle, Snare Drum, Bugle (signaling trumpet), and Wind Band' imitating the British and French troops, are placed as far as possible from the orchestra so that it tries to give spectacular effect. Furthermore, British and French troops play 'Rule Britannia' 'Marlborough March', respectively, and the British national anthem is used as a motif in the later part. (ref.: performance memo by Beethoven)

Instrumentation

Orchestra

Piccolo1, two Flutes, two Oboes, two Clarinets, two Fagots, Contra-Fagot, four Horns, four Trumpets, three Trombones, Timpani, Cymbal, Triangle, Bass Drum, five Strings (First Violin, Second Violin, Viola, Cello, Contrabass)

British Troop

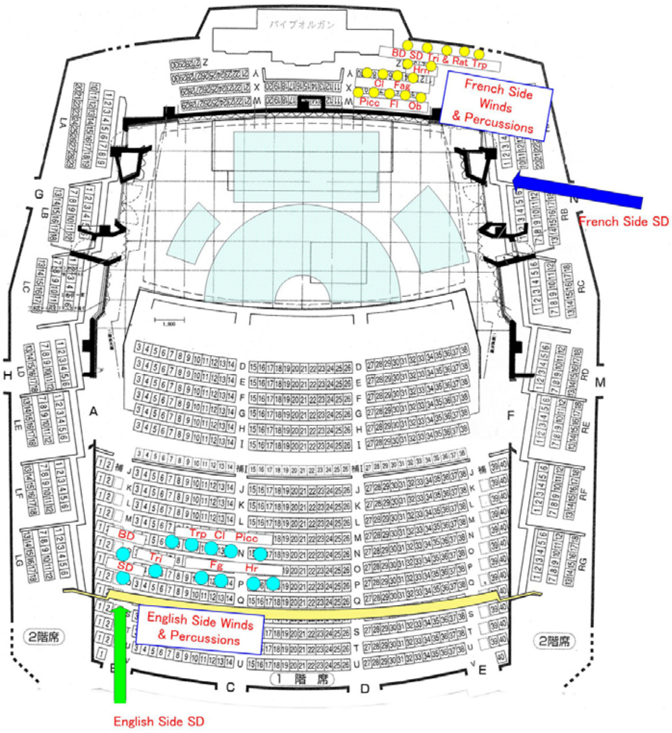
Piccolo, two Clarinets, two Fagots, two Horns, Trumpet, Triangle, Snare Drum (as many as possible), Cymbal, Bass Drum, Ratchet (as many as possible)

French Troop

Piccolo, two Flutes, two Oboes, two Clarinets, two Fagots, two Horns, two Trumpet, Triangle, Cymbal, Snare Drum, Bass Drum, Ratchet (as many as possible)

This piece is the largest orchestral music among all of his works. It even excels Symphony No.9 in orchestral composition point of view.

Performance Note



We can not perform at a concert unless we have all players as organized. As far as brass bands are concerned, it plays solos at the beginning, then later it should play the brass instrument part together with orchestra (according to Beethoven's performance memo). However if they do not overlap each other, we can doubly employ brass band players for the orchestra. Since this was a session recording, we could reduce the number of players by recording each parts and letting them move to other specified places. In such a case, one brass band was sufficient for the performance.

Often both troops are located at both wings or behind the stage. In our experiment, we placed a British troop on the left and behind the rear seats, and the French troop on the right above the hall organ to prove the merits of surround. If two are allocated on a diagonal line instead of straight front-rear line, it is because we wanted to keep a separation from the orchestra as much as possible and that both troops are heard separately left and right in a stereo down-mix.

Performance of a snare drum march in the beginning conventionally uses crescendo only, we have both British troop (green arrow) and French troop (blue arrow) walk in (drilling) from the hall entrances to enhance the stage effect.

First British troop drum march drills in from a left door behind the seats, then a marching trumpet plays a fanfare at the rear seats' left. A brass band stationed on the rear left in the seats plays a march 'Rule Britannia' then strings on the stage joins to play. Listeners get sandwiched by the rear brass band and the front string concert.

French troop drum march drills in from a front right door on the second floor, marching trumpet plays a fanfare. A brass band stationed in the front right seats on the second floor starts playing 'Marlborough march', strings join halfway.

Check point: The check point at this moment is whether or not listeners can feel distance between the orchestra and the brass bands as a function of the recording methods.

After marching trumpets exchange yells, comes battle section where British and French troops use a bass drum and a ratchet imitating a canon and rifles, two marching trumpets each, two snare drums for the British troop. It is worth listening to bass drums and ratchets furiously yelling each other between front and rear.

In the last part, imitating an English troop's fierce attack, marked 'Charge' on the score, snare drums are played one after another from the rear. In the course of time French troop's fire attacks are over, English troop canon delivers the coup de grace now and again while the orchestra is playing weakly 'Marlborough March' in minor.

The second movement is entitled the 'Victory Symphony'. This movement is played by the orchestra only. Halfway a motif from the British national anthem appears, develops itself, end up with a fugue; such a composition is also seen for Beethoven's later symphonies and proves evidently his matured orchestration.

Performance Memo written by Beethoven himself (summary)

1. To perform this piece, one needs two group of brass ensembles. They are divided into the British troop and the French troop. Beginning of the march shall be played by each group, half way or later they are played together with orchestra.

Therefore the orchestra should be as large as possible to keep a balance between them.

2. One needs two very large bass drums imitating cannon shots. (In Vienna, we use a 5-feet-square gran cassa for thunder.) British troop as well as French troop should be located at both sides of the audience and as far as possible from the orchestra. A conductor should stand at a position from where he can see both well. Especially a player who plays a cannon should stand not in the orchestra but at a far away place and of course should be a skilled player.
3. Instruments called a ratchet –imitating rifle fires- need to be placed at the both sides of the audience and closer to it. The performance should be at his discretion, except at the beginning of the piece and the part on the score where it is marked 'Charge'.
4. The British troop group places trumpets in E flat closer to a cannon and so does French troop group trumpets in C. One needs also four other trumpets for the orchestra.
5. At both sides, two snare drums need to play a drum march prior to play its corresponding march. In order to imply a troop's march, these need crescendo so that they are gradually approaching and conspicuously.
6. Tempo consideration:

    1) British troop's march is not very fast. (but livelier than the French counterpart) The French troop's march should be played more slowly. At and from the indication 'Charge', the tempo becomes gradually faster. The last 6/8 andante's tempo is not very fast.

    2) The introductory fanfare to the 'Victory Symphony' in the 2nd part, the corresponding tempo is not very fast.

    ' Victory Symphony' has a very active tempo. Last 3/8 tempo is not very fast. At the point where I assigned two players for each of the following four instruments, i.e. 1st & 2nd violin, viola and cello, they may be replaced by three or four players in a big hall.

7. In the performance, another conductor besides a principal conductor has to beat time for the whole piece. The constitution of the orchestra should be arranged according to the size and proportions of the hall.
8. In the 'Victory Symphony' there are likewise two brass bands throughout, but the second band does not play in the pianos and solos.

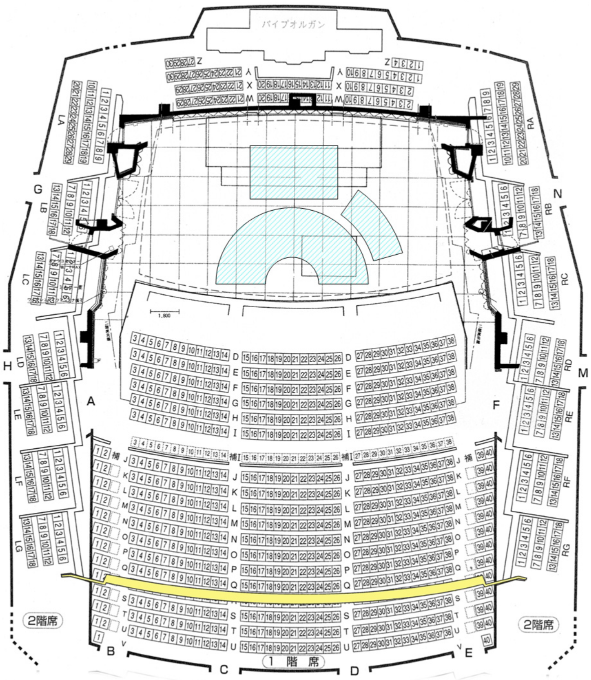
Vienna, December, 1815 Ludwig van Beethoven

3）Overture to Opera 'The Marriage of Figaro', Wolfgang Amadeus Mozart

Composed by Wolfgang Amadeus Mozart on its original satirical drama of French playwright Caron de Beaumarchais.

The opera consists of an overture and four acts in the style of opera bouffe. The work may be considered to be a modified version of two acts, since the first and the third acts have no regular finale.

Today the overture gained greatest popularity among the category and is often played alone at concerts.



Instrumentation

Two Flutes, two Oboes, two Clarinets, two Fagots, two Horns, two Trumpets, Timpani, five Strings (First Violin, Second Violin, Viola, Cello, Contrabass)

Checkpoint:

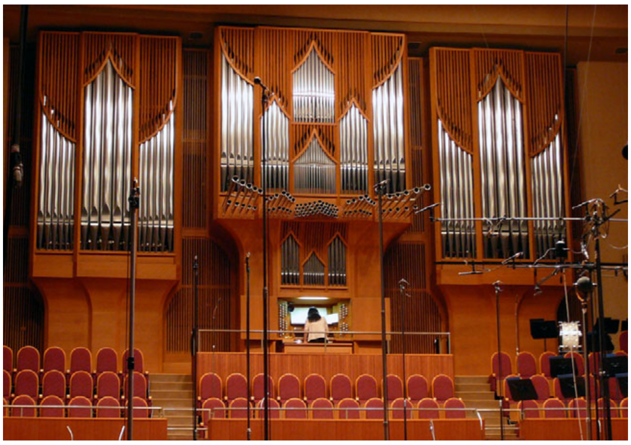
The overture to 'the Marriage of Figaro' offers us a most suitable sound source in order to compare listening to the balance among instruments or their tonality; in fact the overture has been often used for psycho-acoustic experiments.

Furthermore as we filed four tracks mixed by four audio engineers, you may be able to hear different musical expressions from the different approaches.

4）Toccata and Fugue in D minor, BWV 565' Johann Sebastian Bach

Organ music composed by Johann Sebastian Bach in ca.1706. This is the most famous one among his organ works. It is so dramatic and gorgeous. Often played on piano as well.

Toccata begins with a powerful melody so that it overwhelms listeners, develops itself by playing the chords on which organ sounds vividly and speedily over the entire piece.



Its motif is simple but full of novel ideas and change. Fugue on the other hand is very simple vis-à-vis other Bach's fugues, though it is written by four voices, it has a weaker counter-melody nature. It closes while the gorgeous sound is staying followed by a recapture of a part of Toccata

Its motif is simple but full of novel ideas and change. Fugue on the other hand is very simple vis-à-vis other Bach's fugues, though it is written by four voices, it has a weaker counter-melody nature. It closes while the gorgeous sound is staying followed by a recapture of a part of Toccata

Checkpoint:

Recording: Main microphones are being set to capture the orchestra, they are not allocated to the best positions for the organ. However hall acoustics and tonality may be comparatively assessed by these sample recordings. You may also check LFE channel effect in the case where it is used in a mix-down.

5) Heidenröslein' D257, Op.3-3 (1815)Op.3-3 Franz Peter Schubert

Composed by Franz Peter Schubert. Together with 'Erlking', 'Heidenröslein' is his masterpiece among his early works. A simple tune of a typical example of Lied style is merged well with a sensitive world of Goethe's poem.

Piano accompaniment plays simply the principal chords by right and left hand alternatively so that piano does not impair identity of a singer's voice.

Checkpoint:

The microphone setting was originally dedicated to the recording an orchestra. Under such circumstances, the condition is not at all optimal for the song recordings. Nevertheless the samples are best fit to hear the hall ambiance.

As to the surround recording, this piece offered us a good material of thought whether we should position a singer at a hard center or at a phantom center.

§3 Mixers Notes

In this chapter, the view and the technique of a mix are explained by the mixers themselves about demonstration of this surround experiment. You are invited to refer to their views for your listening experiences. Please note that a 'mixing note' will give information mainly for mixing engineers, and that a 'hearing note' will give listeners additional information about how a demo-mix will sound.

1） General statement for producers

Akira Fukada

There may be an opinion as such: setting main microphones at an optimum position for an orchestra recording and that's it! Is it really that simple?

Music has been a creation by a composer in a variety of historical backdrops. For its interpretation likewise, it is not unique either, as it may differ according to times. Composers wrote notes also taking a space into consideration in which how the music pieces they created might sound. In a space having a long reverberation time such like a church, if one plays loud and in staccato, all notes may merge together. So is a balance between low and high notes. Ability of an instrument, too. Even the keyboard instruments could not represent dynamics until a piano had been introduced. Today players play a music piece of that age on a piano and find eventually new expressions.

Imagine that two music pieces are to be played in a hall: Mozart's symphony and Stravinsky's 'Le Sacre du printemps'. Can I record these with best results using one main microphone array set at an appropriate position? Even if you could capture beautiful acoustic out of Mozart, it may sound too resounding and indistinct out of loud sounds of Stravinsky. In the same way, what if it is with a piano concerto? Microphones that are optimally set for an orchestra might be positioned just above a piano. Will it optimally balance the piano and the orchestra? How does the piano sound? Recording an orchestra in a hall would bring us such many issues. In the case of surround recording, extra care has to be taken into account as to not only acoustic impression but also a spatial image. Otherwise, the music shall be completely ruined

Let me take another example. If a music piece of Bach, that is implicitly based on the premise that it is not played in a large space like a concert hall, is to be recorded by surround with an acoustic image of a large space, I feel something wrong with the ‘music’. (It could be that kind of interpretation.)

What is the purpose of classical music recordings?

If you want to reproduce an orchestra on the stage physically just like a photograph, you will be inclined to use main microphones that would give you that image. Then, the choice may be a pair of directional microphones placed closer to one another such as ORTF style. Recently however, omni-directional microphone settings are widely used such as an AB method, I think it is because listeners want to ‘appreciate musical contents’ rather than a ‘pictured image’. In fact, an orchestra sound captured by a spaced microphone setting gives us wider image and scale. It is not a photographed miniature landscape garden so to speak. Then, what is a microphone setting like that can ‘appreciate musical contents’?

What is the meaning of ‘appreciate musical contents’ by the way?

Music pieces are created by composers; but their interpretation may differ a lot according to performers. It is a conductor that controls a whole orchestra. As a result, completed music is set freed into a hall space. Recording that sound is our job. Anyone who has experienced a recording may understand that the sound in a hall is not the one that is transferred via a microphone at all. The sound is very much subjective because what you hear is profoundly dependent of your consciousness. It is true that the audience in a concert hall is not only listening to music but also looking at players at the same time. They may know what instruments they play on the next phrases if they know that music piece. What shall we do if you want to represent it only by sound while they sense information available from hearing and vision with such and such consciousness? Unless we can spin ‘music’ out of sounds, people will not be moved by the recording. In other words, mixing engineer’s work is to re-create ‘music’ out of the sound captured by the microphones and nothing else. And in order to move someone’s soul, he or she has to understand that music sufficiently. It is by this sense that mixing is an art of expression.

This time, we are offered by a very rare experiment in which a variety of microphones and arrays are used to record simultaneously the same music pieces. Every microphone array has been invented by its developer with his philosophy. But the most important thing is how that music is expressed using these tools as I mentioned earlier. Combination of microphones and arrays may be, figuratively, cars and planers. I choose my favorite car to drive among the ones with different performance. The purpose is to drive comfortably and not that car itself. A sharp planer may be a very important tool, but it is what you make or what you can do with it that rally matters.

What we can get out of this gorgeous experiment is not the resultant soundscape, but it can rather be some sort of representation of diversified tools. It is the producer himself who could impress listeners with the music by using these tools.

2) Commentary to listeners

Akira Fukada

The view and the technique of a mix are explained about demonstration of this surround experiment.

1. Stereo recording vs. Surround recording

Orchestra is played in the front of the stage in a hall. The sound is resounding in a hall space. We listen to both the sound from the front on the stage and the resonant sound in a hall at the same time.

With 2-channel stereo recording, we reconstruct orchestral performances and hall resonance as a front stereo image. Though it is slightly different from the sound we actually listen to on site in a hall, carefully mixed work in 2-channel stereo may merit being called as an artistic recording.

While surround recording is a method by which we put orchestral performances and a hall resonant sound into what we hear. We do not intend to reproduce acoustic sound in a hall in a strict sense, rather to help listeners feel deeper emotion out of it, as we are now able to capture sounds from all directions which has been impossible by 2-channel recordings.

Music has been elaborated according to times. Composers have been creating their woks keeping a space into account in which they are played. The way how a harmony sounds or how the sound changes depending on its intensity may influence their works themselves. Pipe organ in a church, chamber music in a salon, large scale orchestral music, etc., every one of them has its space to be represented.

Surround recording is a method by which we can represent music together with its spatial sound, therefore it should result in those which will have deeply pursued musical meanings.

2. About the recording method

Surround recording has to capture not only orchestral performances precisely but also spatial reverberations and resounding appropriately. Therefore, a different approach from a stereo recording has to be taken care of as to main microphones.

At present, there is an international guideline in relation to surround sound reproduction environment and recording methods are being developed according to this guideline. The reproduction method is called an ITU-R layout (ITU-BS775-1). It consists of front 3 channels (L, C, R) and rear 2 channels (LS, RS) plus another channel for low end effect. Thanks to the front 3-channel layout we can precisely represent the position of musical instruments in the center direction the main microphones can capture. For this reason, three microphones for L, C and R are often used. Further, to pickup rear sound, it is known to date a method that capture sound of a hall using plural microphones or the one that simply uses two microphones and so on.

Usually main microphones are set above a conductor's head or at about the stage end at a height of 3 to 5 meters. It is the best position where sound out of each instrument of an orchestra blends together in a space. Such an optimum position is determined by calculated values and hearing though it depends on the types of microphone, orchestral composition, hall dimensions and shape and so on.

A microphone set closer to an individual musical instrument is called a 'spot microphone'. These are also used for recording. The balance of the overall orchestra sound and resounding shall be decided using the main microphone. If we want musically more loudness or clearness from particular instrument(s), we use spot microphones secondarily. Spot microphone is also used to clarify position of the sound source (localization), but the position is defined by electrically controlling the difference of sound volume. For instance, in the case of 2-channel stereo and if you want to localize a specific instrument to the left a little, you let the left volume up a bit vis-à-vis the right volume. If you want it completely left, you let the right volume zero. If you want center, you let the L and R volume equal.

In contrast, in the case of front 3-channel and if you want to localize a specific instrument to the left a little, its position is determined by the volume balance between L and C. If you want center, you take care of control of the center volume alone. For a delicate localization, left sound does not hear from right or vice versa with the surround recording.

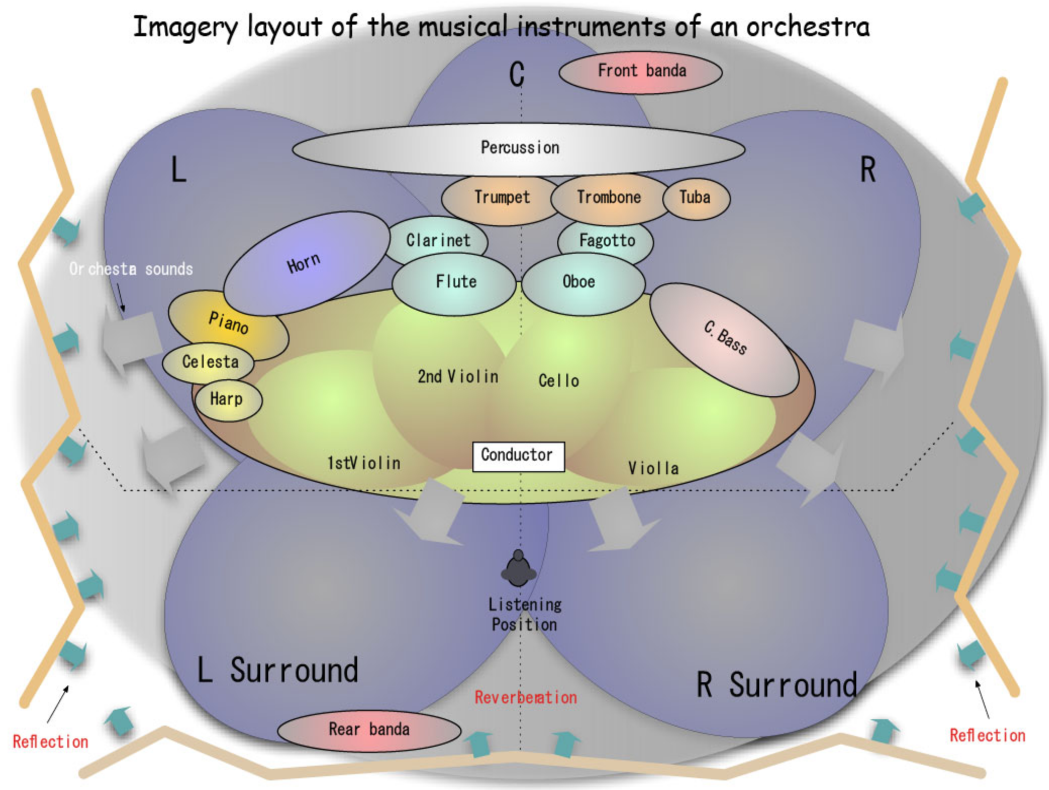
Sounds from left and right loudspeakers then become immune to the crosstalk effect from the opposite channel. It results in clear and high resolution sound.

This too is one of the advantages of the surround recording. Furthermore, with 2-channel stereo, the localization is influenced by the listening position, in other words it is determined by the difference of sound volume between right and left channel. With a surround setup, the sound played back by a center loudspeaker is heard always from the center wherever you are positioned.

3. Actual mixing

First thing I have to do is to check the condition under which the whole sound from an orchestra and hall acoustic are blended together. An orchestra consists of many musical instruments, but I perceive it as one instrument as a whole. The role of the main microphone is to capture the sound of this big instrument. By this main microphone's balance, the impression on the width and the depth of the orchestra and the overall sound are defined. You can of course record an orchestra by the main microphone alone, but in this case you should need time for a trial and error approach: not only the microphone position but also the layout of the musical instruments have to be arranged in order to have optimum balance to be inputted to the microphone. I can not use such an approach for this experiment.

Once the main microphone balance is fixed, I will understand sufficiently how the 'music' is performed. Then I have to make a judgment whether I add more information to the main microphone sound. If I need more detail from the string instrument group, or more definite position of a horn, or clearer sound from a fagot etc., I will add spot microphones. Too much spot microphones would spoil the orchestral soundscape (as 'one big instrument') because particular instruments are heard closer to the listener, therefore careful usage is primordial.



The purpose of mixing is to make 'music' livelier. Although music is written by the composer, the expression changes widely by players. In the case of orchestra, 'music' is re-created by a conductor who changes tempo, the strength of harmony, dynamics, etc.

The act of recording by which 'music' is captured using microphones has two objectives:

*1- Capture the music completed on site faithfully.

*2- Realize 'music' by the novel expressive medium, i.e. the recording technique.

The act of mixing that gives life to 'music' has more or less a secondary element. The judgment about the art of the completed 'music' is left to listeners.

The previous page illustrates the image how musical instruments are heard

As shown in the figure, the orchestral performances will be emitted in all over the hall. These sounds will spread in a hall in such forms as reflected or reverberated components. Overall sounds of the orchestra and the hall are caught by the main microphone. Of the orchestra there are the string instrument groups in the first front line, contrabass from center to right, harp, celesta, piano etc. to the left. In the next line, woodwinds: flute slightly left from the center, oboe a little right from the center, clarinet behind the flute, fagot behind the oboe, horn a bit left of the woodwinds. Trumpet almost in the center, trombone and tuba to its right.

In the far Please check carefully if you can perceive the depth of an individual musical instrument from strings to percussion. You may be able to hear reflections from the rear together with the sound of that instrument when it is played loud.

I hope this demonstration mix will eventually help you find something new. hest line, there are a wide range of instruments out of percussion groups from left to right and timpani almost in its center.

3) Track①② Respighi: 'Pines of Rome' 22:35 Fukada Tree

Please be reminded that this music piece is divided by the first half and the latter half due to the MLP file capacity's limitation. If it is played, it fades out and fades in at the splitting point.

Akira Fukada

Mixing Note

Fukada-Tree was used for the mix that I am accustomed to use. The reason is the following:

1. Main microphone

My view about the surround main microphone for an orchestra recording in a hall is based on the idea that is to record an overall sound of the orchestra and the hall as simply as possible.

In the case of stereo recording, omni-directional microphones are arranged as a main microphone to Critical Distance of a hall in many cases. But even it is the surround recording, there is no difference in the view of arranging a main microphone in the optimal position of the hall ambiance.

For example, suppose there is an optimal place that maintained the balance between orchestra sound and a hall ambiance. If there is an ideal one-point microphone that can capture the whole sound, one may think that we can reproduce that sound space appropriately.

Fukada-Tree has been devised as a big one-point microphone in principle.

I used five microphones in a studio space, (as I thought at least five microphones were necessary for 5 channels) and evaluated the recordings by varying the directivity on each one. I found that unless separation was kept to some extent between the sound inputted to a front microphone and that of a rear microphone, I got less localization and loss of clearness.

Therefore I formed an array of seven microphones: five cardioid microphones to keep separation between front and rear and that between the front three microphones, and two omni-directional microphones set to the array's left and right to compensate lateral relationship to that we are insensitive and to capture ambiance from the front.(AES Preprint 4540 1997 New York)

Omni-directional microphones are only used for front ambiance.

At beginning, a triangular array of three microphones had been applied, but later I arranged a center microphone to be about 20 cm ahead from the L-R line to improve front localization. Distance from a front microphone group to a rear group is 3.0 m or smaller.

For an actual recording, I opt for an approach (not omni- or directional type) to get more appropriate separation (by adapting an acoustic pressure equalizer (Kugel) which adds directivity to high frequency for example) or an omni-directional microphone that has high frequency directivity. In short, I am treating these seven microphones as one without labeling front or rear microphones by this approach.

There is another thought by which ambiance microphones are to be added to the front microphones, but I do not take this approach because I think every microphone has a relativity, in other words sound components of a rear ambiance microphone has too much influence upon the sound of the front microphones one can not ignore.

2. About microphone arrangement

As we had to install a wide range of main microphones for this experiment, it was impossible to set them in the best position because there were no specific and theoretical values available that would give a Critical Distance. It was not possible either to adjust its best position by hearing. Therefore I can not say with confidence that this microphone position is the best one.

As a result, microphone position is the following:
Height: 340 cm, Distance between L and C: 140 cm, Center microphone position: 24 cm ahead from the center of L-R line, Angle of L and R:+/-30 degrees.

3. Actual mixing

It seems that the sound of orchestra is made from harmony of synthesis of the individual musical instrument sound emitted into space. It becomes a particular orchestra sound. Since a simple synthesis of individual spot microphones can not catch such sound, we have to use a distanced microphone setting for an ensemble recording. It is what we call a main microphone.

The main microphone can capture information of an orchestra for the most part in terms of overall sound image, width, depth etc. But that is not enough. There must be some instruments buried that should be heard originally or those that need clearer sounds. A work then becomes necessary to raise the level of completion as 'music' by complementing these sounds in the course of mixing. It is the spot microphones that are used for this event

3-1. Spot microphone

Sound of a musical instrument itself in addition to the sound of the surrounding instruments are inputted into a spot microphone. Moreover, the sound from back is influenced by the back characteristic of the microphone.

A spot microphone is used to balance the sound of a musical instrument and that of others or to make the sound balance which I think are necessary for the music without modifying the fundamental sound signal caught by the main microphone.

Moreover, two sounds may be heard when the spot microphone level is raised, because the distance differs between the musical instrument and the main microphone or between the spot microphone and the musical instrument. The delay of appropriate length may be inserted in a spot microphone channel. Use of the delay to a spot microphone needs special attention, since the microphone is not only picking up the sound of the target musical instrument alone but also the surrounding sound from other instruments. The sound may degrade as a result. To minimize the influence by proximity arrangement and leakage from other musical instruments, I use some processing by EQ.

This time, delays were inserted for percussion, woodwind and other adjacent groups vis-à-vis the main microphone. Spot microphones for individual instruments were EQ'd a little. I do not use LR PAN to form a phantom image in this mix. I used LCR PAN. The phantom image is formed between L and C, and C and R in LCR PAN as well. However as the crosstalk of L and R may decrease, the localization and resolution improve.

3-2. Processing of rear sound source

In the experiment, there is a music piece in which some instruments located in the rear of a hall were playing. When that sound was monitored only with the main microphone, spatial relationship was indefinite.

It is hard to believe that the sound source from rear is keeping the sound quality or it has clear directivity although actually the judgment is influenced by the listening position in a concert hall. In fact, main microphone might have caught that situation appropriately.

However, I needed spot microphones for the rear musical instruments in order to complete recording work as an art of expression in pursuit of the realism. Because of the distance between the spot microphone and the main microphone, I had to insert delay.

4. Mixing procedure for an individual music piece

As earlier mentioned, mixing is not a faithful reproduction of a musical performance but rather an art of representation. My basic approach to mixing on any music piece is identical as the following steps:

1) Before anything, I listen to main microphones' sound and decide their position by which I get the best balance on 5 channels. I let this a basic balance of the main microphones.
2) Check precisely a musical performance to on the score and understand what a conductor tries to realize with the orchestra.
3) Check a balance and tonality on the musical instruments from spot microphone assisted groups such as strings, woodwinds, percussions, harp, piano and celesta.
4) Mix spot microphones' sound with that of the main microphones and decide necessary delays.

5) See if I can use main microphones only for a most part of the score or check if I need some spot microphones. Let me call that microphone balance a 'basic balance for expression'.

6) If a spot microphone is added, check tonal or acoustic change by ON and OFF the spot microphone. Introduce EQ or Reverb if necessary.

If reverb is added to a spot microphone, it is because I can sense the acoustic change even though I get a good balance. In such a case, I use reverb with which acoustic of the spot microphone gets as close to that of main microphone as possible.

7) Based on the 'basic balance for expression', if I want to use more microphones in each movement, I use them accordingly. But I do not use microphones that are not necessary as far as possible. This is because too many microphones would make the sound more complicated, as a result loose clarity.

According to the above approach, mixing has been conducted on every music piece. There is no difference in the approach to an individual music piece, however I would like to mention some features as follows.

When I am producing surround music, I do not use LFE channel myself, especially if it is a classical one. As I understand, a classical music does not need any effect at all, because LFE is a low frequency effect channel.

But for this piece I utilized an LFE channel other than the effect purpose for a trial. Specifically it is where an organ is coming in to play that has to be enhanced to add up more of a scale to this piece.

It is also true that this type of LFE application can be dangerous because it is very much influenced by the playback environment. I do this for a challenge.

Another point is a control of a dynamic range. Do you do it for a classical music mixing too? The answer is yes. For an academic recording believer, he or she may think a recording without any modification is the best. I do a dynamic range control by just the same reason as 'then, why did you use a spot microphone to boost for certain phrases?'

For a broadcasting mixing, we are often obliged to compress a dynamic range, however in this case it is the way how a music is listened to by our ears alone that is in question. In this music piece, there were cases where a very low level passage was slightly boosted its level.

Finally, there is a scene of birds ' twittering. Please note that the mix of this part is a post-production by Mr. Irimajiri due to time constraints.

4）Track③④ Respighi:‘Pines of Rome’ 22:35 DT+OSQ-M

Please be reminded that this music piece is divided by the first half and the latter half due to the MLP file capacity's limitation. If it is played, it fades out and fades in at the splitting point.

Hideo Irimajiri

-1) Mixing Note

Main microphones: Decca Tree ; Ambiance microphones: Omni Square (OSQ-M)

They are all omni-directional. In my case, quite often I use a Decca Tree. It features a center microphone protruding toward an orchestra; it looks like a triangular shape at each apex of which hung a microphone. Decca tree is originally devised for stereo recording. For stereo, signals from the center microphone are electrically divided between L and R channels. Note that although the type of the microphone is omni-directional, it has non-negligible directivity at high frequency domain.

In many cases, Decca Tree is set up by the cross point of the T shaped bar at 4m higher position over the head of a conductor. In such a case, LR microphones are on a line that ties a violin concert master and a viola major player. Angle of L and R: 180 degrees. Microphones are distant by 1m from the cross point of T. Typically, studio uses such a setup.

This time the microphone tree shifted several tens of centimeters closer to audience seats. There are some reasons to this shift. A 4m height setting can not catch brilliant sounds from the brass instruments due to their directivity, therefore we lowered the height by 80cm (H=3.2m) and moved the tree backward to avoid being too close to strings instead. For that matter, LR microphones were inclined by about 30 degrees toward the center, i.e. aiming at centers of violins and violas. Of course, these arrangements are to be done on a case-by-case basis according to instruments' allocation, stage step shape etc.

We used DPA-4006 microphones whose frequency response rises slightly at high frequencies. (With a diffuse grid put on a microphone, we can boost a few decibels at around 12kHz.)

On the other hand, high frequency boost will disappear with the increasing incident angle of the sound waves to a microphone capsule by about 20~30 degrees (with a diffuse grid, several tens of degrees) due to high frequency directivity. If you want to emphasize highs, you may direct your microphone axis toward that direction or in the opposite case, you may off-axis the microphone. This way you can adjust the tone timbre as you wish.

For our recordings, we used a conventional free-field grid for every microphone and the microphones' elevation angle was set to be 0 degree. As a result, a microphone directivity axis shift was about 30 degrees viewed from woodwinds. Since it was about several tens of degrees viewed from strings, highs were a little suppressed (lowered). Although we set the microphones' elevation angle to be zero degree with a view to keep a tone balance between strings and woodwinds, it is a problem of taste so that you can decide it on a trial-and- error basis.

In general, a stereo recording that uses two omni-directional microphones is called 'A-B method': phase difference, i.e. time difference, between the two gives you a localization image. It is known however that such a delay that'd induce localization image is not greater than about 5 ms.

In the case of Decca Tree (two microphones are 2 m apart), though we know it also depends on a distance between a sound source and microphones, let us look at its L-R localization.

Arrival time difference between the two microphones gets greater than 5 ms from a sound source situated at 45 degrees or more from the Decca Tree center. In such a case, the source would localize itself to almost either left or right due to HASS effect.

If two microphones are separated more than 2 m, sounds from center loose correlation between audio signals in the mid-high range of the two microphones, then we will loose center localization. It is a case so called 'Center loss'. You may shorten LR microphone distance to avoid 'center loss', but you will loose width. Generally admitted and well-balanced distance is around 60 cm. 'A-B method' is also called as phase difference recording method. As the name implies, since this method can reproduce spatial phase information, it is not difficult to represent distance or width impressions though localization is obscure.

To improve the center localization, a method was thought of to allocate a center microphone and localize it at a phantom center. This way a Decca Tree with three microphones or otherwise called as '3Omni' method being a one that can improve width and center localization at the same time, localization in-between L and C(enter) or that of R and C can not be better by the same mechanism and therefore remains to be blurred. It is known that the said localization would merge to either L, C or R. (A method abbreviated as '5C' eventually added another two mid microphones between L-C and C-R to remedy the situation.)

Although Decca Tree was originally developed for stereo recording, since it has three microphones, a front 3-Channel reproduction was devised taking advantage of the three microphones: a center microphone is connected to a center loudspeaker system. For convenience sake, let me call it hereafter a 'Front Mic Array'.

By the way, as I pointed out that in-between localization between L,C and R is not stable even with Decca Tree, a new idea is installed to put auxiliary(spot) microphones and insert delays to enhance center localization. Panpot is in a way a forced localization control by means of sound volume difference so that you can relatively easily move the localization between Land R loudspeakers. But that sound image has no spatiality so that it can not mix well with a main microphone's natural environment.

It is because that if you add a spot microphone in the A-B system, sounds from a spot microphone shall be played back prior to sound arrival from the main microphones due to distance difference between spot and main microphones. That value is no more than 5ms as earlier mentioned but rather of more than 30ms in case of percussions playing in the deepest corner on stage or of a piano in the leftmost corner. Under these circumstances, additional spot microphones may make image move by panpotting but spatiality rich in phase information as well as depth information will be lost by masking. In the worst case, one attack on a percussion instrument may be heard twice.

Delay is introduced for the spot microphones. In mid-80's I have been using this method ever since Rolland's inexpensive digital delay equipment product had been marketed. I still remember struggling with degradation of sound quality and its S/N ratio at that time. Today, once the sound data are imported in a DAW, all you have to do is the region move to realize delay. We call such a method of delay insertion and normalization in time domain 'Time Alignment'.

There are some disadvantages in this method. It is impossible to align two data perfectly even by introducing a delay. Also Players may move: short delays may overlap together and trigger coloration. In the worst case, it may sound like one that was put through a chorus machine. In our experiments, all of the (about 20) spot microphones were time aligned. Afterwards, we panned the spot sound to the point where we heard the instrument from the main microphone (and importantly not to the point where the spot microphone should have picked up the instrument) by listening carefully the main microphone.

Moreover, when we use main and ambiance microphones for a surround recording, an issue will come up as to whether time alignment is necessary or not for them. In the case where time alignment is needed, alignment must be adjusted between main microphones and front microphones in a square array. In such a case, you can only move the region of a square array toward an advance direction on a DAW. However in live recording, it is unrealistic to insert a delay for main microphones.

You are requested to refer to DVD-1 for comparison.

For this mix-down, I did not perform time alignment between main and ambiance microphones. It is because I felt more spatiality without the time alignment. If time alignment is done, I had clearer impression but a space seemed to have shrunk a bit. Please note that microphones for a banda were time aligned with microphones of the ambiance microphone array.

As far as 'balance' is concerned, as Mr. Fukada also wrote, a mixer need to read scores, communicate with players and decide how to mix.

Broadcast live recording differs much from a session recording for a CD, for instance.

It is a well known practice that for a session recording, a music piece is divided into pieces each of which is recorded in a plural takes and that OK takes or parts are assembled afterward. It is because it is practically and economically impossible for a player to play a whole piece for takes endlessly until he or she finishes playing without any flaw at all. It is also true that a CD, by its nature, is made by such ideas that it is listened repeatedly and that it has to be scientifically correct.

Much has to be discussed as to take either a live performance powerfulness or preciseness in a session; but in a CD session recording, because quite often a director takes a role of a mixing engineer too, he or she is requested to be able to read scores exhaustively as well as to point out performance flaws.

Above descriptions are some of my approaches for recordings.

-2) Hearing Note

In the 1st movement, spot microphones were applied to a piano, a harp and a celesta. They all are located on the extreme left of a stage. These instruments are hard to be heard with good separation if played except in solo and are very distant from the main microphone. Gains for the spot microphones were set sufficiently high. Therefore there may be the cases where their sound image deviates a bit from that of the main microphone resulting in inhomogeneous impression. This may divide opinions whether you like it or not. Inversely, if you could sense in omogeneity out of it, your system might be well tuned.

Furthermore, as a piano was played fortissimo in its high notes, it created instrumental noise associated with the action. You may hear these noises as well from an elaborate playback system.

If there are instruments having a relatively similar tone timbre in one direction, and that their localization got separated, you can see them move clearly; but if their localization were not separated, you may be unable to tell what is playing any more. This is what a mixing engineer has to take care of.

Small percussion instruments playing in the back are placed from left to right: cymbal (leftmost), ratchet (left), snare drum (slightly right), triangle, tambourine (slightly right), glockenspiel (on their right), Chinese gong (rightmost). In fortissimo part, these instruments tend to blend together. Therefore if you hear them with good separation, your playback system must be a decent one.

In the 2nd movement, you can hear solos of some instruments, halfway, an off-stage trumpet joins to play from the point on the score marked 'Interna'. The trumpeter stays on a staircase landing situated on deep left from the backstage. He is watching a conductor on a TV monitor. Trumpet echoes around the staircase and it should be heard as a sound drifting on the air coming from the backstage. In a surround listening environment, you may hear it with such impression as the trumpet locates itself vaguely in front but one can hardly identify from where. In fact, there is a door for players on the deep left in the backstage. It is left open so that the volume must be higher in the left due to sound leak. It may be of some interest to you if you can hear so.

In the last part of the 2nd movement, at climax, lowest notes of an organ (assigning 8,16,32 feet pipes) are played. If you hear them played on a seated position you may not able to catch them. Microphones were used for a LFE channel so that you can assess its usefulness. If you compare the part of bass continuo during the passage where organ is playing diminuendo after climax, you may get more low end with the LFE.

In the 3rd movement, many solo instruments are used. Especially regarding solos by wood instruments, among which two instruments are sitting closer to one another like a clarinet and a fagot, or a flute and an oboe, you are asked to listen to if such instruments are localized solidly. The mix you may hear is a result of meeting the localized image from the main microphone and that from the panned image. Please refer to DVD-1 as to the localization of a recording by Decca Tree alone. In DVD-1 you may find sound sources from a combination microphone array of Decca Tree and Omni Square in the chapter: 'comparison of listening'.

In the middle, there is a part where all principal string instrument players are performing. If listened in an audience seat, almost everyone should sound from center, however they are spread to 180 degrees from L to R, viewed from a Decca Tree position. Similar deformation is observed for this mix. Also in this movement the celesta being in heavy usage, it is worth paying attention to whether its sound image is separated from a harp or not.

In the last part of the 3rd movement, 'nightingale twittering' is assigned to play a tape. (on the score, Gramophone record No.6105 is designated.) This part is replaced by our original source specifically prepared for surround purpose. Normally this part is played less louder, but we tried a TV oriented mise-en-scene, so it is mixed louder. Please note that this part was mixed in the post production process and not played in the hall.

As we did not add any artificial echo to 'twittering', it should sound completely separated from other play giving such impression that you are suddenly straying off into the woods where you hear music drifting.

A nightingale part is based on a 4ch surround field recording in the Blackwood (Schwaltzwalt) in Germany. However as we could not find one, we mixed with stereo recordings of a nightingale tittering offered by an ORF sound engineer. It could be that they do not twit simultaneously. Your understanding shall be appreciated.

Listening checkpoint:

Apart from the validity of this mise-en-scene, you are welcome to listen to them and see if you can feel the atmosphere and the depth of the woods with its natural dimension.

At our mix-down site, we made the sound image evenly spread from L to R as well as F to R. Please check if it has not deviated very much from the homogeneous soundscape. At the end of this part, there's also a scene in which a bird is flying: a flutter of bird wings moves from the Front Center to Rear via slightly Left over your head.

Please note that this part for Mr. Fukada's mix is also my (Irimajiri) post-production due to unpredicted circumstances.

In the 4th movement, groups of fanfare, called 'banda'(Italian word meaning band) are coming in. The banda consist of three groups whose allocation in the rear seats (refer to a layout figure on P.32) is, from the left, 2nd trumpets trombones and 1st trumpets. From a physical positioning point of view, they should localize in between the rear center to about +/-45 degrees' sectoral range. However, since the ITU recommended rear loudspeakers' allocation angle is very large (=110 degrees), the localization of banda is pulled toward them, hence they sound separately from left and right. Furthermore, the trombone in the rear center tends to stay either left or right depending on your listening position. It is normal because if you turn around and listen to it, it is spreading all around the rear loudspeakers.

This is the limit of 5.1ch surround and it is an issue how we can 'cheat' our ears as a mixing engineer. A 6.1ch or 7.1ch scheme that needs more loudspeaker(s) may improve this situation quite a bit. Although we are usually listening to monitor loudspeakers compliant with the ITU recommendations while we are making a mix-down, you may hear them more naturally if you can set the rear loudspeakers at +/-45 degrees.

Albeit the limit of the 5.1channel, it is fun to listen to 'Front dialoging with Rear', rich in surround effect.   The latter half of this movement may be used to assess whether it can express dynamism including a climax. If a frequency response of a rear loudspeaker system differs a lot from that of a front one, you may feel that something is missing out of winds' dialog at close to the end.

This mix used a limiter but very modestly, so the dynamic range is very big. Normally (for TV production, especially), we sufficiently compress gains, but this mix set a compression gain to a necessary minimum. As a result, if you are listening at night until you can hear solos in the 3rd movement with sufficient loudness, the very last climax would cause neighbors a big trouble. In fact, one of our staff received a complaint from his neighbors. Your consideration shall be appreciated.

5）Track⑤ Beethoven : Wellington's Victory 16:40 Fukada Tree

Akira Fukada

Mixing Note

Basic mixing arrangement is the same as that described earlier (refer to 2. 'Pines of Rome').

So is the main microphone as well. The feature of this music piece is a dialog between F and R. The issue in such a case is a control of acoustics. It is hard to imagine from a hall design point of view that 'front' instruments resonate beautifully to rear or 'rear' instruments to front vice versa. A concert hall is designed so as to diffuse sounds from stage to audience seats. Written notes are therefore not necessarily reproduced fully in a hall. Unless musically meaningful image can be restored as a recorded work, we'd find least of interest.

In short the point to realize what is intended in this music piece is to guarantee good separation (in localization) between Front and Rear. Of course we have relatively good separations out of the main microphones. If our recording session ended with this piece alone, we could have arranged their directivities and not.

On the mix, for the banda we used time aligned spot microphones considering the propagation delay of the main microphones. To avoid 'turbidity' due to sound overlapping and to maintain separation, other spot microphones were not used except for some cases after the dialog part.

6）Track⑥ Beethoven : Wellington's Victory 16:40 INA5

Hideo Irimajiri

-1) Mixing Note

For the main microphone I chose INA5.

Because INA5 has a fairly natural response relating to localization around 360 degrees, it is very much of use in a case where there are instruments in the rear of a hall like the mise-en-scene in this piece. However directivity of these microphones being cardioid, low frequency response is decaying gradually, we can not record such lows as you'd experience from a gran cassa that may eventually vibrate your belly with an omni-directional microphone. The said difference is obvious by comparing the recording of Mr.Fukada.

Spot microphones are time aligned for a rear brass band, snare drums, another brass band on the 2nd floor. Delay for snare drum in a case surpassed 50 ms. In the case where instruments are located in the rear of a hall, delay arrangement might become difficult by a surround microphone system that uses another ambiance microphone array. This is the background reason why we took INA5.

For strings, close-up microphones are set (for every group) as spot microphones, while three sub-microphones are placed in front of wind instruments. The allocation of the sub-microphones is similar to that of 3Omni (L-R approx. 1m apart each). Its height is about 2.5m. But if L&R microphones are completely assigned to L&R, the sound image of wind instruments gets too wide, we panned a little to narrow it toward the center. Spot microphones were used to enhance details of the percussions like a timpani.

Though it gets difficult to pickup a specific wind instrument by such an approach, we can have a natural width from woodwind instruments.

-2) Hearing Note

In this music piece, observe that you are enveloped by dialoging between the instruments virtually allocated at the Front and Rear loudspeakers.

As the first drum march is really marching in from outside the hall, there's a big change of tonality and acoustics at the instant of their entering through a door. When they are outside the hall, the sound is only permitted to enter through the door's opening: so we hear faint direct sound with little reverb. 'Being far' does not mean 'deep reverb'. At the very instant when they enter through the door, drums fill a whole hall with their rich sound. If you could imagine the march's passage by just listening to the recording, then it would fulfill our objective. But if your system is not well adjusted, you may hear that sound get simply louder by crescendo. The British troop will show this move in Rear and the French troop will do likewise deep in Front.

Followed by the march, trumpets' sounding march and a brass band's march will follow. Strings on the stage are joining in the middle of the march. For British troop side, you must be caught between a brass band in the rear and strings in the front. Observe if you can get the distance impression of F-R. For French troop side, a brass band stays a lot far from the orchestra in the back right. Observe if that brass band is heard farther back from the woodwinds and if you could in both cases, then we did the job all right as far as our intention is concerned.

To tell you the truth, French troop's brass band staying in the rear is very difficult to record its 'distance'. In fact they are more than two times farther from the normal position of woodwinds. Therefore, it can be that you could hear it just from your back right, even though your playback system were well tuned.

In the battle scene, the checkpoint will be whether you can feel enough punch out of exchange of cannon fires between rear right and front right. Physically speaking, the rear 'fire' should localize 30 degrees more on the right. But as mentioned earlier in (3)'Pines of Rome', that localization will undergo shifts toward the loudspeakers' position. As a result, they hear at about 60 degrees in the rear right. Unless your system is well tuned, panned components in between the two rear channels may be heard separated for each loudspeaker. Moreover, with a system that is insensitive to change in 'distance' of the sound image between front and rear, the image may overlap each other too hard so that the battle scene may sound muddy.

Because INA5 consists of cardioid microphones, tonality of the cannon fires will loose richness as compared to omni-directional microphones and that even though it is assisted by an EQ'd spot microphone, it may sound like "pong, pong" to our ears.

As to localization, it is diffusing more naturally to all directions than the case of (3)'Pines of Rome'.

This is rather natural since INA5 is originally designed to get better localization. Having said that, very subtle nuance in localization does not necessarily degrade musical quality when we enjoy listening to it.

Though this is my personal view, but rather the impression on tonality or breadth must have to do more with the contentment in appreciation of music. Therefore you do not need to be too nervous for the localization and precisely position your loudspeaker systems when you adjust your playback system for the music appreciation purpose. Especially since ITU's recommended positions for surround rear loudspeaker systems are often the ones that'd bother your everyday life, a little bit of a position change , I think, is to be acceptable. Inversely, producers have to avoid such mise-en-scene that'd require solid ITU specified positions.

By the way, I would like listeners to think of the mise-en-scene by which musical instruments are allocated in Front and Rear. Some listeners may not like them heard from their Front and Rear as they can not concentrate, some may like that spectacle.

This is again my personal view, but music can be more free. A trial approach to 'surround sound' has been existing since early times: In the times of Renaissance, at Venice, some sound generating devices were placed here and there in an auditorium not mention plural orchestras.

I do not know when it started but it is customarily an admitted style of concert such that audience is sitting properly and listen in a hall. I hope surround sound can be enjoyed together by those who are engaging in music, not only engineers but musicians and the audience.

The 2nd movement represents a standard orchestral constitution in the classical music. Sub-main microphones were used for wind instruments but no spot microphones which however are often used in classical music recordings. If an orchestra balance is impeccable, there are cases where we do not use main-sub microphones either. However spot microphones are often in use to add details for percussion instruments.

From the Fugue (string ensemble) and later, this part is suitable for us to listen to localization. The melody is played on a different instrument one by one, you may check if you can hear them separately.

As I mentioned earlier that the INA5 is a complete one point surround, and that it can give you good sound localization, it is interesting to observe how the sound from the orchestra is reflected by rear walls of the hall. I can not generalize which one is best, however from what I can feel like listening in a real hall on site, the INA5 can better realize that impression.

7）Track⑦ Mozart: Overture to 'The Marriage of Figaro' (4:25) 30+Asahi

Hideaki Nishida

-1) Mixing Note

When I mix-down classical music, I usually check scores of the recorded music piece before anything else. I am making every effort to understand what the composer intended by reading dynamic marks, harmonies of the ensemble etc. Then, I participate in the orchestra rehearsal and eventually narrow the gap between the conductor's intention and my score reading. Then, I check and fix position of the main microphones in that hall, their heads' directions, instruments that need spot microphones and their positions prior to the actual recording. If I can finish the work with the richest musicality as a result of these efforts, I shall be most happy as a mixing engineer.

Because Asahi Broadcasting Corporation is the owner of the Symphony Hall, my classical music recordings have been taken place at this hall in most of the cases. The position of the present microphone hanging mechanism or microphone arrangements etc. have been established in collaboration with Mr. Maeda, chief person in charge of hall acoustics, through trial-and-errors based on our predecessors' know-how.

Main microphones used in Asahi Broadcasting Corporation are three omni-directional microphones fixed on a stereo bar (210~300mm) and three other omni-directional ones are positioned at each corner of a T-letter (L-R distance=4m, C to center of L-R=2m, height=2.6m) just like a Decca Tree.

We mix them to represent basic size and balance of the orchestra. As to the ambiance microphones, one point hanging microphones are distanced with an interval of 3 m over the audience seats, each one has two omni-directional microphones that are directed ahead and rear. So total of four omni-directional microphones consist of an ambiance array named as 'Asahi'. They all use DPA4006 including the main microphones with an option of a Grid cap attached for an ambiance rear microphone depending on the need. Furthermore, for this recording a 'Diffuse-field Grid' was attached to a rear microphone to increase highs on its axis.

' Asahi' ambiance microphone array differs from others in spatial placement in a hall. In other words, other arrays are approximately 5 m high, but 'Asahi' array is set much higher than this at about 7.4 m high. This is because we want to avoid direct sound and get hold of a hall tone as much as possible.

From our experience with 'Asahi', we customarily move ambiance microphones up and down by several tens of centimeters to find the best position during rehearsal.

A mixing ratio of Asahi's Front to Rear is 1:1 used for psycho-acoustic experiments; actually I maneuvered this ratio. In this mix-down, rear ratio is slightly greater than front so that we get more reverberation (hall ambiance) without loosing clearness of the orchestra sound in the front.

At our surround experiment, a wide variety of main microphones are placed in a hall at a time. Since our usual microphone setting position and interval are different from those of Decca tree or 3Omni, I set such a criteria in selecting microphone settings that main microphones shall be the ones by which I do not feel uncomfortable vis-à-vis those I'm used to use for mixing in the Symphony Hall and that they mix well with 'Asahi'. As a result our option was '3O'+Asahi' to be used with occasional spot microphones.

As a nature of this kind (experiment) and also that I could not arrange nor adjust the setting of all the microphones I employed by myself, I felt a bit confused at the onset of my mixing.

(Just for your information, in a composition of such a size I usually use spot microphones hanging from the ceiling and the distance between the instruments and the microphones is about 1.3~1.6m: one for two woodwinds, one for horn, one for timpani and one standing on the floor for contrabass.)

It is also true that it was a precious experiment for me that I could never have been able to experience except such a circumstance that trying to narrow a sound image closer to mine with a unusual microphone arrangement.

In fact, as soon as I started mix-down manipulating a fader, I found myself concentrating on concretizing a sound that I felt comfortable.

Furthermore, spot microphones were not time aligned because I did not think it necessary. As to LFE channel, I did not use it either because I could find no lucrative sound sources and that I wanted to keep consistency with the down-mix.

-2) Hearing Note

The composition of this piece being relatively compact (eight 1st_violins, six 2nd_violins, four violoncellos, four violas, two double basses, two flutes, two oboes, two clarinets, two bassoons , two French horns, two trumpets, and timpani), the width of orchestra was recorded surely in front L, R and C. Actually, strings would spread from L to R, and woodwinds would localize a little behind this line and in the center, horns on its left, timpani in center. This mix should sound like this: the tune is peaceful and soft at the beginning but from the last part to the ending as if each instrument plays one step ahead.

8）Track⑧ Mozart: Overture to 'The Marriage of Figaro' (4:25) OM8

Tohru Kamekawa

Mixing Note

Because this piece has a standard composition as an orchestra, I used principally main microphones and very rarely spot microphones. Two omni-directional microphones (L&R) and a bi-directional one (Center) are used for the main microphone system with its acronym OM8, and two front microphones of a OSQ-M for the surround microphones: the surround sound field was created by these five microphones. Spot microphones are placed before a contrabass and timpani to sharpen their edges slightly. No other processing such as reverb was necessary, because I could get sufficient acoustics out of it. I tried hard to express the mix by using the minimum number of microphones so as not to damage clarity and avoid being too muddy.

9）Track⑨ Mozart: Overture to ‘The Marriage of Figaro’(4:25) Fukada-Tree

Akira Fukada

Mixing Note

The basic philosophy is just the same as the one described earlier. (refer to (2) ‘Pines of Rome’)

I would have moved the microphone position from the present one: little higher and about 10 cm toward the audience seats. Because I felt the strings in the ensemble a bit louder than the woodwinds being inputted in the main microphones. Accordingly, level of a spot microphone for strings was enhanced just a little to balance the level of the woodwinds. As a result, spot microphones for woodwinds were very seldom used. Spot microphones were used to enhance clarity of the attack of timpani, or to give a distinctive edge of a horn and so on.

10） Track⑩ Mozart: Overture to ‘The Marriage of Figaro’(4:25) 3O+OSQ-F

Hideo Irimajiri

-1) Mixing Note

For this mix, 3O(mni) was opted for main microphones in which, like a Decca Tree, three omni-directional microphones are used. It is placed about 70cm toward the rear from the Decca Tree. The center microphone itself is distanced 1 m farther to the rear. As the 3O is not my usual array, I am not fully aware of its benefits yet but the localization that this array offers is quite different from the Decca tree. It may be because the center microphone is not poked out ahead like a Decca tree. For ambiance microphones, I employed a distant omni-square named OSQ-F.

In the case of the ’Pines of Rome’ where banda (brass band) was joining to play, OSQ-F was not used because it was too close to banda. In other case, if the Omni-Square is distanced longish, i.e. direct sound loses its energy. I think the Omni-Square mixes well with main microphones. In this case again, no time alignment was carried out between ambiance and main microphones. As long as spot microphones are concerned, they were time aligned by the same method described earlier in (3) the 'Pines of Rome' as well as adjusting the balance.

-2) Hearing Note

Since the composition became compact, one may feel L-R breadth much reduced compared with the 'Pines of Rome', wind instruments localize just as the 'Pines of Rome' or one may make surer judgment on it, since the number of instruments is smaller. Orchestra itself has a narrower width due to a compact composition. It is less likely that localization was pulled in either one of the loudspeakers because of Hass effect. Because five kinds of string instrument are sufficiently close to microphones (not that close as a Decca Tree), one can hear with enough width from L to R. Since a tonality balance is good, I would presume that this music piece might be useful to assess tone quality of your playback system such as comparative listening of loudspeakers.

11） Track⑪ Bach: 'Toccata and Fugue' (9:10) DT+Asahi

Hideaki Nishida

-1) Mixing Note

In Asahi Broadcasting Corporation, we use usually a combination of a 3O(mni)[though the microphone-to-microphone distance is smaller than the 3O original] and a Decca Tree (DT) for the main microphone, as mentioned earlier in the 'Marriage of Figaro'. I monitored with the same viewpoint in search of a main microphone array and selected a Decca Tree among others set together, because it was close to the sound source and hence I could feel much of a power. Furthermore, an LFE channel was applied for the organ using a spot microphone because I thought this channel is valid. By mixing more 'Asahi', I could enhance massiveness of the organ sound in addition to a rich hall tone. I did not use reverb because 'Asahi' array had sufficient reverberation.

-2) Hearing Note

We get very good front localization relative to the organ in the Symphony Hall, as if the organ were resonating all over the hall. I hope I could convey the said impression.

12)　Track⑫ Bach:‘Toccata and Fugue’(9:10)　5C+HSQ-N

Tohru Kamekawa

Mixing Note

In this down-mix, I have looked to express scale of the organ using a combination of five cardioid microphones placed at an interval of 2 m (5C) and Hamasaki Square Near (HSQ-N). The position of the front microphones being relatively far from the organ, its distinct localization is not available out of these five cardioid microphones. However since no omni-directional microphones were used, clear sound has been obtained as a whole. To enhance low-end powerfulness, a spot microphone is set in the vicinity of the low frequency pipes for an LFE channel.の

13)　Track⑬ Schubert: Heidenröslein (1:50)　DT+Asahi

Hideaki Nishida

-1) Mixing Note

A Decca Tree (DT) was selected for the main microphone after carefully monitoring the contour of a tenor voice and a distance between piano and the tenor. Spot microphones were used to solidify their localization: they were lightly mixed with the main microphone. Reverb was not used because I had sufficient reverberation from the ‘Asahi’ ambiance microphones. The usage of LFE channel was abandoned because I found no sound source effective for it for one thing and that I took care of the compatibility with the down-mix for the other.

-2) Hearing Note

A simple constitution of a tenor and a piano; as a musical structure, the core is a presence of the tenor and an accompaniment piano that does not bother him.　A scene is represented by sound: A tenor is standing in the middle of the stage and his voice is enveloped by a rich echo in the Symphony Hall.

14） Track⑭ Schubert: Heidenröslein (1:50) OM8

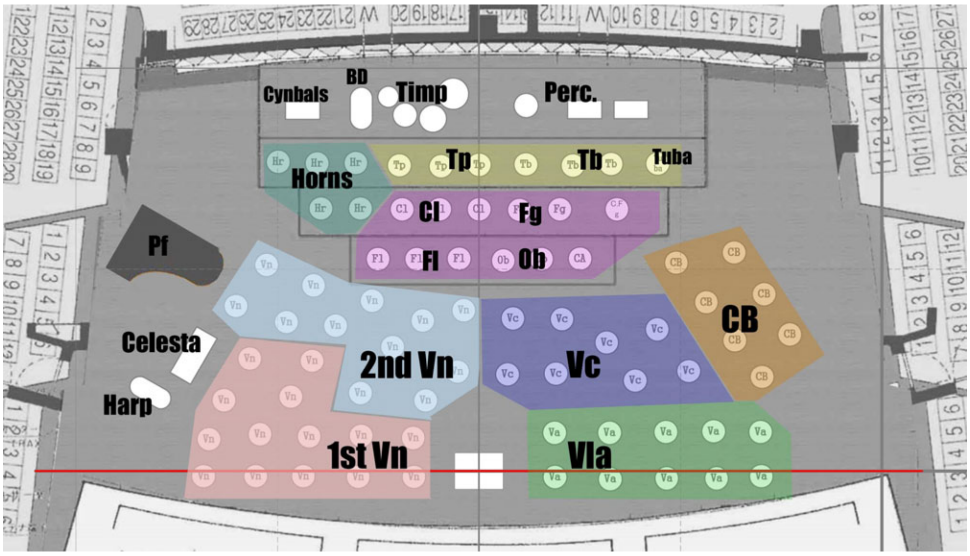
Tohru Kamekawa

Mixing Notes

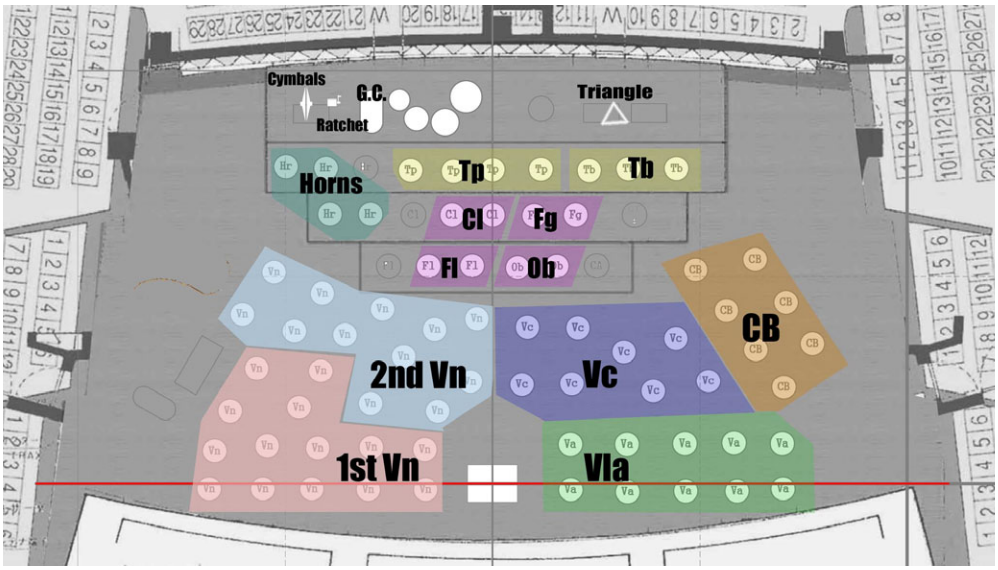
In order to represent best a simple constitution of a tenor and a piano, I tried to create a natural soundscape and opted for five microphones in total: two omni-directional microphones to L and R together with a bi-directional (figure of eight) microphone in the center for the main microphones, front two microphones out of the OSQ-M array for the surround microphones. For mix-down I added slightly spot microphones to the vocal and piano and strove to make a soft sound that makes the most of the beautiful acoustics of the Symphony Hall.

§4 Layout of the Microphones

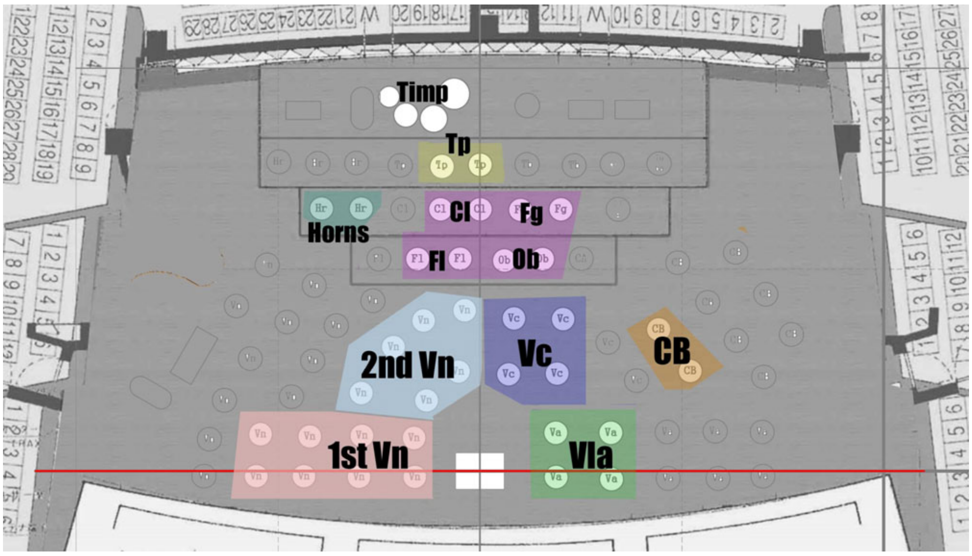
1) layout of the microphones (plane view) 'Pine of Rome'



2) layout of the microphones (plane view) 'Welloington's Victory'

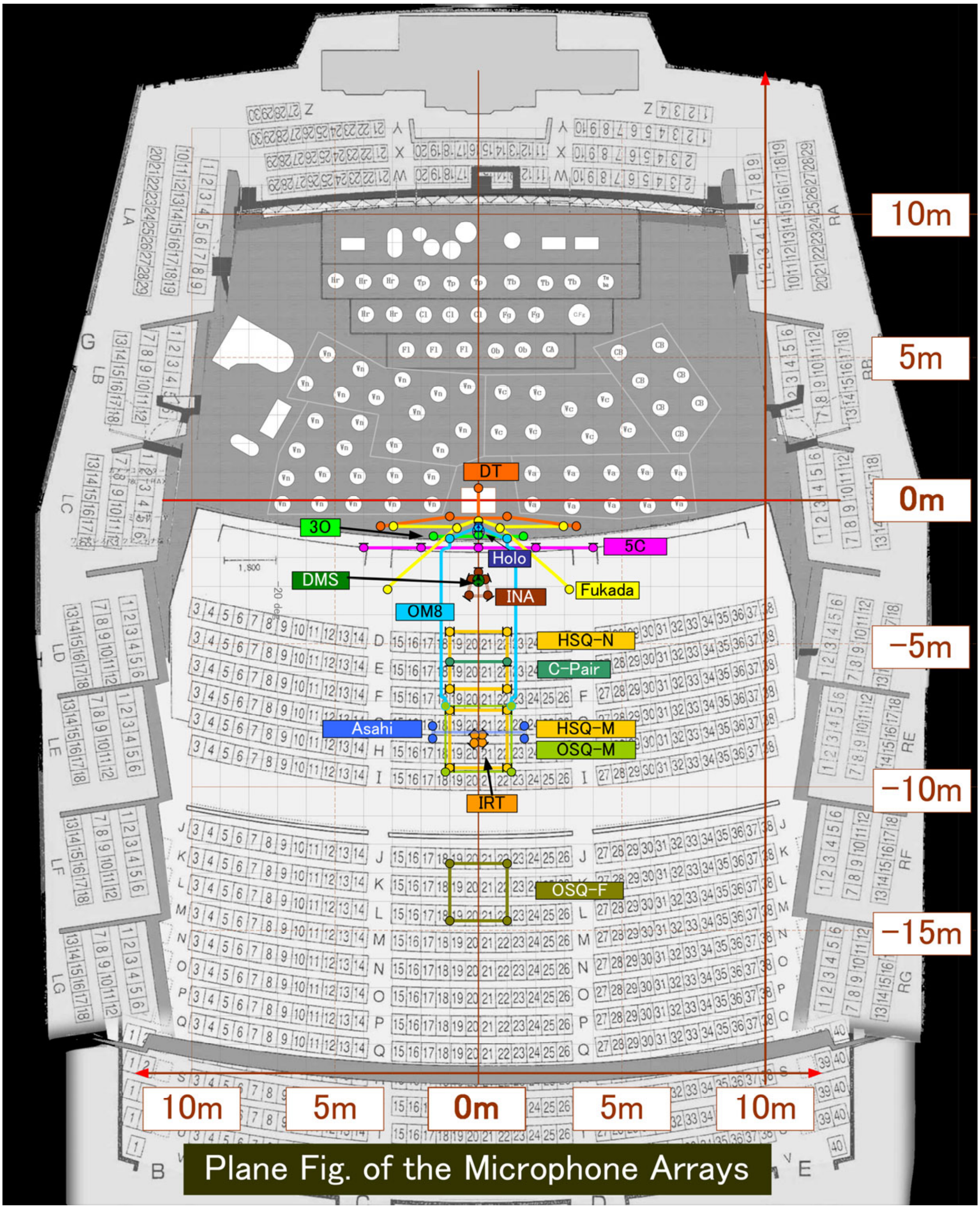


3) layout of the microphones (plane view) Overture to 'The Marriage of Figaro'

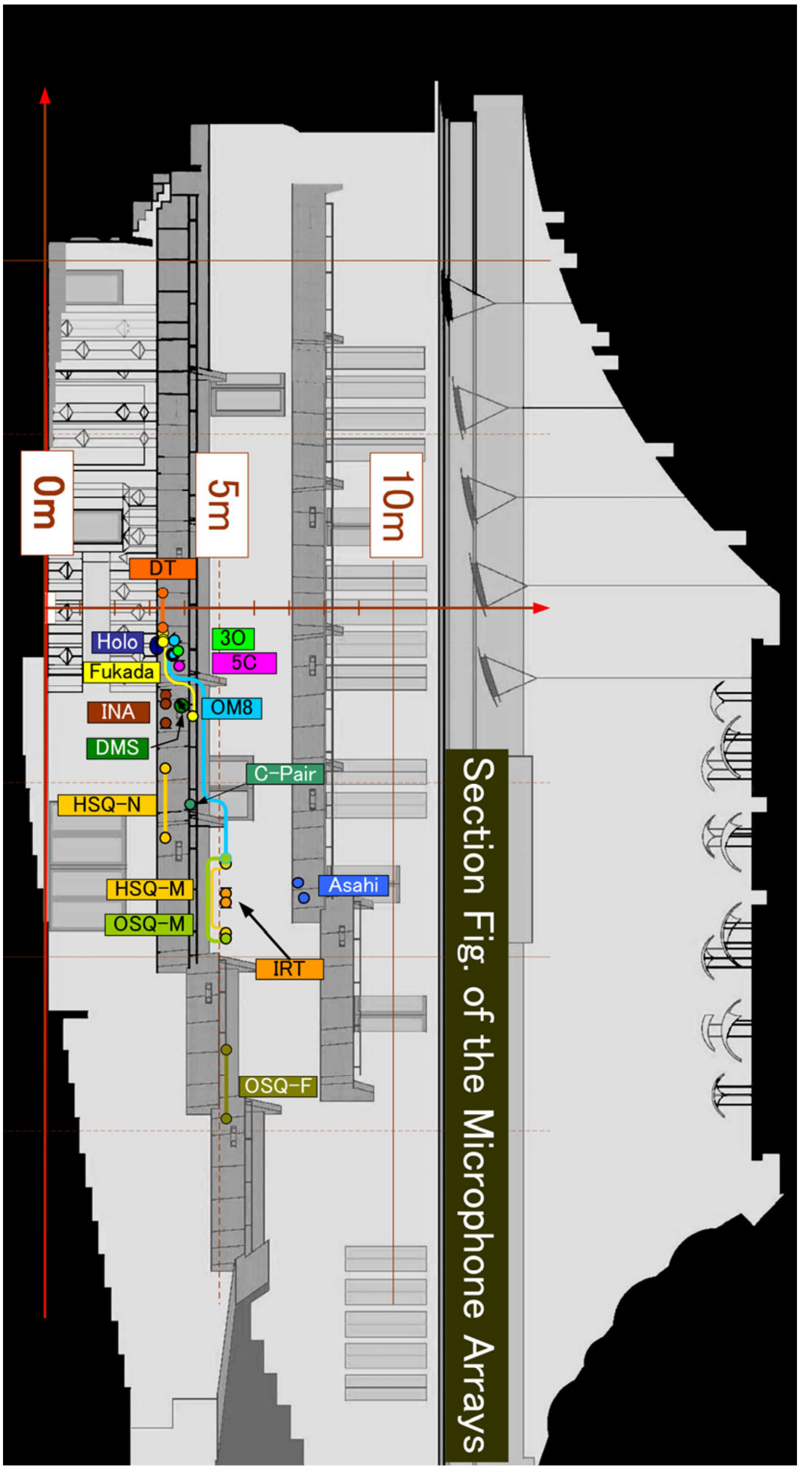


§5 Layout of the Microphones

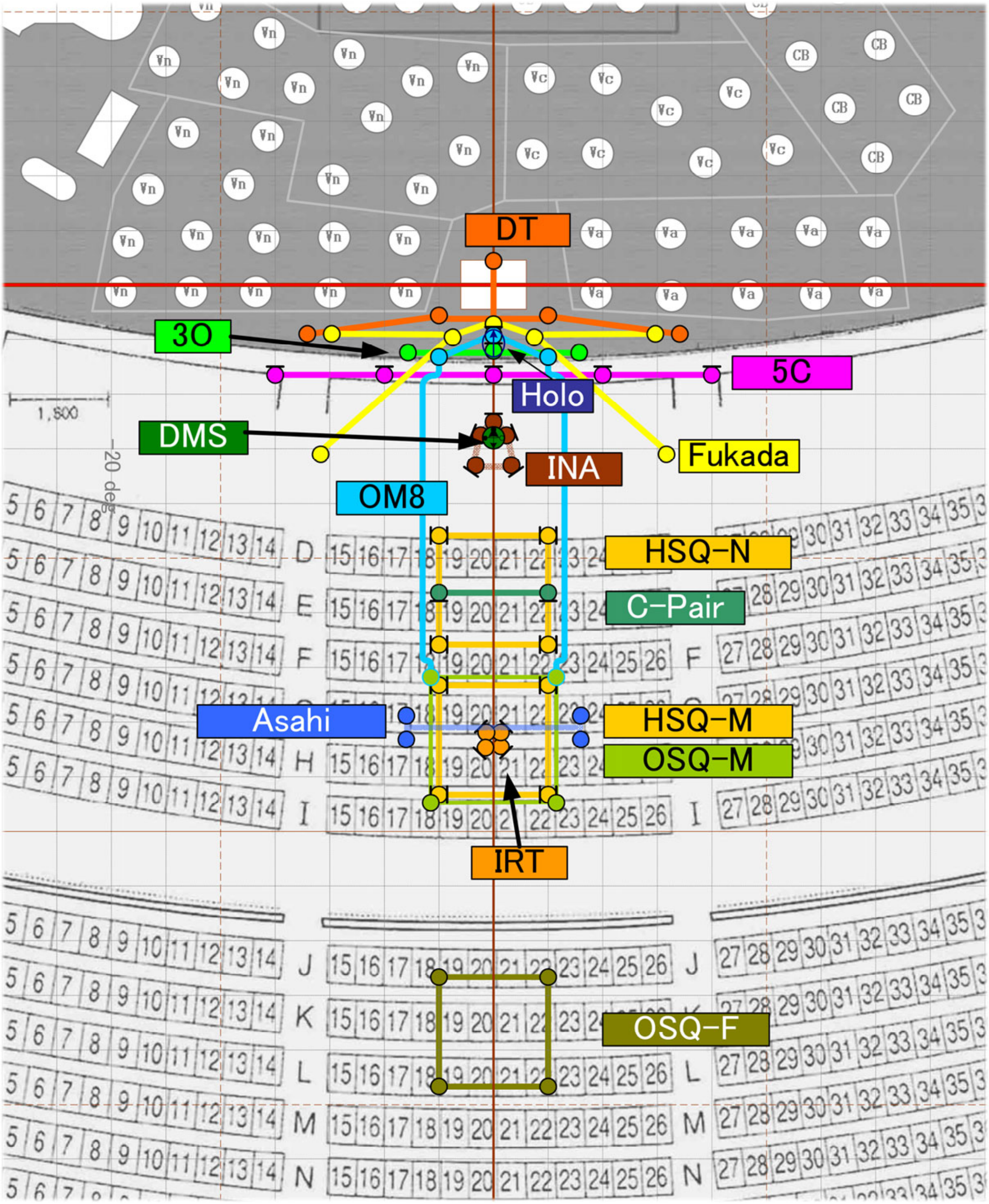
※layout of the microphones (plane view)



※layout of the microphones (cross-sectional view) :stage floor as a height reference



※layout of the microphones (plane view): enlarged view around the center



§6 Test signals

Masatoshi Maruya

1) Category of Test signal

This CD-ROM contains four categories of test signals, namely surround test signals compliant with the ARIB test signal patterns, four types of stereo pink noise signal. CD-ROM is composed with the followings:

①-18dBFS_TEST_FILE_FOLDER

This folder contains surround test signals specified with –18dBFS, 48kHz sampling rate and 24 quantization bits in a 'split mono wave file' format. By sorting by name, we get the following list.

ARIB_TEST_SIGNAL_-18dBFS_48K24bit.C.wav	C Channel wave file
ARIB_TEST_SIGNAL_-18dBFS_48K24bit.L.wav	L Channel wave file
ARIB_TEST_SIGNAL_-18dBFS_48K24bit.Lf.wav	LFE Channel wave file
ARIB_TEST_SIGNAL_-18dBFS_48K24bit.Ls.wav	Ls Channel wave file
ARIB_TEST_SIGNAL_-18dBFS_48K24bit.R.wav	R Channel wave file
ARIB_TEST_SIGNAL_-18dBFS_48K24bit.Rs.wav	Rs Channel wave file

②-20dBFS_TEST_FILE_FOLDER

This folder contains surround test signals specified with –20dBFS, 48kHz sampling rate and 24 quantization bits in a 'split mono wave file' format. By sorting by name, we get the following list.

ARIB_TEST_SIGNAL_-20dBFS_48K24bit.C.wav	C Channel wave file
ARIB_TEST_SIGNAL_-20dBFS_48K24bit.L.wav	L Channel wave file
ARIB_TEST_SIGNAL_-20dBFS_48K24bit.Lf.wav	LFE Channel wave file
ARIB_TEST_SIGNAL_-20dBFS_48K24bit.Ls.wav	Ls Channel wave file
ARIB_TEST_SIGNAL_-20dBFS_48K24bit.R.wav	R Channel wave file
ARIB_TEST_SIGNAL_-20dBFS_48K24bit.Rs.wav	Rs Channel wave file

③PINKNOISE_FILE_FOLDER

This folder contains four types of pink noise signal. Each noise has a different RMS value specified with a 48kHz sample rate and 24 quantization bits in an ‘interleave stereo wave file’ format.

pink24bit48k-18dBFSrms.WAV	pink noise with -18dBFSrms
pink24bit48k-20dBFSrms.WAV	pink noise with -20dBFSrms
pink24bit48k-21dBFSrms.WAV	pink noise with -21dBFSrms
pink24bit48k-23dBFSrms.WAV	pink noise with -23dBFSrms

2) Surround Test Signals

Surround test signals recorded in the disc are mono signals having a ‘WAVE’ file format and six channels (L, R. C. LFE, LS, RS) form one set. The reference levels are either –20dBFS or –18dBFS.

The signal patterns are compliant with the standard surround test sources detailed in the ARIB’s ‘Technical guideline for 5.1ch surround sound program production’

(Fig.6.1) Test signals consist of ‘level adjusting signal for recording’ and ‘monitor loudspeaker adjusting signal’. Their patterns are shown in the following figure.

For a line test etc., you may transplant this file on your digital audio workstation (DAW) and output as they are or lay back on a VTR for your use.

Time base 1:00:00 1:00:30 1:01:00 1:01:30 1:02:00 1:02:30 1:03:00 1:03:30 1:04:00 1:04:30 1:05:00

Channel																			
1:L	1k						1k	1k	1k	50	P						P	P	P
2:R		1k					1k	1k	1k	50		P					P	P	P
3:C			1k					1k	1k	50			P					P	P
4:LFE				50				50	50	50				P				P	P
5:LS					1k				1k	50					P				P
6:RS						1k			1k	50						P			P

1k =OSC (1kHz)

50 =OSC (50Hz)

P =Pink Noise (Full Range)

ARIB TEST SIGNAL PATTERN

Fig6.1 Test signal patterns compliant with the ‘Technical guideline for 5.1ch surround sound program production’ (ARIB)

(Note 1) Sine waves at nominal 50Hz and 1kHz are in fact the ones at center frequency of 49.8Hz and 1002.0Hz respectively that are optimized to FFT measurement. If these signals are measured by FFT, the minimum number of the sample data are the following.In the case of 49.8Hz: 16384 samples

In the case of 1002.0Hz : 8192 samples

(Note2) Insertion of a low pass filter (fc=80Hz, 18dB/oct) in the recording/playback path of an LFE channel may result in a very little loss at 50Hz. It theoretically is 0.2dB for a typical butterworth filter.

(Note3) Since this pink noise signal has either –20dBFSrms nor –18dBFSrms, its playback level shall be 3dB louder than the sine wave RMS value of a reference level.

Therefore, when you make adjustment of a playback level using this signal, you are requested to add +3dB for a playback setting level, then this level shall become the correct value, for example 82dBC in the case of 79dBC setting.

For precise adjustment, you are requested to opt for one of the stereophonic pink noise signals.

3)Pink noise recorded on this disc

Other than the surround test signals, the current disc contains four kinds of stereophonic pink noise whose RMS value is different each other. Their signal level is specified by the average RMS value measured during 60 s.

The signal levels are set to -20dBFSrms and -18dBFSrms according to the literature. Other than these two, this disc also includes pink noise signals of –23dBFSrms and –21dBFSrms. Either one of these values corresponds to the RMS value of a sine wave at the corresponding reference level. It is one of these pink noises that are more useful at a real operation.

It is difficult to correctly measure the RMS value of a pink noise on a VU meter or using a measurement apparatus by its nature. In general, an average value obtained by full wave rectification is often converted to show its RMS value in an analog meter or in a measurement device. In such cases, only a sine wave shows a true value, but measurement errors are inevitable for noise signals, a rectangular wave and a triangular wave. For instance, a pink noise that shows in the vicinity of 0VU, a true RMS value shall be in between +1VU and +2VU. (Note that as to a VU meter, the errors are dependent on manufacturers or an individual meter.)

If a conventional average-value type voltmeter is used, it would show approx. 1dB below the true RMS value. Therefore, you should use a reliable pink noise that clearly indicates showing its RMS value for your adjustment.

[Signal level of the included stereophonic pink noise]

・–20dBFSrms：RMS value is greater by 3dB than a sine wave of –20dBFS.

・–23dBFSrms：The same RMS value (reference value) as a sine wave of –20dBFS.

・–18dBFSrms：RMS value is greater by 3dB than a sine wave of –18dBFS.

・–21dBFSrms：The same RMS value (reference value) as a sine wave of –18dBFS.

[Relationship between analog and digital signals in professional apparatus]

For a professional apparatus that uses –20dBFS as a reference level, let us look at the relationship between analog and digital signals of a sine wave as an example.

① In the case of analog to digital conversion (at recording)

0VU sine wave = +4dBu sine wave 1.228Vrms 1.736Vpp

⇒ A/D conversion（–20dBFS／peak value 0x0CCD = reference level）

② In the case of digital to analog conversion (at playback)

・Peak value 0x0CCD sine wave = –20dBFS sine wave = –23dBFSrms

⇒ D/A conversion（0VU 1.228Vrms 1.736Vpp = reference level）

If a pink noise of –20dBFSrms is played back on such an apparatus, the output RMS value turns out to be 3dB greater than the reference level. In the same manner, a pink noise of -23dBFSrms is played back, the output RMS value turns out to be the reference level.

[An example of level adjustment using a pink noise]

For an apparatus that uses –20dBFS (or –18dBFS) as a reference level, if a pink noise signal of –20dBFSrms (or –18dBFSrms) is used, level adjustment is necessary. Since we know an RMS value of a digital signal relative to any one of the included pink noise signals, we can make adjustment with the following procedures:

① First, set your apparatus at the reference level, then playback a 1kHz signal at –20dBFS. Make sure that the monitor output level is 0VU (or peak level meter ' s reading at –20dBFS). If it is not the case, adjust the level.

② Next, adjust the level so that the monitor output level becomes –3VU (or –23dBFS). (-3dB gain control)

③ If a pink noise signal of –20dBFSrms is played back under this condition, you will get a pink noise of a reference level (0VU) at the monitor output.(Note that an actual meter needle does not point 0VU.)

Use this pink noise to adjust the playback level of your system (amplifier plus loudspeaker systems).

④ After playback level adjustment is completed, go back to ①'s state.

Moreover, you may get the same result without adjusting the level at first and make adjustment thereafter with the premise that a pink noise level is higher by 3dB. In this case, adjust your level at a ' playback level to be set +3dB ' . For instance, if you want to set a playback level at 79dBC, adjust the level at 82dBC (79dBC+3dB).

If a pink noise signal of –23dBFSrms (or –21dBFSrms) is used, no level adjustment is necessary.

4) Measured value of the stereophonic pink noise.

A frequency bandwidth of a pink noise is half the sampling frequency. With the cooperation of NHK research laboratory, we measured RMS values on digital data and fine tuned at a unit of 0.1dB. Measured values for 60s are the followings:

・Measured value of -18dBFSrms: -18.0075dBFSrms(Theoretically -18.01dBFSrms)

・ -20dBFSrms: -20.0071dBFSrms(-20.01dBFSrms)

・ -21dBFSrms: -21.0072dBFSrms(-21.01dBFSrms)

・ -23dBFSrms: -23.0079dBFSrms(-23.01dBFSrms)

The figure shows a frequency spectrum of a pink noise (stereo spec.) at –20dBFSrms where the number of FFT sample data is 8192 for 60 sec. Though there are some errors below 1000Hz due to the number of data, we obtain a spectral characteristic of approximately –3dB/oct. Pink noises with other levels show a similar spectral characteristic except their RMS values.

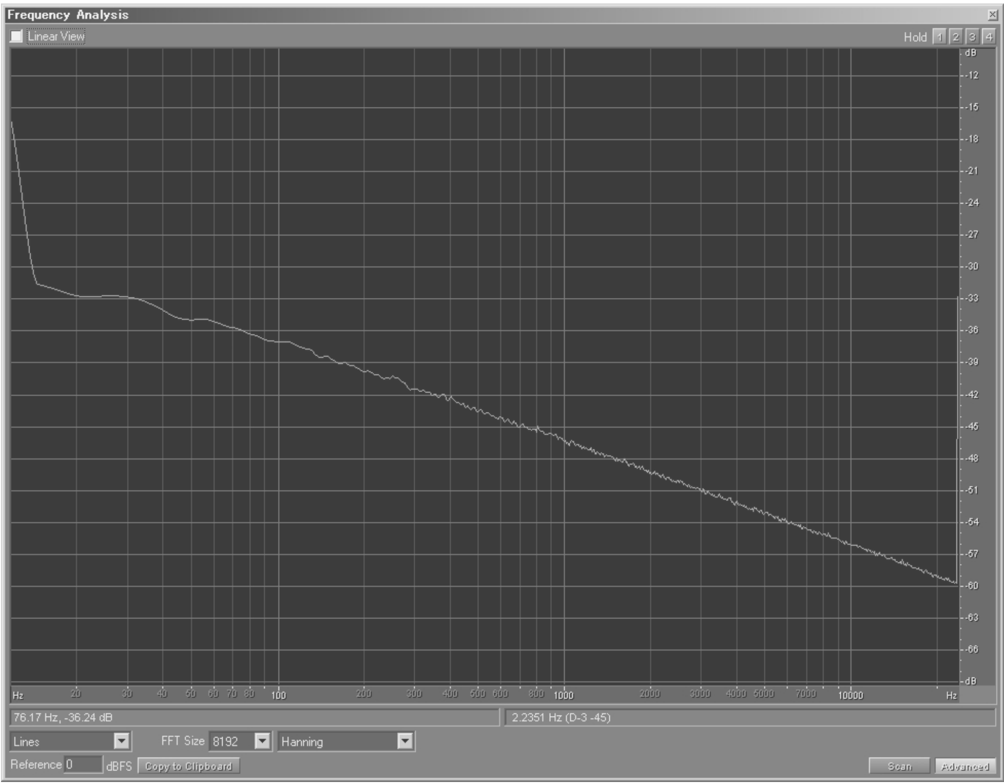


Fig.6.2 Spectral characteristics of the stereophonic pink noise included in the CD-ROM.

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Osaka Philharmonic Orchestra

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Shigeo Genda

Organ :

Seiko Katagiri

Tenor :

Hiroyuki Yoshida

Piano :

Toshiko Urabe

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