

Multichannel Monitoring Tutorial Booklet

Masataka Nakahara
SONA Corporation

June 2002
rev 2.3.0

©2002 YAMAHA Corporation



©2002 SONA Corporation

sona

Contents

Introduction

1. What is surround?

1-1. Stereo and surround

1-2. Channel configuration

1-3. Key points for multi-channel monitoring

2. Multi-channel formats

2-1. Surround processing methods

2-2. Encoding and compression methods

2-3. Recording response

2-4. Playback response

2-5. Down-mixing

3. Playback environment

3-1. L, R

3-2. LS, RS

3-3. SUB

3-4. Monitor alignment

3-5. THX[®]pm3[™]

4. Bass management

4-1. Acoustical treatment of the room

4-2. Speaker placement

4-3. Electro-acoustic methods

5. Monitor systems

5-1. Monitor matrix

5-2. Bass management

5-3. Monitor alignment

6. Measurement and adjustment

6-1. Test signal

6-2. Main channel level balance

6-3. LFE channel level balance

6-4. Narrow-band pink noise

7. Summary

8. Reference materials

Introduction

The most important consideration for a studio monitoring environment is that “the response of all channels be consistent.”

The second most important consideration is that this consistent response be “good response.”

We could list numerous parameters for deciding whether the response is “good,” ranging from subjective to physical, but the key point is that there be no large peaks or dips in the frequency response.

In the case of two-channel, it is fairly easy to create an environment in which “the response of all channels --- i.e., L and R --- is consistent.” We simply need to ensure that the shape of the room and the placement of the speakers is symmetrical between left and right.

In the case of multi-channel, on the other hand, it is often difficult to obtain a consistent playback response for all channels simply by creating a symmetrical speaker placement and room shape.

Mixing of the final product must be done in a properly configured playback environment.

No matter how high the grade of your equipment, it is impossible to create a final mix unless you have a good-sounding playback environment.

The essential identity of a professional studio is in its good monitoring environment.

The arrival of multi-channel is a good opportunity for us to reconsider the question of “what is a studio monitoring environment?”

1. What is surround?

1-1. Stereo and surround

“Multi-channel” is sometimes called “surround,” and “two-channel” is often called “stereo.”

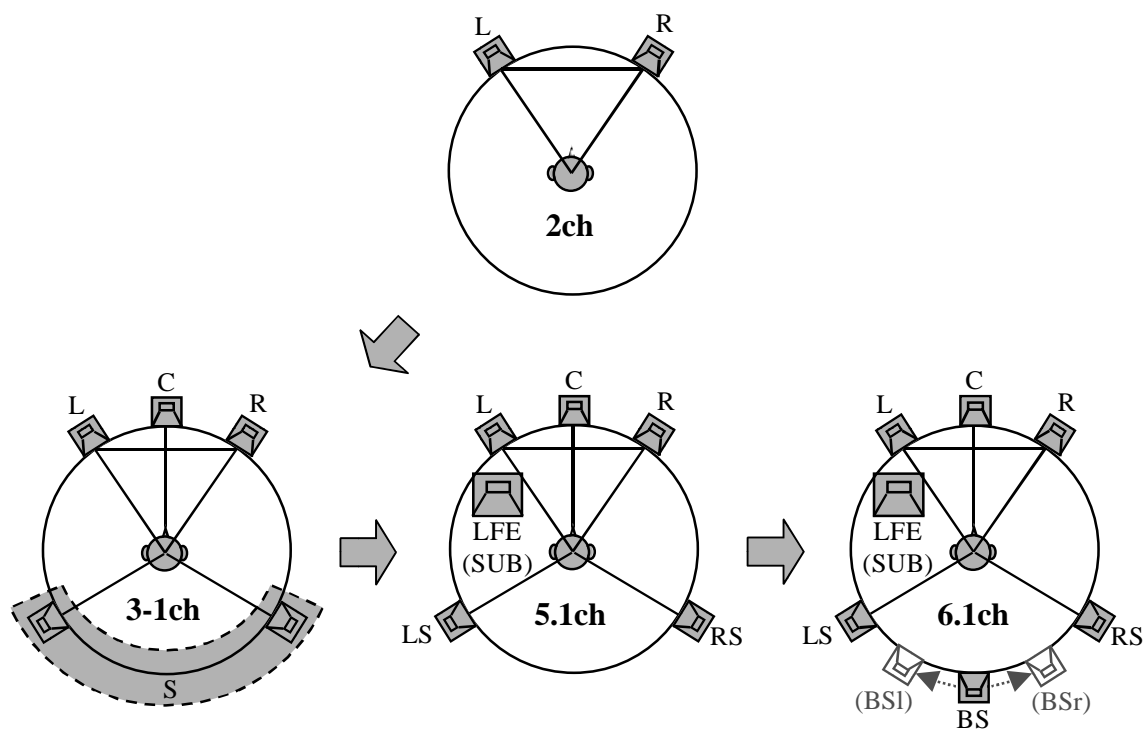
The precise terms are as follows.

Correct term	two-channel stereophonic
Abbreviation	two-channel
Common term	stereo
Correct term	multi-channel stereophonic
Abbreviation	multi-channel
Common term	surround

“Stereo (-phonic) = spatial acoustics”

1-2. Channel configuration

At present, a variety of channel assignments have been proposed for various types of media. The most popular of these are shown below.



[Fig. 1] 2ch, 3-1ch, 5.1ch, 6.1ch

1-2-1. 3-1 ch

This method is based on a two-channel system (L, R), and adds a center channel (C) and surround channel (S).

Although there are two surround speakers, one each at left and right, the playback is monaural.

The “3” in “3-1” indicates L, C, and R, and the “-1” indicates S.

Note that if “3-1” is expressed as “3.1,” this means “L, C, R” + “LFE” (discussed later).

1-2-2. 5.1 ch

This method is based on the 3-1 ch system, but changes the surround to stereo (LS, RS) and adds an LFE (Low Frequency Effect) channel for low-frequency effects.

The LFE channel is played back through a dedicated subwoofer designed for low-frequency playback.

1-2-3. 6.1 ch

This method is based on the 5.1 ch system, and adds a new back-surround channel (BS).

If two speakers are provided to play back the back-surround channel, these are sometimes called BS_L and BS_R, but the signal that is played back is a monaural signal where BS_L = BS_R.

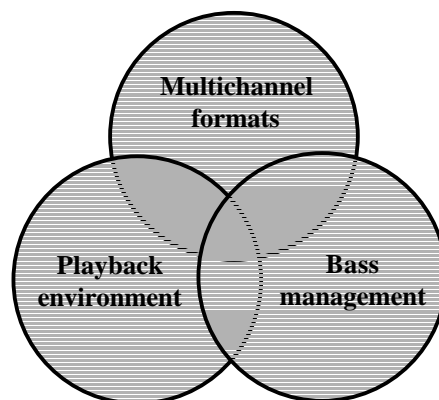
Current multi-channel systems were developed to maintain compatibility with previous systems, and have not been researched or developed in order to reproduce a 360° virtual acoustic space.

This means that if you expect current multi-channel systems to deliver full virtual acoustic playback capability, you will be at your wits end.

The key to multi-channel production is how to make effective use of the newly-obtained channels to create a product with the maximum “entertainment value.”

1-3. Key points for multi-channel monitoring

In our consideration of multi-channel monitoring, it is important to understand the following three key points.



[Fig. 2] Three keys of multichannel monitoring

In addition to the above three points, this document will discuss the construction of a monitor system, and the measurements and adjustments that are necessary in order to create a multi-channel playback environment.

2. Multi-channel formats

At present, multi-channel playback is supported by numerous types of consumer media, of which DVD is one. The playback method for each type of media is defined by the following organizations or manufacturers.

Media	Organization specifying the playback method	Storage method used	(Note)	Media standards organization
Film	Dolby, DTS, SDDS	Dolby, DTS, SDDS	<<	SMPTE, ISO
DVD-Video	Dolby, DTS	Dolby, DTS	<<	DVD Forum WG1
DVD-Audio	DVD Forum WG4	MLP (Packed PCM)	=	DVD Forum WG4
SACD	Sony, Phillips	DST	=	Sony, Phillips
Digital broadcast	Broadcaster organizations (Management side)	MPEG2-AAC*	<	ISO, IEC
	Dolby	Dolby**	–	–
	Broadcaster organizations (Management side)	MPEG2***	<	ISO, IEC
GAME	Dolby, DTS	Dolby, DTS	<<	Hardware manufacturers

(Notes) “<<” Each company provides a specific recording/playback type within the recording format specified by the standards organization.

“<” The recording format specified by the standards organization is used, and the applying organization considers the playback method.

“=” The standards organization directly specifies the recording/playback method.

* Japan, etc.

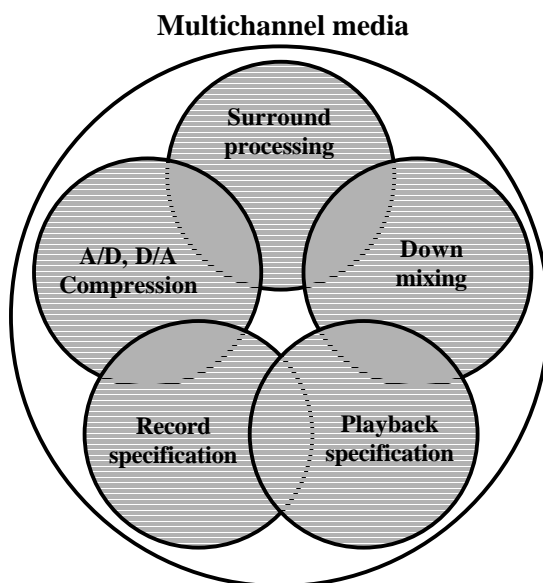
** USA, Korea, etc.

*** Europe, etc.

[Table 1] Multi-channel formats and standards organizations

Each format of multi-channel media is characterized by a combination of “surround processing method,” “encoding and compression method,” “recording response,” and “playback response.”

Many of these media formats include “down-mixing” specifications to allow two-channel playback.



[fig. 3] Factors that feature multichannel media

Currently, the following major multi-channel formats exist as mass consumer media.

Media	Video cassette tape, etc.	
Method	3-1 matrix	
Name	Dolby Surround	DTS Stereo
Manufacturer, Organization	Dolby lab.	DTS
Surround processing method	4-2 Matrix Encode	4-2 Matrix Encode
Compression method	–	–
Recording response (media)	L,C,R: full range S: 100Hz–7kHz	L,C,R: full range S: 100Hz–7kHz
Playback response (speaker, amp)	L=C=R=S(LS+RS) L,C,R,S: full range	L=C=R=S(LS+RS) L,C,R,S: full range

Media	Film	
Method	3-1 matrix	
Name	Dolby Stereo	DTS Stereo
Manufacturer, Organization	Dolby lab.	DTS
Surround processing method	4-2 Matrix Encode	4-2 Matrix Encode
Compression method	–	–
Recording response (media)	L,C,R: full range S: 100Hz–7kHz	L,C,R: full range S: 100Hz–7kHz
Playback response (speaker, amp)	L=C=R=S(LS+RS) L,C,R,S: full range	L=C=R=S(LS+RS) L,C,R,S: full range

Method	5.1 discrete	
Name	Dolby DIGITAL	DTS
Manufacturer, Organization	Dolby lab.	DTS
Surround processing method	–	–
Compression method	Dolby AC-3	APT-X100
Recording response (media)	L,C,R,LS,RS: full range LFE: to 120Hz	L,C,R: full range LS,RS: 80Hz– LFE: to 80Hz
Playback response (speaker, amp)	L=C=R: full range LS=RS=–3dB: full range LFE=+10dB*: 20–120Hz *in-band gain	L=C=R: full range LS=RS=–3dB: full range LFE=+10dB*: 20~120Hz *in-band gain
Method	6.1 matrix	
Name	Dolby DIGITAL Surround EX	DTS-ES Matrix
Manufacturer, Organization	Dolby lab.	DTS
Surround processing method	LS,RS: 4-2 Matrix encode	LS,RS: 4-2 Matrix encode
Compression method	Dolby AC-3 (L,C,R,LFE) Surround back channel- Encode (LS, RS)	APT-X100
Recording response (media)	L,C,R,LS,RS,BS: full range LFE: to 120Hz	L,C,R: full range LS,RS,BS: 80Hz– LFE: to 80Hz
Playback response (speaker, amp)	L=C=R: full range LS=RS=–3dB: full range LFE=+10dB*: 20–120Hz *in-band gain	L=C=R: full range LS=RS=–3dB: full range LFE=+10dB*: 20–120Hz *in-band gain

Media	DVD-Video		
Method	3-1 matrix		3-1 discrete
Name	Dolby Surround	DTS Stereo	Dolby DIGITAL
Manufacturer, Organization	Dolby lab.	DTS	Dolby lab.
Surround processing method	4-2 Matrix Encode	4-2 Matrix Encode	–
Compression method	–	–	Dolby AC-3
Recording response (media)	L,C,R: full range S: 100Hz–7kHz	L,C,R: full range S: 100Hz–7kHz	L,C,R,LS,RS: full range LFE: to 120Hz
Playback response (speaker, amp)	L=C=R=S(LS+RS): full range	L=C=R=S(LS+RS): full range	L=C=R=S(LS+RS): full range

Method	5.1 discrete		
Name	Dolby DIGITAL	DTS	
Manufacturer, Organization	Dolby lab.	DTS	
Surround processing method	–	–	
Compression method	Dolby AC-3	DTS Coherent Acoustic	
Recording response (media)	L,C,R,LS,RS: full range LFE: to 120Hz	L,C,R,LS,RS: full range LFE: to 120Hz	
Playback response (speaker, amp)	L=C=R=LS=RS: full range LFE=+10dB*: 20–120Hz *in-band gain	L=C=R=LS=RS: full range LFE=+10dB*: 20–120Hz *in-band gain	
Method	6.1 matrix		6.1 discrete
Name	Dolby DIGITAL Surround EX	DTS-ES Matrix	DTS-ES Discrete
Manufacturer, Organization	Dolby lab.	DTS	DTS
Surround processing method	LS,RS: 4-2 Matrix encode	LS,RS: 4-2 Matrix encode	–
Compression method	Dolby AC-3 (L,C,R,LFE) Surround back channel- Encode (LS, RS)	DTS Coherent Acoustic	DTS Coherent Acoustic
Recording response (media)	L,C,R,LS,RS,BS: full range LFE: to 120Hz	L,C,R,LS,RS,BS: full range LFE: to 120Hz	L,C,R,LS,RS,BS: full range LFE: to 120Hz
Playback response (speaker, amp)	L=C=R=LS=RS=BS: full range LFE=+10dB*: 20–120Hz *in-band gain	L=C=R=LS=RS=BS: full range LFE=+10dB*: 20–120Hz *in-band gain	L=C=R=LS=RS=BS: full range LFE=+10dB*: 20–120Hz *in-band gain

Media	Music	
Method	5.1 discrete (6ch)	
Name	DVD-Audio	SACD
Manufacturer, Organization	DVD Forum WG-4, ISC	Sony, Phillips
Surround processing method	–	–
Compression method	MLP (Packed PCM)	Direct Stream Transfer
Recording response (media)	L,C,R,LS,RS: full range LFE: full range	L,C,R,LS,RS: full range LFE: full range
Playback response (speaker, amp)	L=C=R=LS=RS=LFE: full range	L=C=R=LS=RS=LFE: full range
Other methods	3-1, 4 (2-2) etc.	

Media	Digital broadcast		
Method	5.1 discrete		
Main countries	Japan, etc.	USA, Korea, etc.	Europe, etc.
Name	BS digital	Dolby DIGITAL	–
Manufacturer, Organization	ISO, IEC	Dolby lab.	ISO, IEC
Surround processing method	–	–	–
Compression method	MPEG2-AAC	Dolby AC-3	MPEG2
Recording response (media)	L,C,R,LS,RS,LFE: full range	L,C,R,LS,RS: full range LFE: to 120Hz	L,C,R,LS,RS,LFE: full range
Playback response (speaker, amp)	L=C=R=LS=RS: full range LFE: not specified, SMPTE compliant	L=C=R=LS=RS: full range LFE: +10dB* 20–120Hz *in-band gain	L=C=R=LS=RS: full range LFE: depends on implementing group
Other methods	3-1, 4 (2-2) etc.		

Media	Games	
Method	5.1 discrete	
Name	Dolby DIGITAL	DTS
Manufacturer, Organization	Dolby lab.	DTS
Surround processing method	–	–
Compression method	Dolby AC-3	DTS Coherent Acoustic
Recording response (media)	L,C,R,LS,RS: full range LFE: to 120Hz	L,C,R,LS,RS: full range LFE: to 120Hz
Playback response (speaker, amp)	L=C=R=LS=RS: full range LFE: +10dB* 20–120Hz *in-band gain	L=C=R=LS=RS: full range LFE: +10dB* 20–120Hz *in-band gain
Other methods	Interactive, etc.	

[Table 2] List of multi-channel formats (typical examples)

2-1. Surround processing methods

There are two types of surround processing method; “matrix” and “discrete.”

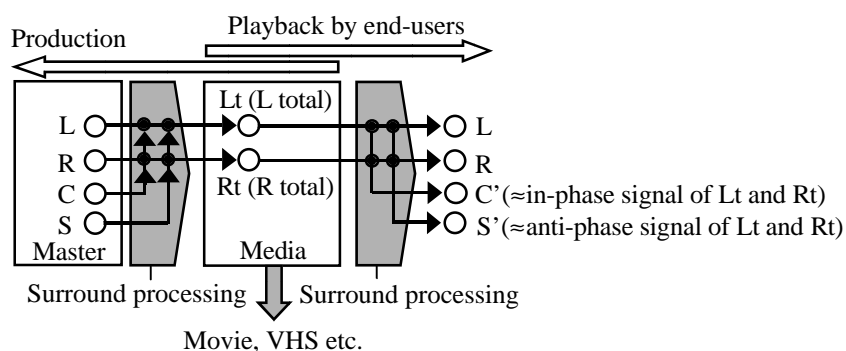
2-1-1. Matrix

This method uses phase synthesis technology to record a larger number of channels on a limited number of tracks.

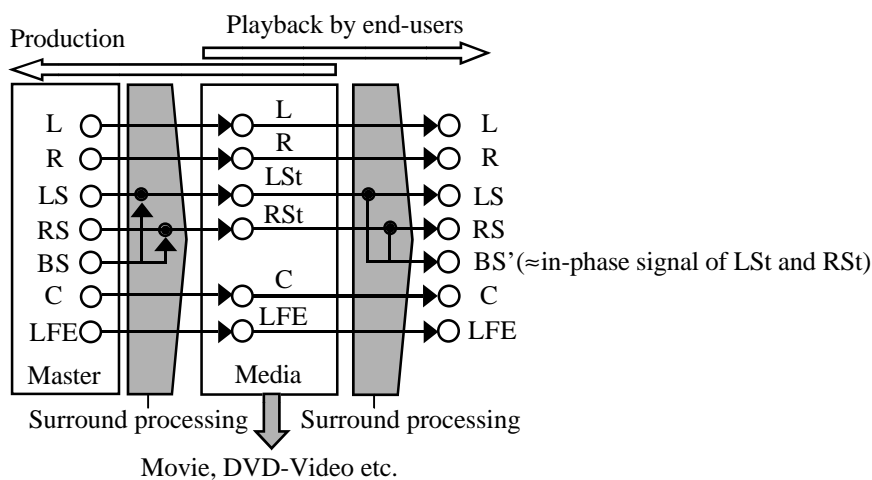
This means that for some channels, there may be restrictions in playback bandwidth and channel separation (crosstalk).

Matrix processing is often used for analog recording where the number of tracks is limited, such as for the analog tracks of a film, or on video cassette tape.

However in principle, it could also be applied to digital media such as CD.



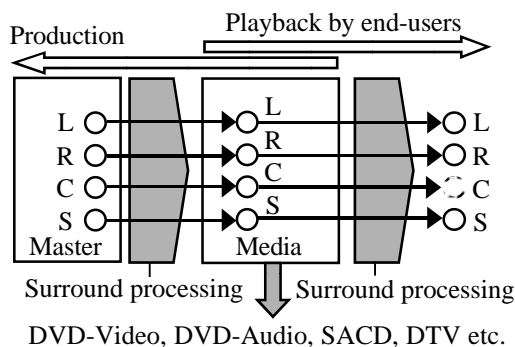
[Fig. 4] 3-1Matrix



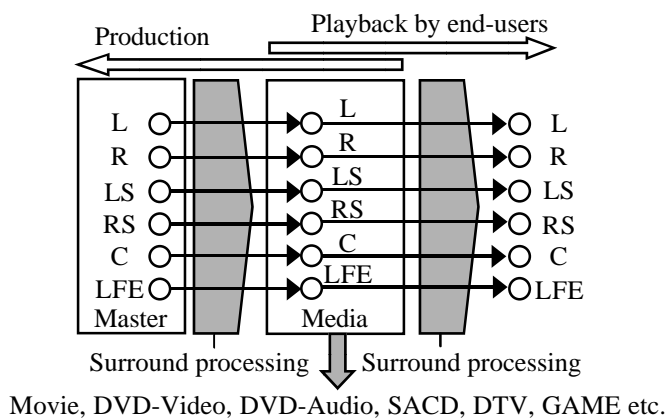
[Fig. 5] 6.1Matrix

2-1-2. Discrete

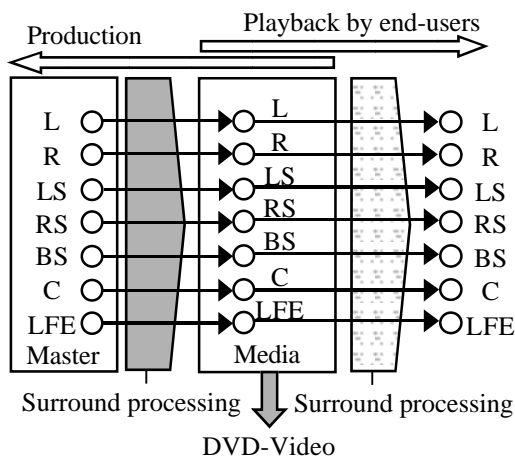
This method allows each channel to be recorded as a completely independent track. This became possible with the advent of high-capacity media such as DVD, and with the advance of digital compression technology.



[Fig. 6] 3-1Discrete



[Fig. 7] 5.1Discrete



[Fig. 8] 6.1Discrete

2-2. Encoding and compression methods

2-2-1. Encoding methods

There are two encoding methods; PCM and DSD.

PCM (Pulse Code Modulation) is a method of digital recording in which the A/D and D/A quality is determined by f_s (sampling frequency) and Q_b (quantization bit depth), and is used widely on a variety of digital media.

When PCM is recorded on the media in uncompressed form, it is often referred to as LPCM (Linear PCM).

DSD (Direct Stream Digital) is a method that uses 1-bit high-speed sampling to digitally record the sound in a form close to the compression wave that is the physical character of sound.

In consumer media, this method is used by the SACD (Super Audio CD), which claims a frequency response of DC~100 kHz and a dynamic range of greater than 120 dB.

2-2-2. Compression methods

Compression methods can be broadly divided into two types; lossy compression and lossless compression.

With lossy compression, the original signal cannot be recovered in its entirety from the compressed signal that is recorded; i.e., this is irreversible compression.

This method generally takes advantage of psychoacoustic phenomena to lower the redundancy of the original signal, thus compressing it.

Lossless compression allows the original signal to be completely recovered from the compressed signal that is recorded; i.e., this is reversible compression. This method is used to compress files on a computer. It uses mathematical means to lower the redundancy of the original signal, compressing it.

Thus, lossless compression delivers a lower compression ratio than lossy compression.

Examples of lossy compression	Method	Dolby AC-3, DTS Coherent acoustic, MPEG (-AAC), etc.
	Media	Film, DVD-Video, digital broadcast, games, etc.
Examples of lossless compression	Method	MLP (PPCM: Packed PCM), DST (Direct Stream Transfer)
	Media	DVD-Audio, SACD

2-3. Recording response

By “recording response” we mean the response when the master tape produced by the studio is recorded onto the production target media.

The response of each channel recorded on the media will depend on the encoding method and compression method as described above.

In the case of analog recording, the response will depend on the specifications of the recording media.

In other words for most types of media, all channels are recorded with full-range recording.

However in the case of LFE and surround channels, there will be differences depending on the media.

2-3-1. LFE channel

For media that is recorded in Dolby DIGITAL, such as film and DVD-Video, the bandwidth is restricted to 120 Hz at the time of recording.

This also applies to DTS. (However in film, the range to 80 Hz is the DTS recording range.)

For other media (DVD-Audio, SACD, MPEG2, MPEG2-AAC), full-range recording is allowed for the LFE channel in the same way as for the main channels.

2-3-2. Surround channels (S, LS, RS, BS)

For 3-1 matrix, the recording bandwidth of the S channel is restricted to 100 Hz–7 kHz.

For DTS in film (5.1, 6.1), the recording bandwidth of the surround channels (LS, RS, BS) is specified as 80 Hz and above, but since sound that lies below this point on the master tape is collectively recorded on the LFE channel, the resulting playback is full-range. This is what is known as “bass management” (described below in section 4).

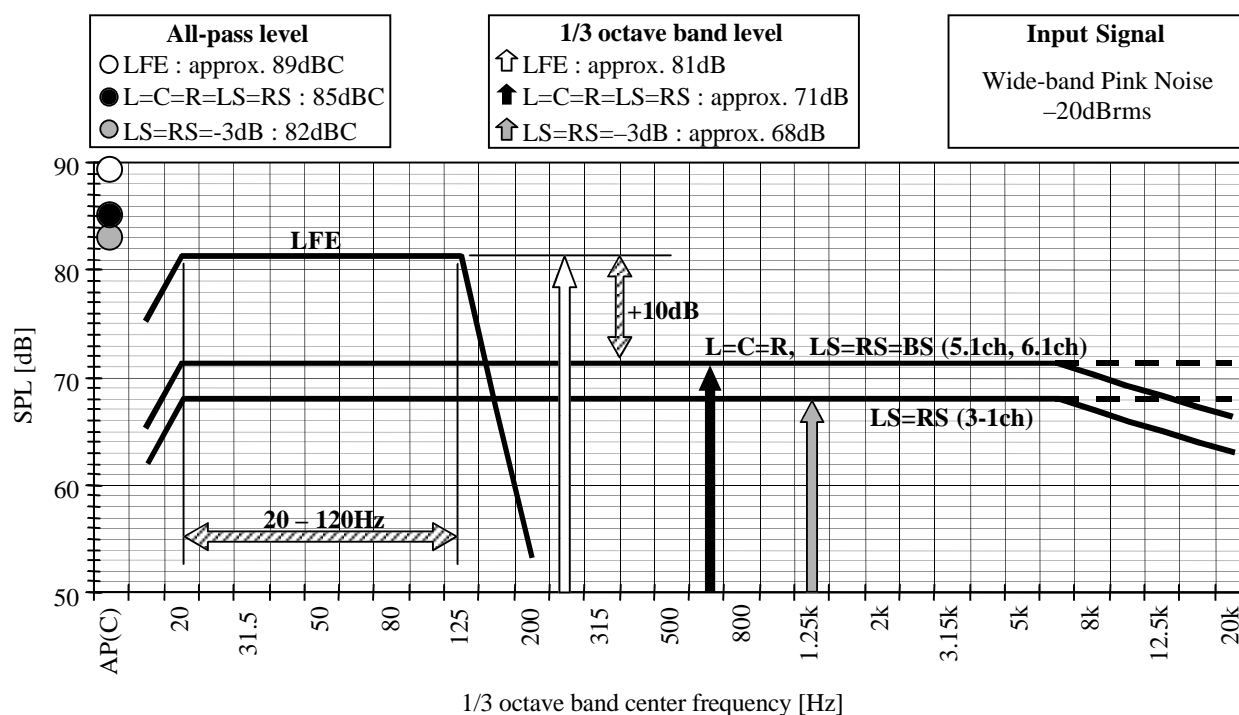
2-4. Playback response

By “playback response” we mean the desired (recommended) response of the playback system that plays back the media. For example, this corresponds to the frequency response of each speaker and the level balance.

It is important to be aware that depending on the media and the channel format, playback response may not be the same as the recording response.

The following pages describe playback response for typical media.

2-4-1. DVD-Video: Dolby, DTS



[Fig. 9] Playback specification for DVD-Video program

[Front channel]

■ Level

L = C = R (= 85 dBC)

Match the playback level of all channels.

■ Playback bandwidth

Full-range

[Surround channels]

■ Level

3-1: S (LS+RS) = L/C/R

Set the LS and RS playback levels lower than for 5.1 (LS = RS =: 82 dBC)

5.1: LS = RS = L/C/R (= 85 dBC)

6.1: LS = RS = BS = L/C/R (= 85 dBC)

■ Playback bandwidth

3-1: In the case of matrix, 100–7 kHz (it is best to use full-range speakers)
 In the case of discrete, full-range

5.1: Full-range

6.1: Full-range

[LFE channel]

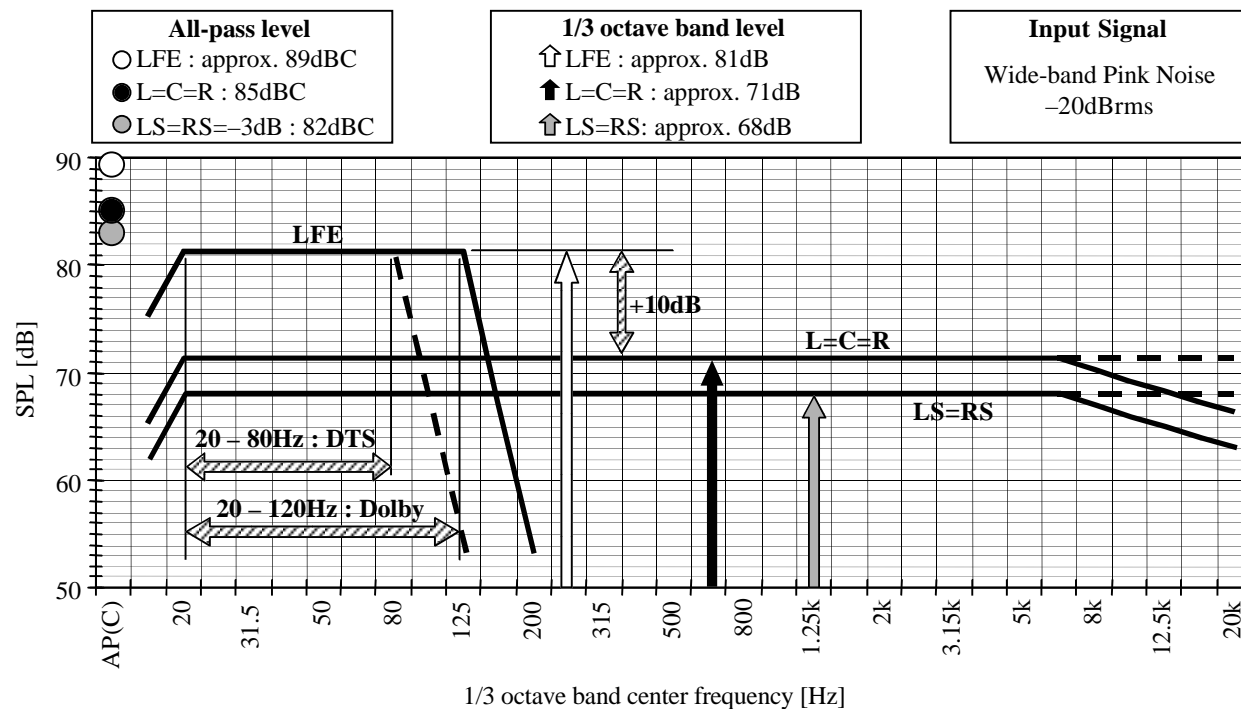
■ Level

“Band level” is +10 dB compared to the main channel.

■ Playback bandwidth

(20 Hz) – 120 Hz

2-4-2. Film: Dolby, DTS



[Fig. 10] Playback specification for Movie program

Differences with DVD-Video are as follows.

[Level balance of the surround channels]

For film productions, set the playback level of the surround channels at -3 dB relative to the front channels.

In the case of L = C = R = 85 dBC,

3-1: LS = RS = 82 dB; in other words, S (LS+RS) = 85 dBC

5.1: LS = RS = 82 dBC

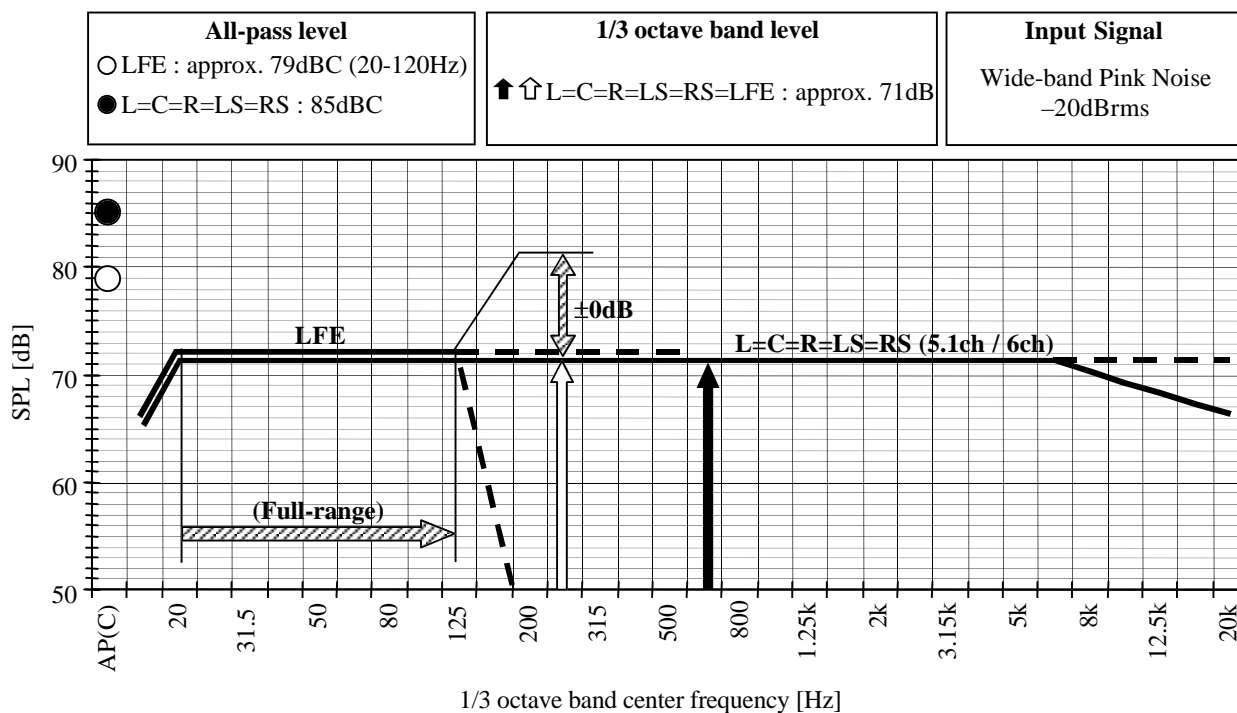
6.1: LS = RS = BS = 82 dB

[X Curve of the B-Chain: SMPTE 202M-1998]

In a large space such as a movie theater or dubbing studio, the B-chain curve is generally used as the standard for playback response (frequency response). However in a medium or small studio, the same flat response as described for DVD-Video is generally used even when creating film productions.

2-4-3. Music: DVD-A, SACD

The 5.1 channel (6 channel) playback response for DVD-A or SACD is shown below.



[Fig. 11] Playback specification for Music program (DVD-Audio, SACD)

[Front channel]

- Level
L = C = R
- Playback bandwidth
Full-range

[Surround channel]

- Level
5.1: LS = RS = L/C/R
- Playback bandwidth
5.1: Full-range

[LFE channel]

- Level
In general, band level ± 0 dB (same as main channels).
- Playback bandwidth
Not specified (full-range is possible)

2-4-4. Broadcast: Dolby DIGITAL, MPEG2, MPEG2-AAC

In the case of Dolby DIGITAL, the response is the same as for DVD-Video.

In the case of MPEG2 and MPEG2-AAC, the specification depends on the implementer (e.g., broadcast organization), particularly for the handling of the LFE channel.

2-4-5. GAME

Audio for games falls in two categories; multi-channel playback for the “movie” portion of role-playing games etc., and “interactive” multi-channel playback that occurs in response to movements within the game.

These multi-channel formats will depend on the audio processing method used by each manufacturer. Currently, Dolby DIGITAL or DTS are widely used.

In this case, the playback environment will be as described for DVD-Video.

2-5. Down-mixing

Most multi-channel media requires two-channel playback.

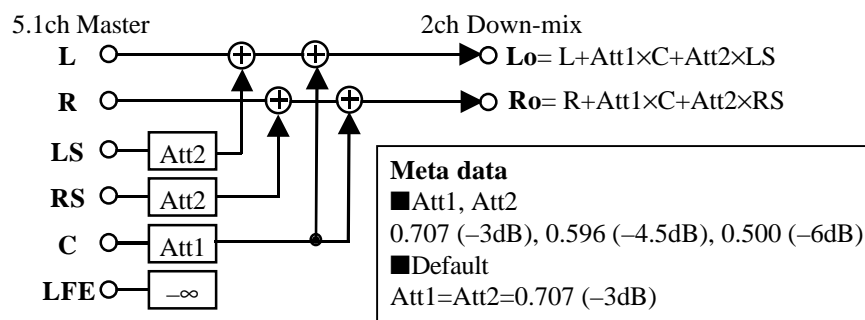
There are two possible ways in which a multi-channel production can be mixed to two channels.

One way is to generate a separate two-channel mix using the individual musical materials (stems) that were used for multi-channel mixing.

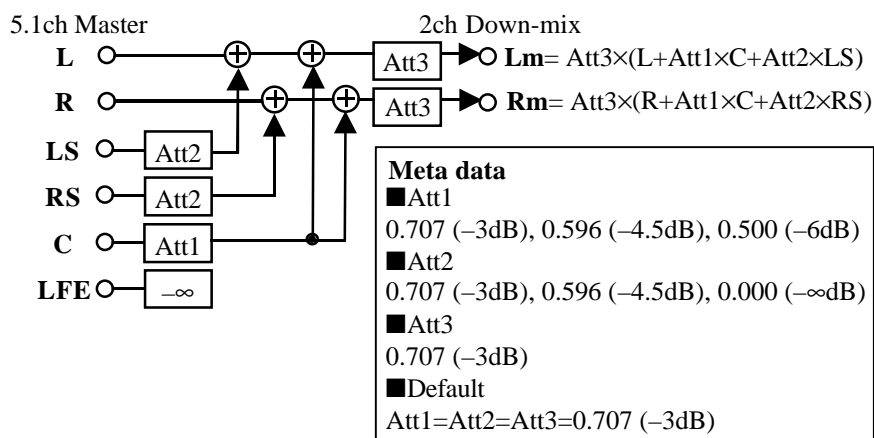
The other way is to use electrical circuitry to forcibly create the two-channel program (fold down).

The fold-down algorithm is defined for each type of media, and the production side must store attenuator values etc. on the media as meta-data.

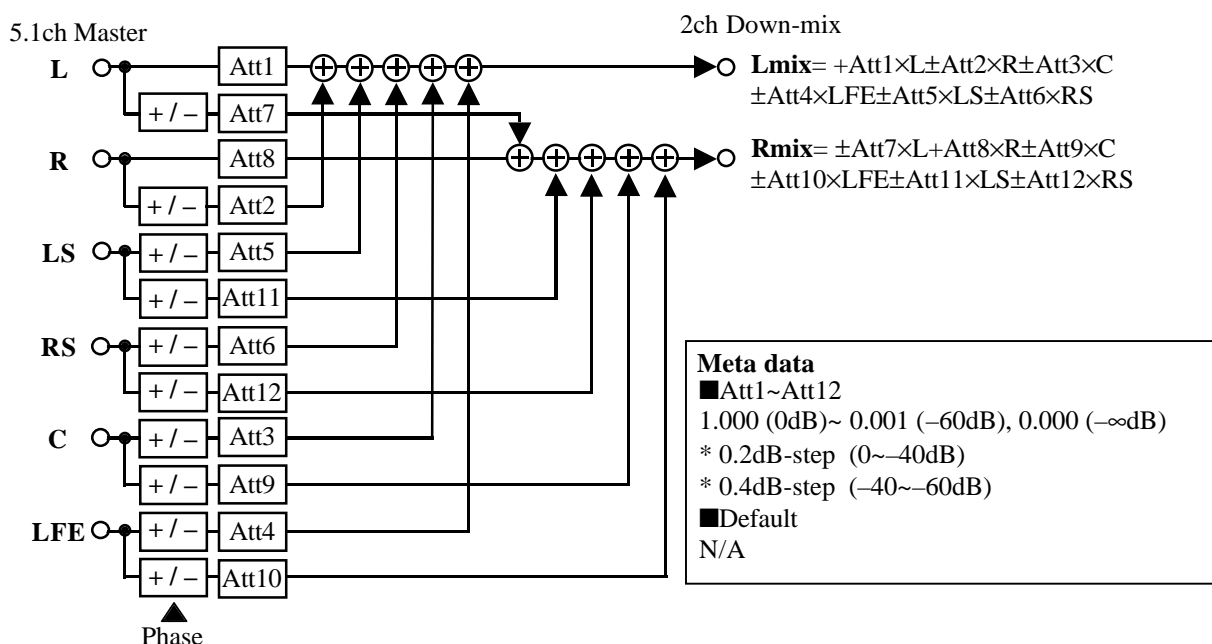
Typical examples of fold-down are shown below.



[Fig. 12] Flow of a Down mixing : DVD-Video (Dolby DIGITAL)



[Fig. 13] Flow of a Down mixing : MPEG2 (-AAC)



[Fig. 14] Flow of a Down mixing : DVD-Audio

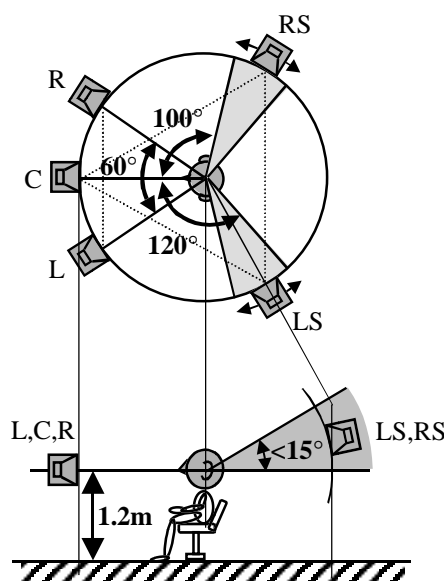
3. Playback environment

The playback environment consists of two aspects; room acoustics (which include the room shape, absorptivity, reflectivity, and dispersion characteristics), and speaker placement.

This chapter will discuss speaker placement.

Discussions of music-related media commonly refer to Rec. ITU-R BS. 775-1 ([Fig. 15]) recommendations. For other media as well, references are often made to ITU-R standards, or to compliance with the above-discussed DVD-Video environment.

Incidentally, the ITU-R speaker configuration is a proposed recommendation (Rec. = recommendation) from the International Telecommunication Union – Radio Communication Sector, and is not proposed as a “standard.”



[Fig. 15] Loudspeakers' placement :Rec. ITU-R BS. 775-1

The ideal speaker placement will depend on the size of the room, the monitoring radius (the distance from the speakers to the listening point), and the acoustical treatment of the room (absorption, dispersion, etc.). Thus, decisions regarding speaker placement must take into account both the character of the media produced in the studio and the physical environment of the studio (the size of the space, the monitoring radius).

Speaker placement is determined largely by two factors; the angle of L-R separation, and the placement of the surround speakers.

3-1. L, R

We will consider two angles of separation for the L-R speakers; 60° and 45°.

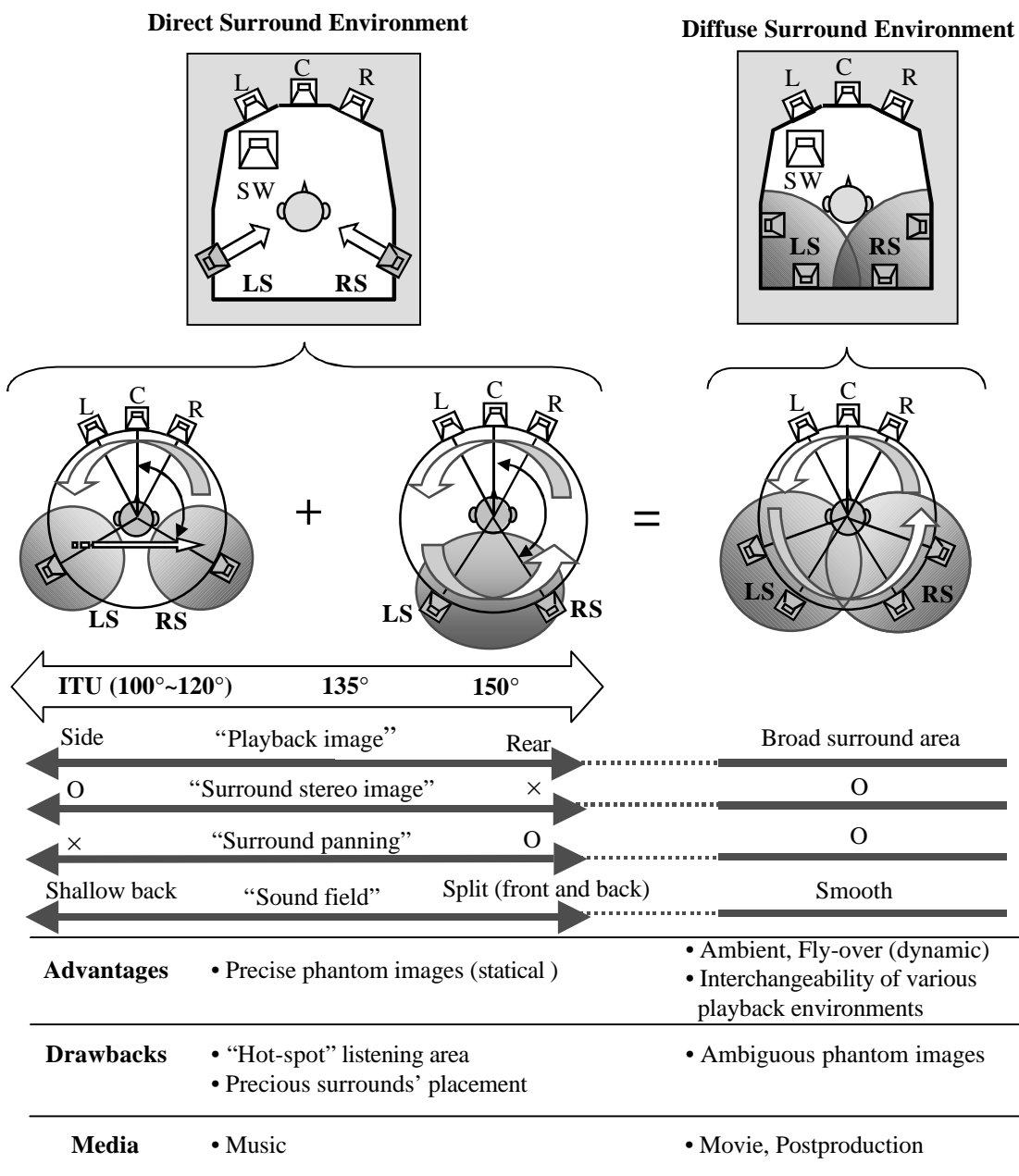
If emphasizing compatibility with conventional two-channel systems such as used for music playback, we give priority to the 60° placement. If we are emphasizing post-production for TV or movie theater as the end-user playback environment, we give priority to the 45° placement.

3-2. LS, RS

For the surround speakers (LS, RS), we have two types; a “direct surround” environment or a “diffused surround” environment ([Fig. 16]).

Direct surround is a method in which one pair of surround speakers is aimed directly at the listening point. ITU-R placement is an example of this.

Diffused surround, on the other hand, does not have pin-point sound source localization for the surround speakers. It is a placement method for expanding the coverage area. Movie theatres are an example of this.



[Fig. 16] Direct surround environment, Diffuse surround environment

3-2-1. Direct surround

In the case of direct surround, the placement of the surround speakers involves a trade-off between “surround panning” and “sense of rear stereo.”

3-2-1-1. ITU-R: 100°–120°

In the ITU-R placement, which locates the surround speakers at the “side” rather than at the “rear,” there is good left/right separation between the surround speakers, and it is easy to produce a phantom image. However, surround panning is typically limited to expressions such as blurring the image behind the listener’s head, and it is not easy to produce surround panning expressions with depth. (In other words, sound-source movement via surround panning does not describe a circle.)

3-2-1-2. 135°

In order for a sound source to be perceived as being “behind” rather than “beside” the listener, it is said that the surround speakers need to be placed at least 135° to the rear.

In most households, it is common for the speakers to be placed not at the “side” as in ITU-R, but rather “behind” at approximately 135°.

If the surround speakers are to be placed “behind,” or if end-user compatibility is to be emphasized, it is best to place the speakers at the 135° position.

3-2-1-3. 150°

If you require that the surround L and R have the same acoustical conditions as the main channel L and R, placing the surround speakers at 150° will produce a placement that is completely equivalent between front and rear.

With such a placement, L/R and LS/RS will have the same angle of separation, and it will be easy to move the sound in a 360° path by surround-panning.

However, as the surround speakers are placed farther to the rear, the surround sound field will tend toward monaural, and there will be a more distinct separation between the front and rear sound fields.

3-2-2. Diffused surround

The most common method of creating diffused surround is to use several surround speakers (side + rear). With such a configuration, it is possible to create a monitoring environment that allows both a sense of stereo and 360° surround panning.

3-2-3. Direct surround and diffused surround

The advantage of direct surround is that you can easily create precise phantom images (static) in a 360° range.

Thus, direct surround is often used in a production environment for musical content, such as DVD-Audio or SACD.

However, precise compatibility of surround speaker placement is necessary in order to precisely reproduce the phantom sound field.

As described above, direct surround is suitable as a production environment for media that requires “ultimate” aural expression.

On the other hand, diffused surround excels in delivering ambient or fly-over sounds, and allows surround panning to create 360° audio source movements. Thus, it is often used as the production environment for multi-channel media that accompanies video.

Another advantage when playing back content created in a diffused surround environment is that precise compatibility of surround speaker placement is not required.

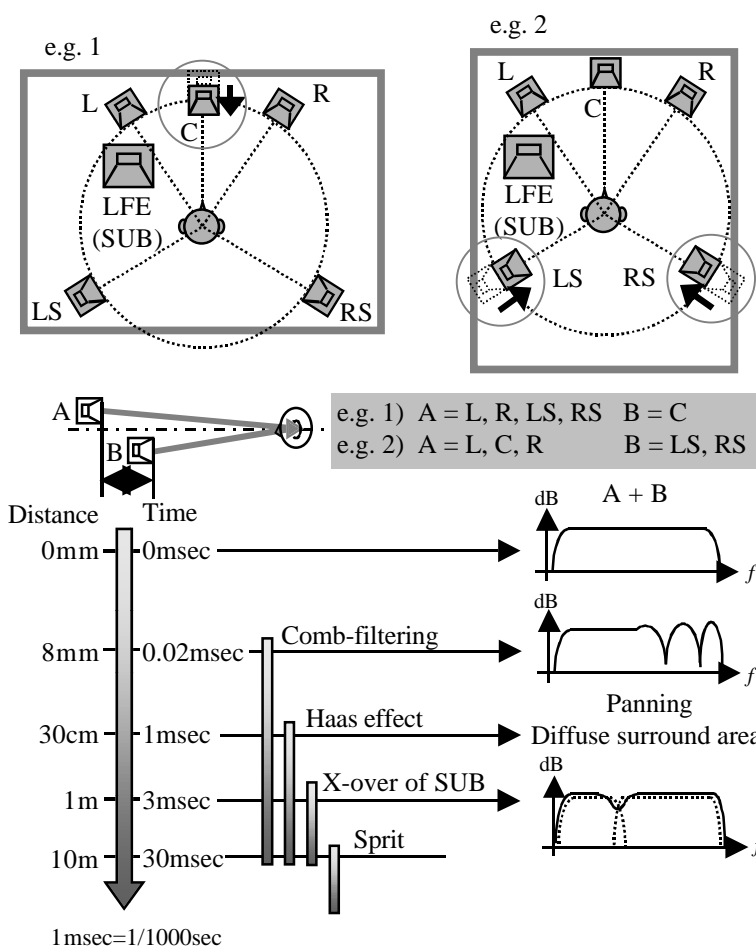
Thus, we can say that diffused surround is suitable as an environment for efficient production of “general-purpose” program material.

3-3. SUB

When placing the sub-woofer, we must take the acoustics of the room into account. For example if we place the sub-woofer in the corner of the room, we will get plenty of power, but the frequency response will tend to be disrupted by standing waves. When placing the sub-woofer, we must consider both the playback power and the frequency response.

3-4. Monitor alignment

In some cases, problems with the size or shape of the studio will mean that it is not possible to place all speakers at an equal distance from the listening point. Such problems can occur particularly when partially renovating a two-channel studio for multi-channel support. In general, the center speaker is placed closer than the L/R speakers, and next the surround speakers are often placed closer. Under such conditions, the following three monitoring problems can occur.



[Fig. 17] Monitoring errors caused by a difference of a monitoring distance

3-4-1. Comb filtering

If the same sound is played back from two speakers whose distance to the listening point differs by 8 mm or more, dips will occur in the frequency region below 20 kHz. Based on the speed of sound, a distance discrepancy of 8 mm is equivalent to a slight time difference equal to 0.025 msec. In addition to physical distance, such delay differences can also occur due to wiring or electrical delay within equipment.

3-4-2. Haas effect

If the surround speakers (LS, RS) are placed more than 30 cm closer than the front speakers (L, C, R), the sound source movement when you surround-pan from surround -> front will not be heard smoothly because the perceptual panning is pulled strongly toward the surround speakers.

Another problem is that in a diffused surround environment, the surround coverage area may not be wide enough, causing the perceived sound image to be located only around the nearest surround speaker.

3-4-3. Crossover with the sub-woofer

If there is more than 1 meter of difference between the distance from the sub-woofer to the listening point and the distance from the other speakers to the listening point, dips are likely to occur in the combined response.

Prominent dips often occur in the region of the sub-woofer cutoff frequency.

If the monitor system uses bass management (discussed below), especial care must be taken to avoid significantly impairing the frequency response of the main channels.

If the above monitoring problems occur, you will need to reconsider the speaker placement, and try adjusting the speaker phase (in particular, the sub-woofer).

If these adjustments do not alleviate the problem, you will need to apply a delay device to each speaker.

In addition to delay, designing your monitor system so that an attenuator or GEQ (PEQ) can be applied to each speaker often provides useful ways to adjust the monitoring response.

3-5. THX[®] pm3[™]

At present, a variety of multi-channel playback environments exist.

When deciding which playback environment you will ultimately construct, you must take into account overall considerations such as the media you will be producing, and the state of your room.

It is also important that the level balance and frequency response of each speaker in your multi-channel monitoring system be adjusted according to the media you are producing.

The THX[®] pm3[™] which THX announced in 1999 is a program for designing this type of small- to medium-sized multi-channel studio, and is currently the only comprehensive design policy.

4. Bass management

In a small- to medium-sized studio, room modes due to standing waves often become a problem, and it is easy for inconsistencies to develop in the low-frequency response of each speaker.

This can impair the following important requirements for multi-channel monitoring.

1. That all channels have a consistent response.
2. That the LFE playback level maintain a gain of the low-frequency response of the other channels +10 dB.

Thus, ensuring that the low-frequency response of each channel is consistent is one of the most important points for constructing a multi-channel monitoring environment.

This is why we need to consider some type of “bass management.”

In a small- to medium-sized studio, we can consider three methods of bass management; acoustical treatment of the room, speaker placement, and electro-acoustic methods.

4-1. Acoustical treatment of the room

Room acoustics can be treated by adding thick absorptive material.

In theory, an 85 cm or greater thickness of absorptive material is required in order to completely absorb low frequencies in the 100 Hz region.

For smaller rooms, physical considerations often make it difficult to install thick absorptive materials.

4-2. Speaker placement

The low-frequency response of a speaker has a closely-linked effect on the room acoustics.

Thus, consideration of the speaker placement is a useful way to improve the low-frequency playback response.

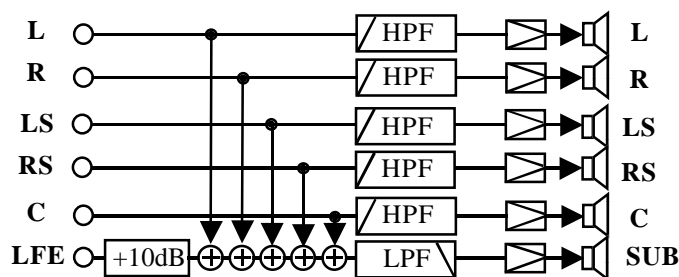
In many cases, we are able to consider only the placement of the sub-woofer, which allows a high degree of freedom in its placement.

Consideration of the sub-woofer placement in conjunction with the use of a bass management controller (discussed later) is one of the most effective ways in which a small- or mid-sized studio can improve its low-frequency response.

4-3. Electro-acoustic methods

The bass management controller shown in [Fig. 18] can be applied to a monitor system to implement electro-acoustic compensation.

In general, “bass management” refers to a bass management controller.



[Fig. 18] Bass management controller (1)

A bass management controller can not only improve the low-frequency response, but also allows us to check the playback of the bass redirection function (DVD-Video, Dolby DIGITAL).

Major advantages of a bass management controller are listed below.

- A. The low-frequency response of the main channels (L, C, R, LS, RS) can be made consistent. By ensuring that the low-frequency response (which is most prone to inconsistency) is consistent, this makes it easier to ensure that all channels have the same response.
 - B. By placing the sub-woofer in the optimal location, the low-frequency response of all channels can be improved.
 - C. The +10 dB band gain for the LFE channel can be ensured.
 - D. All channels can be given a playback response equivalent to large monitors that reproduce the ultra-lows starting at 20 Hz. This is particularly important for studios that do film production.
 - E. The playback result of consumer playback methods* can be checked.
- * The bass redirection function in DVD-Video (Dolby DIGITAL).

Precise filter specifications are crucial for a bass management controller.

A bass management system that simply imitates [Fig. 18] without considering the points discussed below is likely to create various monitoring problems, such as separation of sounds and unnatural localization of sound sources.

■ Low pass filter

[Cutoff frequency]

This must be set to a low frequency that has no sense of directionality.

However if the cutoff frequency is set excessively low, this will narrow the bandwidth that is handled by the sub-woofer, and the improvement in low-frequency response will be less.

In most cases, 80 Hz is set as the cutoff frequency.

[Slope]

If the slope is gradual, sounds higher than the above-specified cutoff frequency may be heard, and this will produce a sense of directionality from the sub-woofer.

Conversely if the slope is too steep, the sense that the sound of one channel is being played from one speaker will be diminished, and sounds will tend to split between the low range and mid/high range.

In most cases, a slope of -24 dB/octave is used.

■ High pass filter

[Cutoff frequency]

The identical cutoff frequency used by the low-pass filter is also used by the high pass filter.

The type of filter (Butterworth, Linkwitz, etc.) must also be identical to that used for the low-pass filter.

[Slope]

The slope must be such that it will cross optimally with the low-pass filter.

Here it is important to consider not only the response of the respective filters, but also the response of the speakers that are used.

For the purpose of discussing the slope of the high-pass filter, we will assume the following.

- LPF $fc = 80$ Hz, -24 dB/oct.
- HPF $fc = 80$ Hz

The playback bandwidth of the sub-woofer usually extends above the cutoff frequency of the low-pass filter that is used.

Therefore, this same " $fc = 80$ Hz, -24 dB/oct." will apply to the low-range response of the sub-woofer playback.

The cutoff response of the main speakers played via the high-pass filter must be targeted to this " $fc = 80$ Hz, -24 dB/oct."

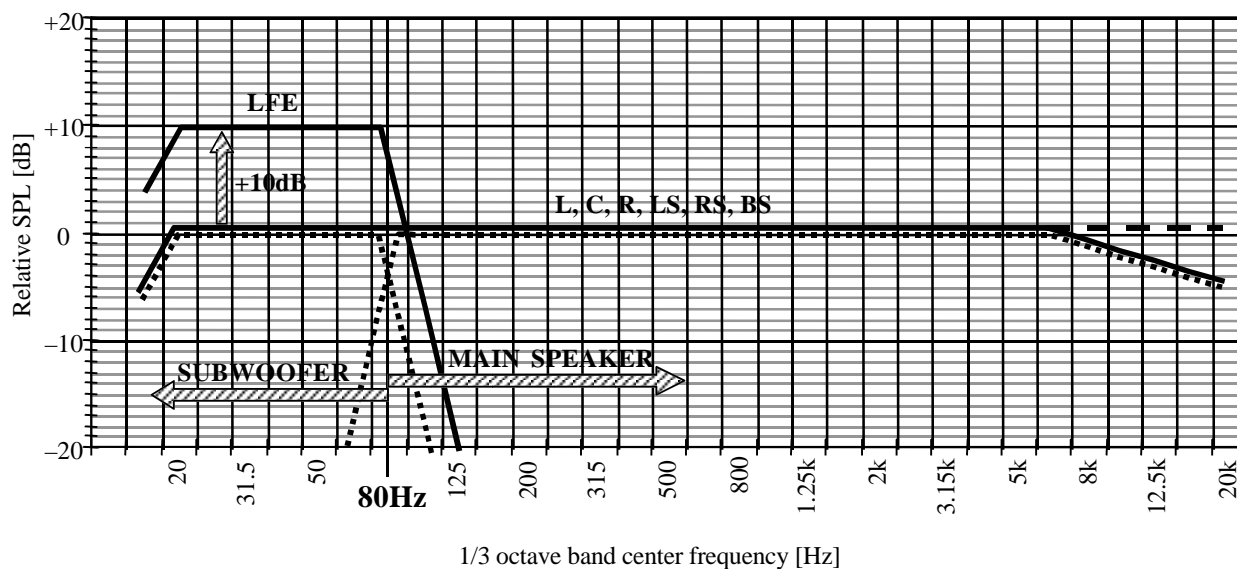
Example 1) If the main speakers are small speakers whose response falls off at 12 dB/oct. from 80 Hz, then "high-pass filter response" = 12 dB/oct.

Example 2) If the main speakers are large speakers that are able to reproduce below 80 Hz, then “high-pass filter response” = 24 dB/oct.

In this way, the slope of the high-pass filter will change depending on the response of the speakers that are used.

In specifications such as Dolby DIGITAL, a recording bandwidth up to 120 Hz is provided for the LFE channel. However in actual production situations, it is rare that signals above 80 Hz (which carry a sense of localization) are used in the LFE channel.

This means that in production processes such as mixing, a bass management controller with the following playback response [Fig. 18] can be used.

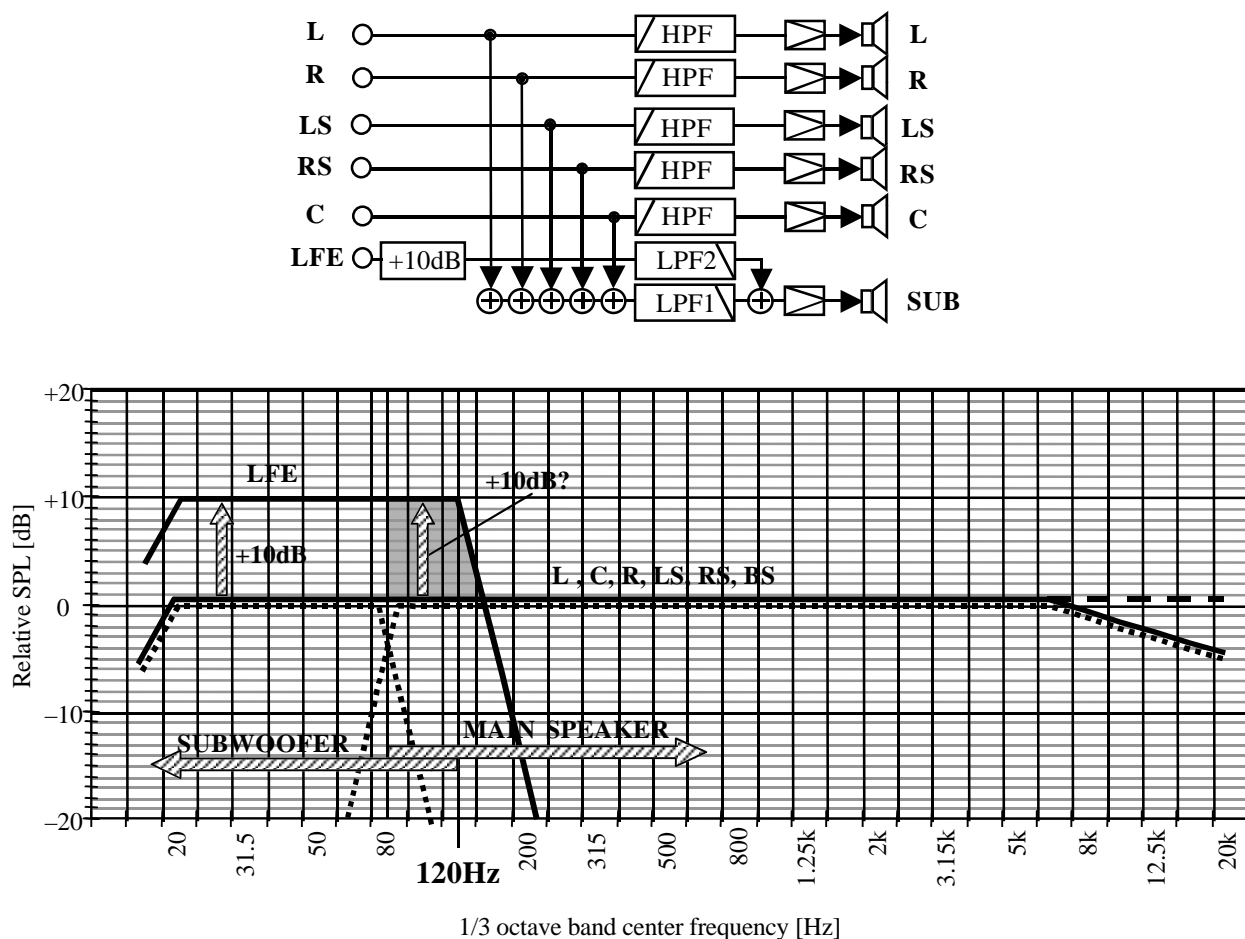


[Fig. 19] Playback characteristics using Bass management controller (1)

On the other hand if it is necessary to play back and confirm all signals included in the media, such as in movie theatres, screening rooms, and authoring/mastering studios, it is necessary to extend the playback bandwidth of the LFE channel to 120 Hz (Dolby DIGITAL, DTS) or above.

In this case, the bass management controller shown in [Fig. 20] is used.

Caution is needed here, since it is more likely that unevenness will occur in the LFE band gain (main channel + 10 dB) for the region above 80 Hz.



[Fig. 20] Bass management controller (2)

As discussed above, bass management specifications must be determined on the basis of an overall evaluation of numerous factors including the speakers used and the purpose of the studio.

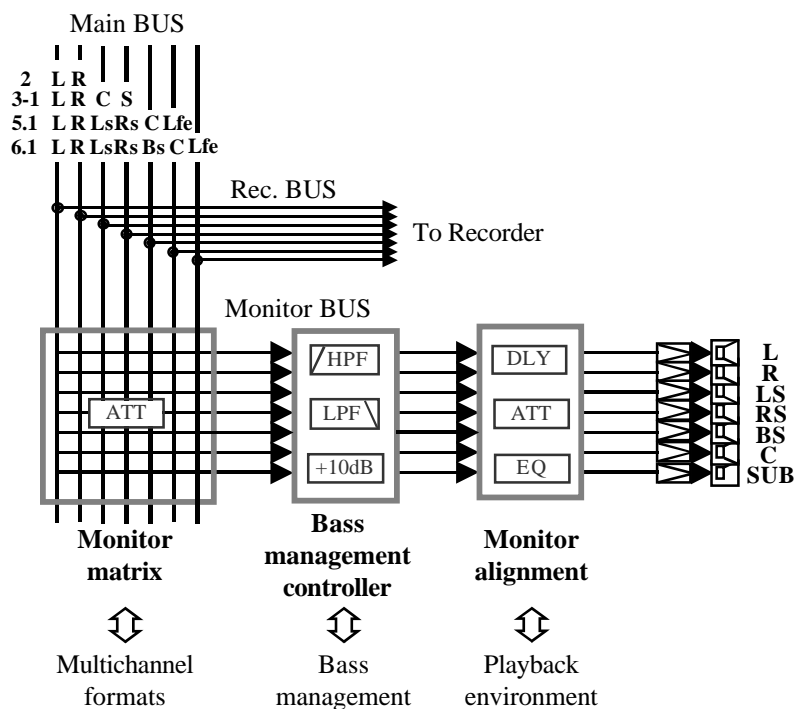
5. Monitor systems

An example of a multi-channel monitoring system is shown below.

The system shown below consists of three main sections; the monitor matrix, the bass management controller, and monitor alignment.

As mentioned at the beginning of this document, multi-channel monitoring involves three key elements, and these areas of functionality correspond to these elements.

Specifically, “monitor matrix” corresponds to “multi-channel format,” “bass management controller” to “bass management,” and “monitor alignment” to “playback environment.”



[Fig. 21] Flow of a multichannel monitoring system

5-1. Monitor matrix

A monitor matrix circuit is required if you will need to perform both 5.1 and 3-1 processing, or if you need to produce audio for different media such as DVD-Video and film even though both of these are 5.1. A monitor matrix is also required in order to audition the down-mixing functionality defined by DVD-Video (Dolby DIGITAL).

The monitor matrix is normally part of the mixing console’s functionality.

5-2. Bass management

In order for a small- or mid-sized studio to obtain a monitoring environment that manages the response down to the low frequency range, it is desirable that the monitor system include a bass management controller.

In the case of DVD-Video (Dolby DIGITAL) production as well, it is desirable that there be a bass management controller in order to verify the operation of bass redirection.

5-3. Monitor alignment

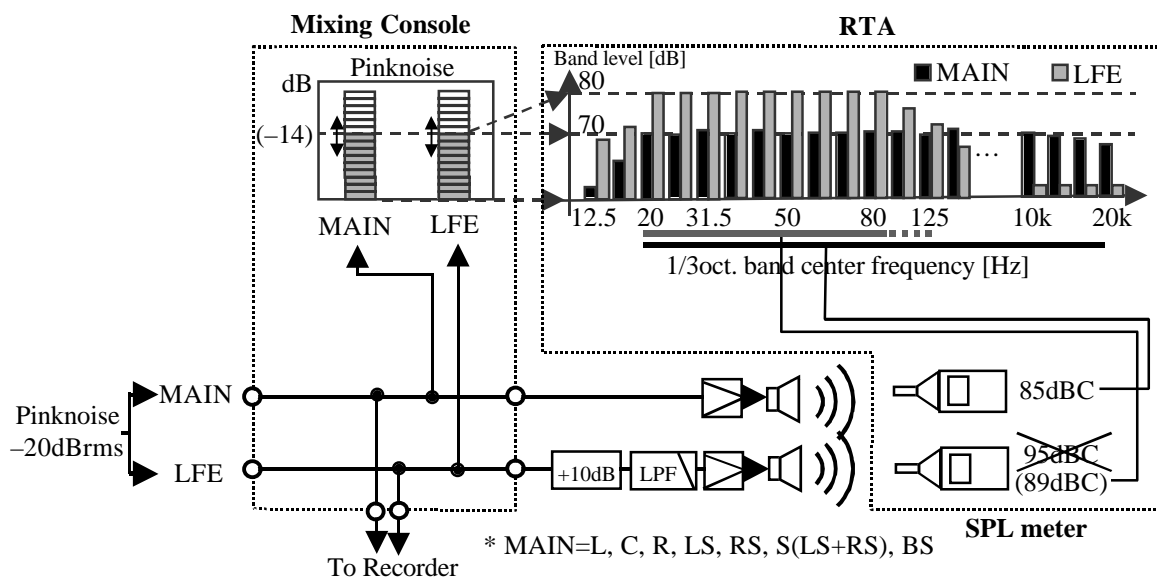
In order to correct the time alignment of each channel to a precision of 0.02 msec or better, it is desirable that a delay be provided for each speaker.
Considering the actual adjustment process, attenuators and GEQ (PEQ) will usually be necessary as well.

6. Measurement and adjustment

In order to set up a multi-channel monitoring environment, we need to make measurements and adjustments at the time of setup.

To measure the monitor response and make adjustments, we need pink noise as a source signal, and a sound pressure level meter to measure the playback sound pressure level of the speakers.

In most cases, a 1/3 octave realtime analyzer (RTA) is also necessary when making the actual measurements.



[Fig. 22] Measurement and tuning for DVD-Video

6-1. Test signal

-20 dBrms wide-band pink noise is used as the test signal for measurement.

Pink noise contains significant change in amplitude, and it is difficult to verify its input level using the level meters of the console. (The level changes intensively in the region of -14 dBFS.)

For this reason, we must obtain a pink noise test signal whose effective value is certified to be -20 dB.

6-2. Main channel level balance

Play back the above pink noise from the main speakers, and adjust the gain of each amp so that the sound pressure level of each speaker is 85 dBC at the listening point.

The sound pressure level (85 dBC) at the listening point is measured using an SPL meter ("slow" response, "C-weighted" frequency curve).

The sound pressure level indicated by the SPL meter is the "all-pass level" that is the sum of the levels of all bands.

If the all-pass level is 85 dBC, the band level of each band displayed in the RTA (the 1/3 octave band level) will be approximately 71 dB.

6-3. LFE channel level balance

For a DVD-Video (Dolby, DTS) or film production (Dolby, DTS, SDDS), adjust the amp gain so that the band level of the LFE channel is +10 dB relative to the main channel.

Note that if you make adjustments so that the SPL meter indicates a pink noise playback level of 95 dBC at the listening point, the adjustment will be incorrect.

The most reliable way is to use the RTA, and make adjustments so that the 1/3 octave band levels are approximately 81 dB. In this case, the all-pass level indicated by the SPL meter will be approximately 89 dBC, not 95 dBC.

In cases such as DVD-Audio or SACD, where you set the band level of the LFE channel to the same level as the band level of the main channels (± 0 dB), the all-pass level shown by the SPL meter will be approximately 79 dBC.

The all-pass level value is easily affected by the frequency response and playback bandwidth. For this reason, it is desirable to also use an RTA during the actual measurement, and use the octave band level to verify the level balance of each channel (particularly the LFE channel). Using the SPL meter to adjust the all-pass level is a simplified method whose accuracy is assured only if the playback conditions are ideal for all speakers.

6-4. Narrow-band pink noise

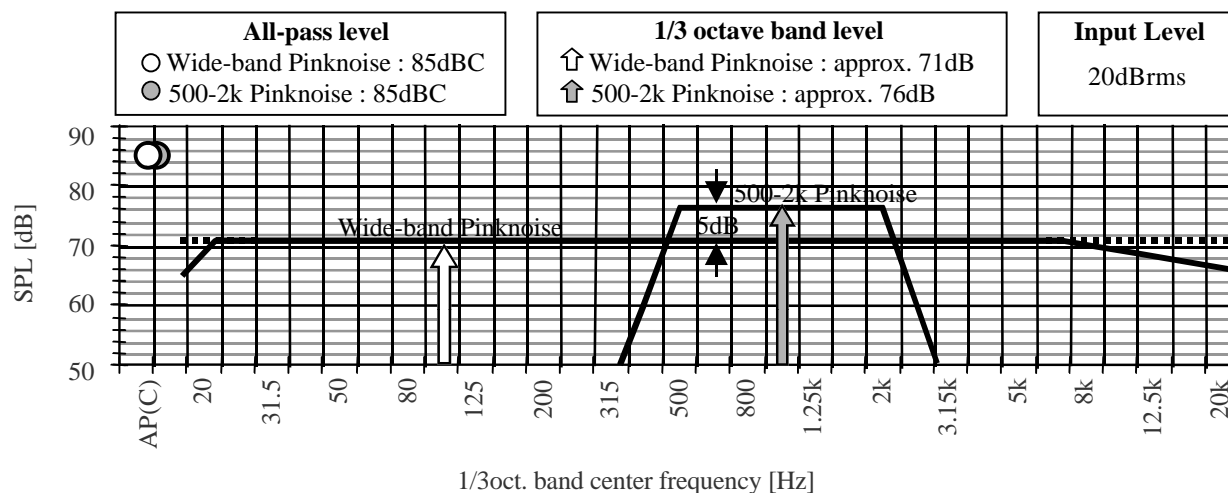
Listed below are some elements that can make the SPL meter give an unreliable reading for the all-pass level.

- Acoustical character of the room disrupted low-range response
- Playback bandwidth of the speakers low-range playback limit
- Response of the SPL mic high-range limit

If you suspect any of the above problems, you may be able to obtain good results by using pink noise that is band-limited to 500 Hz–2 kHz.

Since 500 Hz–2 kHz pink noise does not include the low- and high-range that can become problematic, it allows you to minimize the error for a simple measurement taken using only a SPL meter.

If you use 500 Hz–2 kHz pink noise, you must set the playback level of each channel to approximately 80 dBC rather than 85 dBC.



[Fig. 23] 500-2kHz Pinknoise and Broadband Pinknoise

* The 500–2k pink noise built into the DM2000 has already been level-adjusted for the 85 dBC adjustment. Thus on the DM2000, you simply need to make adjustments so that the pink noise playback is 85 dBC at the listening point regardless of whether you are using wide-band or narrow-band (500–2k).

The method described above is based on adjustments for movie theaters. In other words by defining absolute values for the source signal level and playback sound pressure level, this method ensures that the same program will always be played back at the same level in any theater. Outside of a movie theater it is not necessary to define the absolute level, so any desired value may be used for the pink noise input level and the sound pressure level at the listening point. The important thing is that the relative sound pressure balance be maintained.

In order to make accurate monitor adjustments, it is recommended that you use not only a SPL meter, but also an RTA in conjunction with this to make measurements and adjustments.

7. Summary

The following three points are the key aspects of constructing a multi-channel monitoring environment.

1. Clarify the media that is to be produced.
2. Understand the studio environment (size, absorptivity).
3. Make measurements and adjustments.

Planning for a monitor system begins with consideration of the above points, and determining the required equipment.

By considering the process of measurement and adjustment, we can create a wiring plan that includes the patching needed by the monitor system.

The monitor functionality of the console is a key point in the planning of such a monitor system.

If the console does not provide the monitoring functionality required in order to produce the desired type of multi-channel content, the missing functionality must be provided by external devices.

The DM2000 contains essentially all of the monitoring functionality required for producing multi-channel content.

1. Monitor matrix with a fold-down function is built-in.
=> Numerous multi-channel formats can be supported
2. Various types of bass management are built-in.
Up to two LS and RS speakers are supported as standard.
Monitor alignment (delay, attenuator) functionality is built-in.
=> A variety of studio environments can be supported.
3. Signals used for measurement and adjustment are built-in.
=> Measurement and adjustment can be performed immediately.

It is easy to learn the parameters of DM2000's multi-channel monitoring function and how to operate them.

However, the important thing is to understand the reasons that these parameters exist, and the “content” that lies behind them.

I hope this booklet will be helpful in expanding your understanding of these considerations.

8. Reference materials

- “Surround production handbook” (in Japanese)
Mick Sawaguchi, editor; Kenrokukan Publishing Company
- “Multichannel surround systems and operations”
AES TECHNICAL COUNCIL Document ESTD1001.0.01-05
<http://www.aes.org/>, <http://www.aes-japan.org/>
- “5.1 Surround Sound Up and Running”
Tomlinson Holman writing, Focal Press
- “Sound for Film and Television”
Tomlinson Holman writing, Focal Press
- “Sound recording technical principles” (in Japanese)
Japan Association of Professional Recording Studios (JAPRS)
- “Dictionary of acoustical terms” (in Japanese)
The Acoustical Society of Japan (ASJ), Corona publishing co.,ltd

- AES convention preprints and papers
<http://www.aes.org/>, <http://www.aes-japan.org/>

- Surround Professional Magazine
<http://www.surroundpro.com/>

- PROSOUND Vol.100,101,103,104,105,106,107,108 (in Japanese)
<http://www.jah.ne.jp/~prosound>

- ITU
<http://www.itu.int/>, <http://www.ituaj.jp/>
- SMPTE
<http://www.smpte.org/>
- ISO
<http://www.iso.ch/iso/en/ISOOnline.frontpage>
- MPEG
<http://mpeg.telecomitalia.com/>
- DVD Forum
<http://www.dvdforum.org/forum.shtml>
- Dolby lab.
<http://www.dolby.com/tech/>
- DTS
<http://www.dtsonline.com/>

- BBC Broadcast Research Center
<http://www.bbc.co.uk/rd/>

- NHK Broadcast Technology Research Center
<http://www.nhk.or.jp/str/>

- TMH Corporation
<http://www.tmhlab.com/>

- Miller & Kreisel Sound Corporation
<http://www.mkprofessional.com/>, <http://www.mksound.com/>

- THX
<http://www.thx.com/>

■ Author
Masataka Nakahara

SONA Corporation
Chief Acoustic Designer / Engineer
Officer of the AES Japan section

Masataka Nakahara received a master's degree of acoustic design from the graduate school of Kyushu Institute of Design (K.I.D) in Fukuoka Japan, in 1995.

He then joined SONA Corporation in Tokyo, and has been engaged in acoustic design work for acoustical architecture such as professional studios.

Besides this work, he has researched the acoustic environment in studios with K.I.D since 1997.

In 2001, he cooperated with YAMAHA in the development of a multichannel monitoring system for their brand new digital mixing consoles (DM2000, O2R96).

Recent publications (in Japanese)

■ Semi-monthly "PROSOUND"; Stereo Sound Publishing, inc.

• December 2000, February 2001

"The acoustic design of the Sony PCL 408 THX Suite (preface, afterword)"

• June, August, October, December 2001, February and April 2002

"Acoustic design for multi-channel studios"

■ Surround production handbook, Mick Sawaguchi, editor; Kenrokukan Publishing Company

"Chapter 3. Surround control room design, Small and medium post-production studio design"

■ Sound recording technical principles, Japan Association of Professional Recording Studios (JAPRS)

"Chapter 6. Next-generation acoustical technology: 6-2 Multi-channel playback"

■ SONA Corporation

2-19-9 Yayoi-cho Nakano-ku, Tokyo 164-0013, Japan

URL <http://www.sona.co.jp> (in Japanese)

E-mail info@sona.co.jp / thxpm3@sona.co.jp / mkpro@sona.co.jp

[Principal business]

Acoustic design / Acoustical survey and measurement / Acoustical consultation / Architectural design and construction / Specialty furniture design / Technical support for audio makers / Study and research with an acoustic lab / Official dealer for M&K Professional / International representation company for THXpm3 program