5.1-Channel Music Production Guidelines

Issue 3

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Issue 3
S05/14926/15996

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Chapter 1
Introduction

With the advent of high-quality multichannel consumer music formats, 5.1-channel music producers may benefit from a standardized set of production practices. This document guides the multichannel music professional, working in small- to medium-sized control rooms, in the production of high-quality, 5.1-channel music intended for playback in consumer environments. If followed, these guidelines will facilitate interchangeable critical listening judgments between various locations.

Please note that production practices for 5.1-channel music intended for playback in large cinemas are well-documented elsewhere, and that there are methods by which a home playback system can be made to emulate the cinematic playback experience. Neither of these topics is discussed in this document, which focuses on the 5.1-channel music experience afforded by the expanded artistic flexibility and palate of a 5.1-channel soundscape, as well as its delivery from the production studio directly to the home.

Many aspects of 5.1-channel music mixing are covered in this document, including monitoring, recording levels, practical setup guidelines, and program interchange standards. Subjects such as room design are discussed with recommended target parameter values.

This manual can be used as a quick setup reference (starting in Section 3.3) or as an in-depth sourcebook for the multichannel music creator.

This manual is written from the perspective that:

- 5.1-channel music intended for home enjoyment is a unique art with its own requirements, but these requirements can co-exist with 5.1-channel production practices already established for cinema or broadcast applications.
- It would be harmful to the advancement of 5.1-channel music if every industry using it made conflicting demands on the consumer playback environment.
- A universal multichannel sound system applicable to music, cinema, and broadcasting would be beneficial to the listener.
- Flexibility within these guidelines may be necessary to ensure that the system is as universal and as practical as possible.

Every effort has been made to utilize existing international standards: please see the reference section for a complete list of sources.

Chapter 2
Historical Perspective

2.1 Cinema Sound

5.1-channel audio was first developed for cinema applications. Cinema sound has an advantage over other consumer playback experiences: it is mixed in an environment that is extremely similar to that in which it is enjoyed. Film dubbing stages largely use the same speaker/crossover/amp systems as commercial theatres and both follow the same guidelines for room equalization. All aspects of the sound, such as the recording levels on the film soundtrack, program equalization, and the overall monitor levels during playback have been standardized and calibrated so that what the mixers create on the dubbing stage matches what is heard in the cinema.

This is not the case with 5.1-channel music. Every 5.1-channel music engineer/producer has their favorite monitors to mix on, just as every mastering engineer has their own approach and monitoring system. Because most consumers don’t have systems remotely similar to what the music was mixed and mastered on, the chances of successfully translating the studio experience to the home are currently less then optimal.

Many consumers listen to new 5.1-channel music mixes on their home theater systems, originally bought to watch movies. Also, because tens of thousands of movie titles have audio referenced a particular way, it makes sense to adopt some of the production practices for reference of film calibration, so that the consumer experience does not vary wildly between watching a film and listening to 5.1-channel music on the same system.

These issues present particular challenges to the 5.1-channel music mixer. This manual recommends production practices in an effort to approach a level of standardization that will help ensure that the consumer hears what the 5.1-channel music producer intended.

2.2 The LFE Channel

There are two distinct purposes for a subwoofer in a 5.1-channel music system. One entails the reproduction of the low-frequency effects (LFE) channel information and is discussed here in light of its cinematic origin. The other entails the reproduction of the bass content from other channels (bass management) and is discussed in Section 4.4.

Even though the continuous evolution of power amplifier and loudspeaker technologies have made it possible to reproduce a higher quality of sound in the cinema, it is still not easy to deliver or reproduce deep or accurate bass with a uniform response in large rooms filled with people. The best soundtracks of the late 1970s (70 mm magnetic analog) had reached their maximum recording capability, so it was impossible to increase the bass content without causing overload. Moreover, often playback systems are inefficient at reproducing low frequency information. Even today, the main screen speakers used in cinemas typically do not reproduce below 30 Hz, so if the soundtrack carries additional low frequency information to the amplifiers, it is not necessarily reproduced.

To illustrate how adding low-frequency content can tax a system, consider the relative sensitivity of the human ear to different frequencies at different monitoring levels. As seen in Figure 2-1, at a reference level of 80 dB, it takes an additional 10 dB of 65 Hz to equal the perceived loudness of 1 kHz. Adding this kind of level to the main channels often causes overload and decreases the available dynamic range.

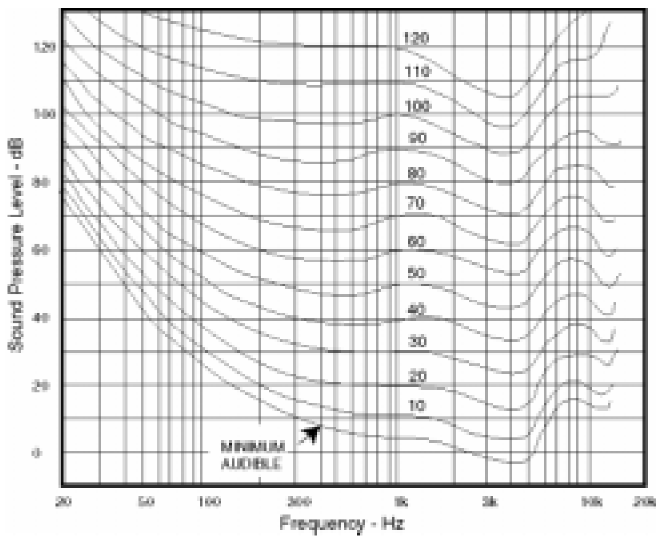


Figure 2-1 Fletcher-Munson Equal Loudness Curve

Subwoofers were installed to increase low-frequency playback capabilities and increase the overall dynamic range of cinema systems. A separate channel, LFE, was added to the soundtrack to provide an additional bass signal to the subwoofers. The LFE channel handles bass created specifically for special subwoofer effects.

Current 5.1-channel music delivery formats such as DVD-Audio allow each channel in a 5.1-channel mix to carry bass content. So why is there an LFE channel in a consumer audio delivery format? Quite simply, it allows movie soundtracks to be transcribed directly without alteration to the home video format. However, this scenario does not dictate the use of the LFE channel for multichannel music. It suggests that the LFE channel may not be the only or the best way to provide loud, deep bass, which becomes more apparent when one mixes multichannel audio using a properly configured and calibrated studio monitor system.

The use of a separate loudspeaker for reproducing the lowest part of the frequency spectrum in 5.1-channel music applications does have several advantages, including:

- Freedom to locate the bass source optimally in relation to room mode pressure distributions
- Reductions in the size (and sometimes cost) of the main loudspeakers
- Reduction in distortion because the main loudspeaker driver displacements can be reduced.

In short, if you really want the widest dynamic range and frequency response but feel that putting additional bass content in the five main channels will cause overload or decrease the overall dynamic range of those channels, the LFE channel may be just the tool to accomplish the goal.

2.3 Current Multichannel Sound Systems

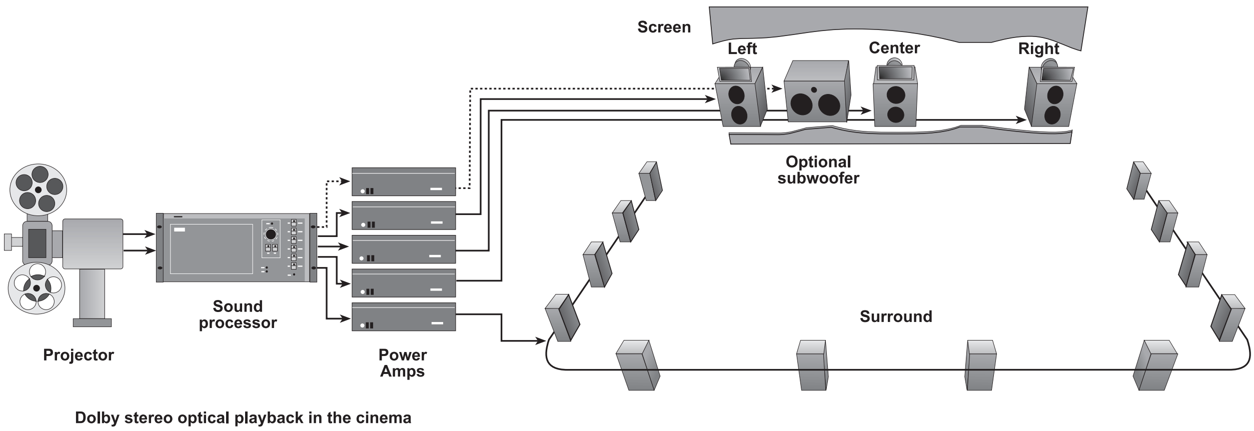


Figure 2-2 Dolby® Stereo (Motion Picture Matrix Four-Channel [4:2:4]) Setup for Theatres

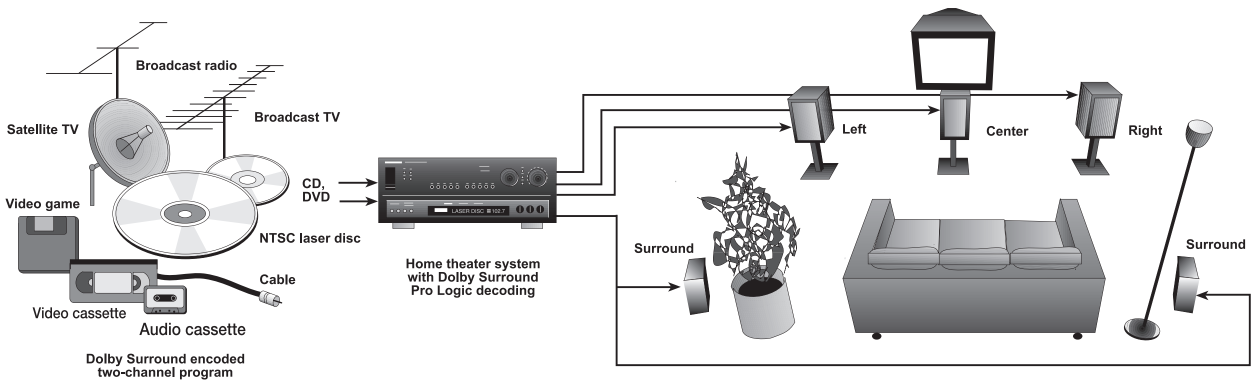


Figure 2-3 Dolby Surround (Matrix Four-Channel [Pro Logic®]) Setup for the Consumer

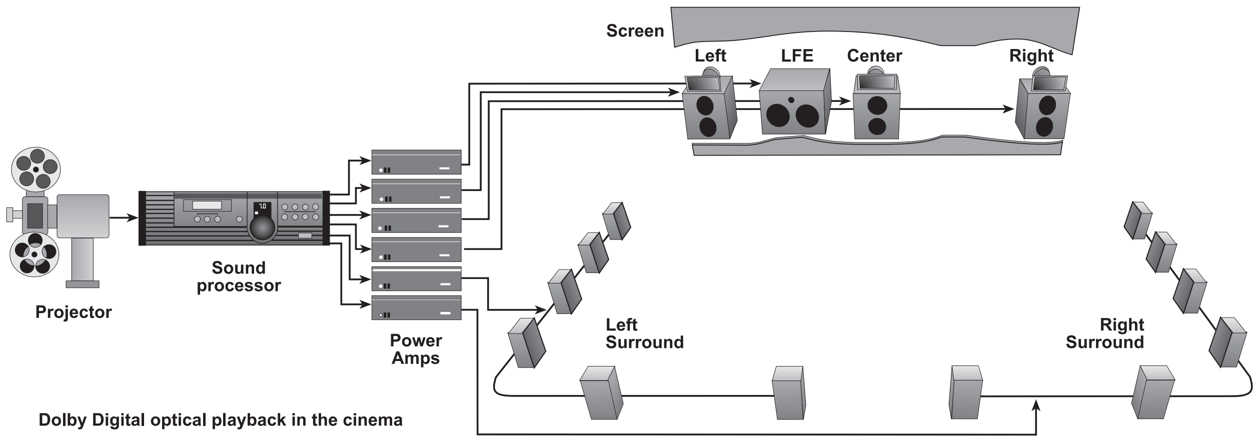


Figure 2-4 5.1-Channel Setup for Theatres

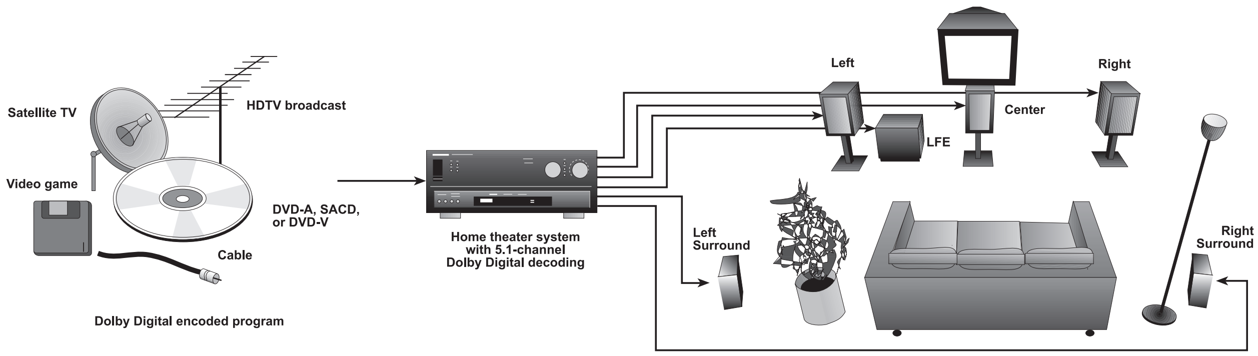


Figure 2-5 5.1-Channel Setup for the Consumer

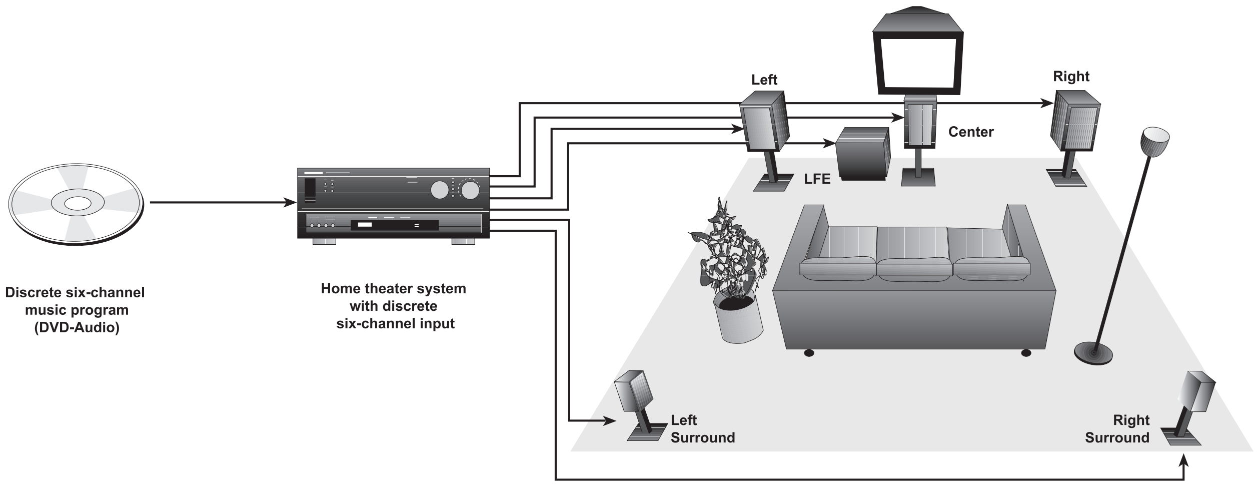


Figure 2-6 Alternate 5.1-Channel Setup for Consumer Music Playback

Chapter 3
The 5.1-Channel Music Mixing Environment

3.1 Room Design

Significant differences exist between stereo and multichannel production environments. Multiple speakers firing in different directions affect such basic factors as the optimum room size and geometry, equipment needs, construction methods, wiring, HVAC, lighting, power, and ergonomics. The addition and placement of equipment necessary for multichannel production often affects room acoustics as well. Whether designing a new facility or planning to retrofit an existing one, consulting a professional acoustician and architect familiar with building critical audio monitoring environments is always recommended.

The primary aim of a reference listening room is to facilitate interchangeable judgments between locations. Please note that these guidelines relate primarily to small- and medium-sized rooms.

3.1.1 Dimensions

The parameters given in Table 3-1 are meant as general guidelines and cannot completely describe the optimal sounding room; however, they are based on existing international standards for reference listening conditions for small- to medium-sized control rooms and provide a good starting point.

While many room shapes may work, the ideal room is symmetrical along the line between the center speaker and the reference listening position. An environment with no parallel walls (including the floor and ceiling) helps prevent the buildup of low-frequency standing waves.

A minimum height of 3 meters (9 feet) is desirable.

Table 3-1 Room Dimensions

Parameter	Units/Conditions	Value
Room Floor Area		>30 m^2 (320 ft^2)
Room Volume		<300 m^3 (10,500 ft^3)
Room Proportions	L = Length (larger dimension, irrespective of orientation) W = Width (shorter dimension, irrespective of orientation) H = Height	1.1W/H ≤ L/H ≤ 4.5W/H –4 with L/H <3 and W/H <3 No ratios of L, W, and H within ±5% of an integer value

For stereo listening, the speakers are traditionally set up on the short dimension of the room firing into the long dimension. Many new production rooms designed for 5.1-channel applications now place the front speakers on the long dimension, firing into the short dimension, which usually provides the most efficient and symmetrical use of the listening space. Multiuse production rooms that have been set up with alternate speaker orientations do not need to be reoriented. If there is an opportunity to redesign a room, reorientation should be considered.

3.1.2 Acoustics

Early Reflections

Any early reflections (within 15 ms) should be at least 10 dB below the level of the direct sound for all frequencies in the range 1 kHz to 8 kHz [6].

Reverberation Field

Reverberation time is frequency-dependent. The nominal value, T_m, is the average of the measured reverberation times in the 1/3-octave bands from 200 Hz to 4 kHz and should lie in the range: $0.2 < T_m < 0.4$ s. T_m should increase with the size of the room; the formula in Table 3-2 is a guide.

Table 3-2 Reverberation Values

Parameter		Units/Conditions	Value
Reflected Sound	**Early Reflections**	0–15 ms (in region 1–8 kHz)	< –10 dB relative to direct sound
	Reverberation Time	T_m[s] = nominal value in region of 200 Hz to 4 kHz V = listening room volume V_0 = reference room volume (100 m^3 [1075 ft^2])	$\approx 0.25(V/V_0)^{1/3}$

The reverberation time *T*, measured in 1/3-octave bands over the frequency range from 63 Hz to 8 kHz, should conform to the tolerance mask shown.

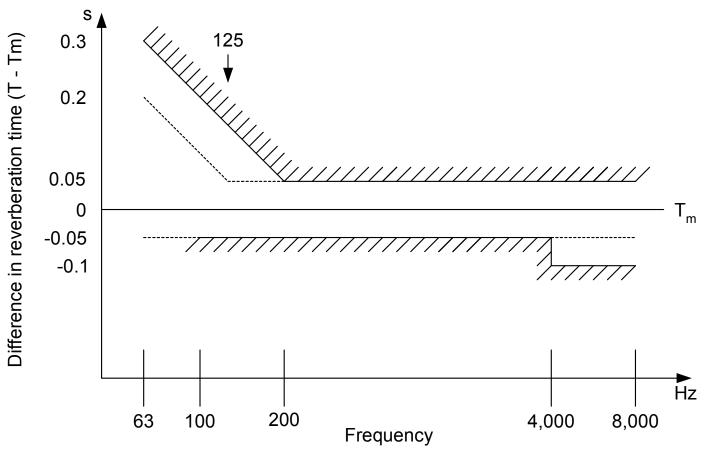


Figure 3-1 Tolerance Limits for Room Reverberation Time
(dotted lines = tighter tolerances proposed by the AES, see [1])

Reflective and Absorbent Surfaces

Large flat reflective surfaces should be avoided in the mixing environment. Placement of doors, control room windows, and equipment should be considered with speaker placement and aiming in mind. A combination of diffuse reflectors and absorptive materials should be used to achieve a smooth RT decay time within the specified range shown in Figure 3-1.

Again, it is recognized that these values may not be achievable in some installations, but is recommended that the room be measured using a real-time analyzer and that architectural solutions (wall treatments, bass traps, room reorientation, and so on) be utilized first to achieve the recommended values. A mixture of diffuse reflective and absorbent surfaces, applied evenly to the whole room, aids in creating an acceptable reference listening condition [12].

Only after considerable effort has been made using architectural solutions to smooth the room response should equalizers be introduced into the monitor chain. See Section 4.2 for more information on room equalization.

Background Noise

The listening area should ideally achieve an NC rating of 10 or below with the equipment off, measured at the reference position. A studio with equipment such as video projectors, video monitors, and other ancillary equipment powered on should achieve a rating of ≤ NC 15.

Any background noise should not be perceptibly impulsive, cyclical, or tonal in nature.

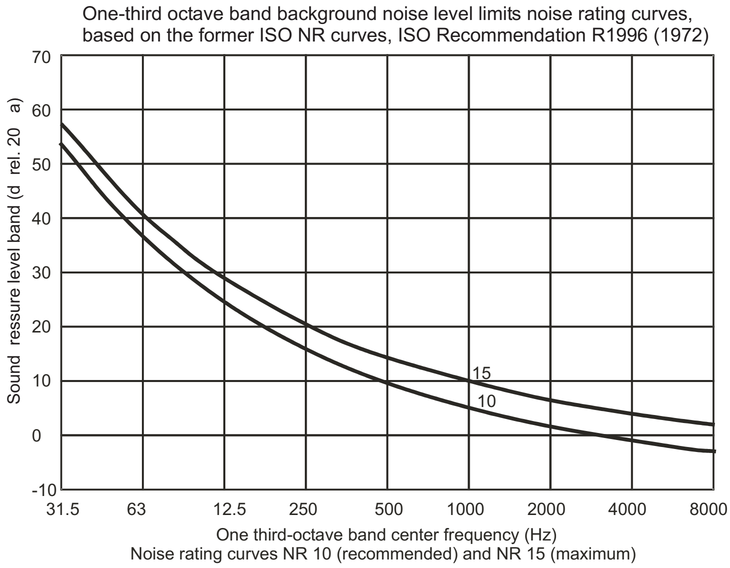


Figure 3-2 Noise Rating Curves

NR 10 or NR 15 may be hard to realize in a practical manner in some installations, in which case, every effort should be made to identify the loudest noise sources and correct as appropriate. The most common noise sources and possible remedies include:

- **HVAC systems**: Increase the surface area of the supply air vent. Separate or float all mechanical connections between high velocity or rumbling motors and ducts and the listening room.
- **Equipment**: Contain computers and other equipment with loud fan noise in noise attenuating, ventilated cabinets.
- **Doors and windows**: Make sure all the doors and windows are aligned properly and form a seal when closed. Adding a second window or door, with air space between it and the original, can reduce unwanted noise considerably.

Other sources of problem noise may need to be addressed. Every effort should be made to approach the recommended values shown in Figure 3-2.

3.1.3 Console Placement

In most cases, the console should be placed equidistant from the listening studio sidewalls. Lateral placement is discussed in Section 3.3.

3.2 Monitoring

3.2.1 Reference Monitors

Note: All five loudspeakers (L, R, C, Ls, Rs) should be identical.

One of the main differences between 5.1-channel setups for music and those for home theater playback is the type of speakers used for the surround positions. The goal of surround reproduction in the cinema (accomplished using multiple speaker arrays) is to provide surround playback to large audiences.

Surround effects are often very diffuse, ambient soundscapes. Dipole speakers are sometimes used in the home environment (rarely used in movie theatres) at the surround locations to help create the wide wash of sound created using an array of speakers in the cinema.

However, in 5.1-channel music production, as well as in films, the surrounds are sometimes used for distinct placement of featured performers. Accurate reproduction in this case requires the use of direct-firing speakers that match the overall characteristics of the front speakers. Use of matched direct radiator or monopole speakers in 5.1-channel music production is recommended for achieving the greatest control of level, timbre, and image location. Due to their dependence on null spot positioning, reflective front and rear listening room walls, and preference of a diffuse surround field, dipole speaker monitoring is not ideal for critical 5.1-channel music production.

It is recommended that all five speakers (L, C, R, Ls, Rs) be identical in all of the parameters listed in Table 3-3.

Table 3-3 Reference Loudspeaker Specifications

Parameter	Units/Conditions	Value
Amplitude/frequency response	20 Hz to 20 kHz* on axis (0°)	4 dB
	±10°	Deviation to 0°, 3 dB
	Horizontal ±30°	Deviation to 0°, 4 dB
Difference between speakers	In the range >250 Hz to 2 kHz	.5 dB
4 Directivity Index	250 Hz to 16 kHz	8 dB ±2 dB
Nonlinear distortion attenuation (SPL = 96 dB)	<100 Hz	–30 dB (=3%)
	>100 Hz	–40 dB (=1%)
5 Transient fidelity Decay time t_s, for reduction to a level of 1/*e*	t_s[s]	<5/*f* [Hz] (preferably 2.5/*f*)
6 Time delay	δt	≤10 μs
7 System dynamic range Maximum operating level (per IEC 60268, § 17.2, referred to 1 m distance)	$L_{\text{eff max}}$	>112 dB (at IEC 60268 program simulation noise or special condition)
Noise level	L_{noise}	≤10 dBA

* 20 kHz is a minimum value. Some delivery formats contain content up to 96 kHz. Choice of speakers may depend on the production format in use.

Source: Modified from AESTD1001.1.01-10

Not every speaker meeting these criteria may be useful for your purposes: conduct careful evaluations to determine the suitability of a particular speaker system.

For the pre-selection of loudspeakers, the frequency response curve over the range 20 Hz to 20 kHz, measured in one-third octave bands using pink noise on the main axis (directional angle = 0°), should preferably fall within a tolerance band of 4 dB. Frequency response curves measured at directional angles ±10° should not differ from the main axis frequency response by more than 3 dB, and at directional angles ±30° (in the horizontal plane only) by more than 4 dB.

The frequency response of different loudspeakers should be matched. The differences should preferably not exceed the value of 1.0 dB in the frequency range of at least 250 Hz to 2 kHz. [11]

The amplifier/speaker system should be capable of reproducing 120 dB without significant distortion.

3.2.2 Subwoofers

The frequency response of the subwoofer should be flat, ±3 dB between 20 Hz and 120 Hz.

Lower crossover frequencies will result in more freedom in the choice of subwoofer placement but with less benefit in the reduction in size of the main loudspeakers, as discussed in Section 2.2.

For almost complete freedom in the choice of location in a typical-sized room, the bass management (see Section 4.4) crossover frequency (when used) should be as follows:

Table 3-4 Reference Subwoofer Specification

Parameter	Value
Crossover frequency	80 Hz
Out-of-band harmonic distortion levels	≤–50 dB (0.3%)
Filter order	Fourth

Because the LFE channel for cinema applications contains content up to 120 Hz, however, the subwoofer itself should be capable of reproducing up to 120 Hz.

Parametric equalization may be needed to flatten the response of the subwoofer in a particular environment. See Section 4.2.

3.2.3 Power Levels

As a rule, the power amp should be able to provide 3 dB more power than the loudspeaker peak rating. The loudspeaker peak rating should be 3 dB higher than the peak allowed by the medium being mixed to.

3.2.4 Monitor Source Selection

Often, it is necessary to A/B between six-channel sources. If the console does not offer this feature, obtain an outboard device with this capability.

3.2.5 Multichannel Ganged Fader

If a multichannel mix is to be done on a console with fewer then six main monitor outputs, an outboard device offering six ganged attenuators (a six-channel volume knob) should be obtained. All six channels should track within a .5 dB tolerance over their full range.

3.3 Reference Positions

The reference position of the mixer's head is typically:

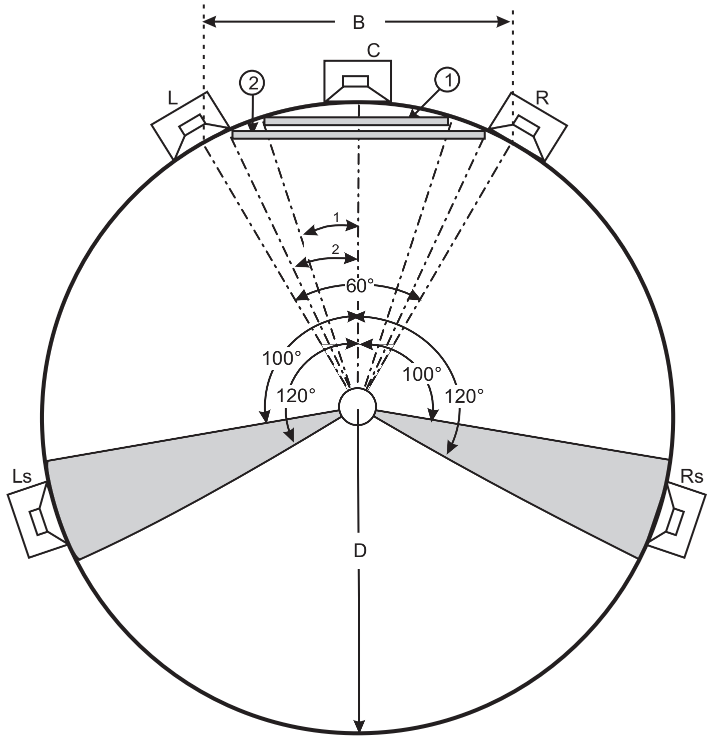
- At the center of the console
- Equidistant from the sidewalls
- Directly above the rear edge (arm rest) of the mixing console
- 1.2 meters (≈ 48 inches) off the floor

This imaginary point is the reference point for all speaker distances and aiming angles.

3.4 Speaker Placement

The speakers should be placed as shown in Figure 3-3. Final placement may depend on uncontrollable conditions, such as the physical dimensions or other constrictions of the facility and/or equipment. Note that these guidelines contain ranges of options, particularly with respect to the surround and subwoofer placement, which it is often necessary to test to obtain the most accurate and pleasing monitoring environment.

Regardless of particular system constrictions, please remember that the goal is a balanced multichannel monitoring system that properly images and facilitates interchangeable critical listening judgments between various locations.



H: height of screen
B: loudspeaker base width
D: listening distance
Screen 1 HDTV: reference distance = 3H (2 $_1$ = 33°)
Screen 2 = 3H (2 $_2$ = 48°)

Loudspeaker	Horizontal Angle from Center (degrees)	Height (meters/feet)	Inclination (degrees)
C	0	1.2/4	0
L, R	30	1.2/4	0
Ls, Rs	100–120	≥1.2/4	0–15 down

Figure 3-3 Reference Loudspeaker Placement

3.4.1 Front-Speaker Placement

The three front speakers (L, C, R) should be the same distance from the reference position. The center channel should be directly to the front of the reference position. With reference to the line formed between the center speaker and the reference position, L and R should be ±30 degrees horizontally.

Each front speaker should be the same height as the reference position (1.2 m/4 ft). All front speakers must be ±0 degrees vertically referenced to each other unless the center speaker needs to be positioned above or below a video monitor, forcing the acoustic centers of the three front speakers out of alignment. If this occurs, attempt to situate the speakers so the tweeters are in as close to a straight, horizontal line as possible. This may require either an inverted or lateral orientation of the center speaker, as well as rotating the center tweeter (when possible) to maintain the proper dispersion characteristic. In any case, keep the speakers equidistant from and directed to the reference position.

Table 3-5 Reference Position

Parameter	Units/Conditions	Value
Base Width	B [m]	2–3 m (6.5–10 ft)
Basis Angle	[°] referred to L/R	60°
Listening Distance	D [m]	= B

3.4.2 Surround Speaker Placement

The surround speakers should also be the same distance away from the reference position as the front speakers and located 110° ±10° from the reference line. They may be elevated to a position not to exceed 15° above the reference position, as long as they remain equidistant from and directed to the reference position.

3.4.3 Subwoofer Placement

Positioning the subwoofer(s) can often be an arduous task and the relative locations are not the same for all rooms. Expect a certain amount of experimentation, particularly when retrofitting an existing production room. The main requirement is that the location of the subwoofer not be audibly apparent.

One method for locating an optimal position is to place the subwoofer(s) near the listening position and play program material with significant low-frequency content. Then, listen at likely subwoofer locations around the room and choose the location that delivers the smoothest bass response. This location is apt to be the best choice for final subwoofer placement. Remember that the signal to the subwoofer is band-limited anywhere from 80 to 120 Hz and that as the crossover frequency rises, the ability to localize the loudspeaker position increases. Because the LFE channel may have content up to 120 Hz, it is recommended that the sub crossover be set at 120 or bypassed (as the bass-management filter will provide the necessary rolloff). However, keeping the crossover frequency for the bass-managed channels low (80 Hz) provides the greatest flexibility in positioning the subwoofer.

As recommended previously, control rooms are often set up in a symmetrical design, making it tempting to locate the sub in an equally symmetrical location (for instance, along the center line under the front speaker). However, a symmetrical placement in a symmetrical room often creates symmetrical standing waves and thus, an uneven room response. Placing the sub slightly asymmetrically may produce a more satisfactory result. Using a second sub can also help smooth out uneven room response problems.

Chapter 4
The 5.1-Channel Music Mixing Signal Path

4.1 Level Calibration

4.1.1 Dynamic High-Frequency Warning

In multichannel music, calibration and reference are vitally important to the outcome of the project. Over the years in broadcast audio production, AES, SMPTE, and ITU standards have been adhered to for consistent and predictable audio performance. Within the music industry, however, mixing levels and practices have been somewhat arbitrary. Consistency in playback of various media, from movies to music, is highly desirable to consumers.

Given today's higher resolution multichannel audio, mixers need to be aware of specific sonic scenarios related to monitoring, mixing, and mastering. High-resolution audio formats have the potential to contain greater dynamic range, span a wider frequency spectrum, and are capable of reproducing loud high-frequency content not realized or reproduced in earlier audio formats. Because such content above 20 kHz can affect the high-frequency drivers of some loudspeakers, and potentially, human hearing, caution with respect to high-frequency distortion is highly recommended when monitoring.

When working with high-resolution audio, make sure to analyze the extended frequency spectrum of the program to identify anomalies that may be harmful to equipment or the listener.

The frequency spectrum analysis, as well as an overall system alignment check, should be done at the start of each project and periodically throughout the duration of the project.

4.1.2 Alignment Signal Level

1 kHz Sine Wave Alignment Level

Considerations:

- The goal of this manual is to produce repeatable reference listening experiences in different listening environments.
- AES, EBU, ITU and SMPTE standards differ regarding alignment levels.

Use –20 dBFS as the 1 kHz sine wave alignment signal level, per current multichannel DVD production standards.

The RMS level of all test signals (see Section 5.6) will be at **Alignment Signal Level**, that is, –20 dB with respect to dB full-scale (FS) digital level in digital devices [3]

A 1 kHz sine wave at this level typically produces +4 dBu (= 1.23 Vrms relative to 0 VU [8]) from professional consoles.

In the case of digital devices, the alignment level must be a 1 kHz sine wave 20 dB below the maximum possible coding level of the particular digital system, irrespective of the total number of bits available. The alignment levels for 16-bit audio systems are shown in Table 4-1.

Table 4-1 Digital Codes for 1 kHz Sine Wave Alignment Levels

Number of Bits	Audio Alignment Level	
	Negative Peaks	Positive Peaks
16	F333	0CCD

The values in Table 4-1 produce the indications shown in Figure 4-1 for various types of program meters.

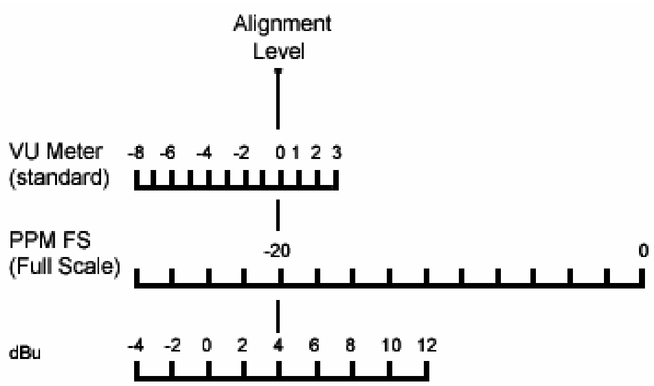


Figure 4-1 1 kHz Sine Wave Alignment Level Metering

Pink Noise Alignment Level

Because of the random nature of pink noise, peak program meter (PPM) readings can vary. The pink noise alignment signal level should be set to 0 on a VU meter after confirming the 1 kHz alignment settings. Again, this value usually produces +4 dBu (1.23 Vrms) on professional consoles.

4.1.3 Loudspeaker Alignment Level

Bass-Managed System Speaker Alignment

Because the subwoofer handles the lower frequencies for every channel in a bass-managed system, care must be taken to ensure a flat frequency response for the combined sub/satellite unit.

1. Feed wideband pink noise to the center channel at alignment level.
2. Adjust the level of the center speaker amp (point A in Figure 4-2) to 85 dBC in its operating range using an RTA.
3. Slowly raise the level of the subwoofer amp (point A in Figure 4-2) to achieve a flat frequency response on the RTA for the center/sub combination.

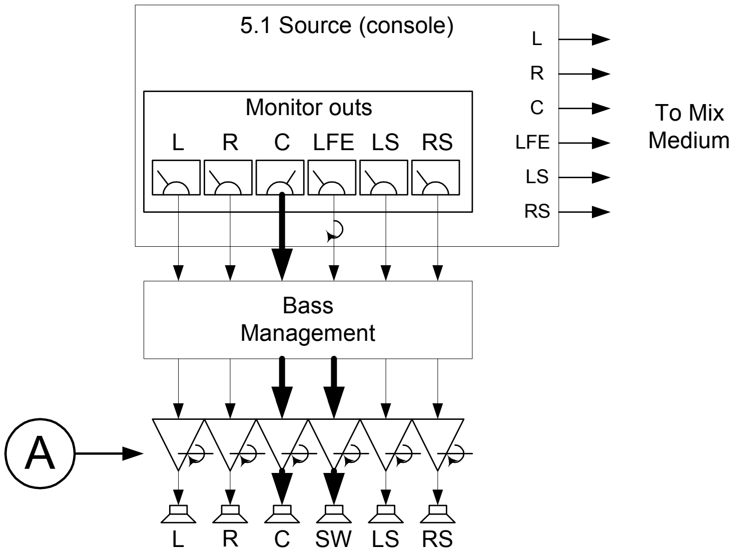


Figure 4-2 Bass-Managed Loudspeaker Alignment

After obtaining a flat frequency response from the bass-managed center/sub combination, the subwoofer and center amp settings (point A) should not be changed. Proceed with the remaining four channels (L, R, Ls, Rs) individually, adjusting their respective amps to a satellite/sub level of 85 dBC and a flat response.

LFE Alignment

The LFE channel should be calibrated in the following way for both bass-managed and non-bass-managed systems. If the system is bass-managed, however, align the LFE channel after adjusting the subwoofer level as described above.

1. Feed wideband pink noise to the LFE channel at alignment level.

2. Adjust the level of the console monitor feed (point B in Figure 4-3) to obtain an RTA reading of 10 dB in-band gain (average gain in the region 25 to 120 Hz) in relation to the same bandwidth measured for the center channel. See Figure 4-4.

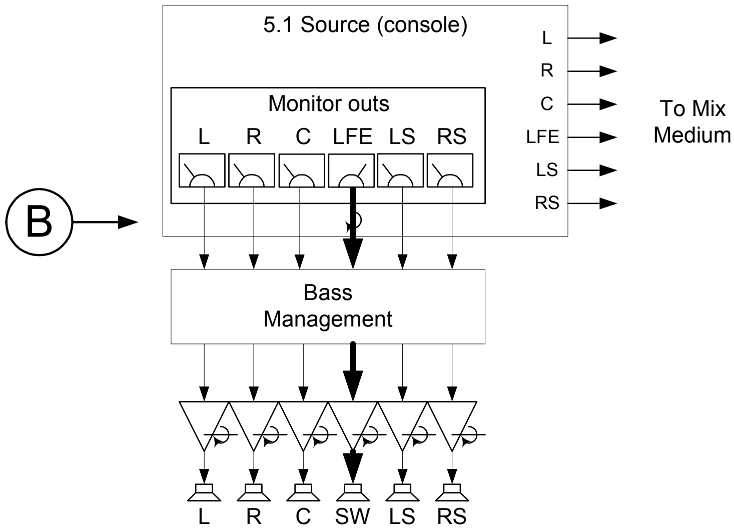


Figure 4-3 LFE Level Alignment

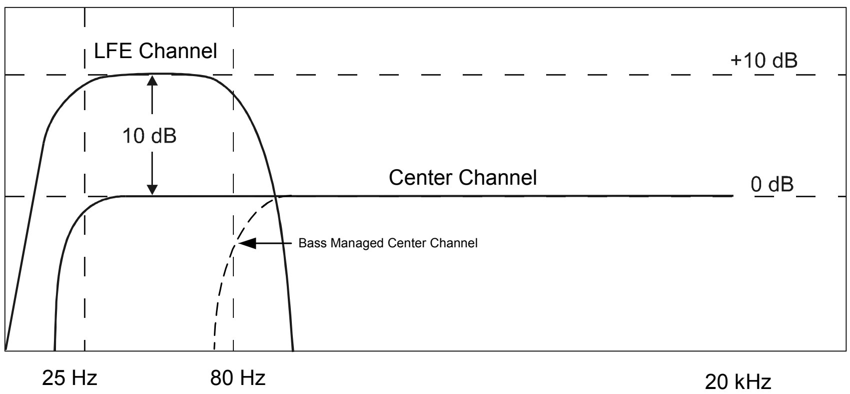


Figure 4-4 LFE RTA Display

Using this method to align the LFE channel ensures the accurate reproduction of the LFE channel commonly used for movie sound effects, and such.

If an RTA is not available, the LFE channel level can be set using the more commonly available SPL meter. Generally, when using wideband pink noise, the SPL reading (C-weighted, slow) should be approximately 4 to 5 dB above the main channels (that is, if the Center channel reads 85 dBC, then the LFE channel should read 89–90 dBC). Note that the SPL meter is taking a wideband measurement (not band-limited to 25–120 Hz) and therefore returns a lower value than the same measurement on a band-limited RTA.

If an RTA is available for the alignment procedure, record the SPL meter readings for that room for future alignments when an RTA is not available.

Alternatively, both the main speakers and LFE channel alignments can be done using specially prepared highpass and lowpass frequency-limited pink noise in which the highpass frequency-limited pink noise is used to adjust the main speaker levels and the lowpass frequency-limited pink noise is used to adjust the subwoofer level. In this case, the SPL will read the same 85 dBC for the LFE as it does for the main channel.

As discussed in Section 2.2, bass content can easily overload a recording system. This is the historical reason for the creation of the LFE channel and the practice of recording to tape 10 dB below reference level and boosting the playback 10 dB to achieve a flat response. In this way, plenty of bass can be included in the recording without causing tape saturation or distortion.

Because lower frequencies require more energy than higher frequencies to be perceived as the same level of loudness, recording energy-sapping deep bass on a separate track at a lower level to allow for a wider dynamic range is still good practice. For these reasons, plus the added bonus of legacy cinematic recording playback compatibility (both in the studio and the home), the +10 dB practice is still recommended.

4.1.4 Reference Listening Level

While Section 4.1.3 describes the correct loudspeaker alignment procedure, many listeners find the resultant level too loud for prolonged listening periods.

Several international listening standards dictate a reference listening level, L_{ref}, which is slightly lower than the loudspeaker alignment level and is determined by:

$$L_{ref} = 85 - 10 \log n \pm 0.25$$

where n is the number of reproduction channels in the total setup. The LFE channel is optional for 5.1-channel music mixing; therefore, n equals 5, making the individual channel level 78 dBC.

Using this level keeps the combined five channels, operating at reference level, at 85 dBC, which is well within the safety guidelines for occupational noise exposure [15].

4.2 Room Equalization

The biggest obstacle to obtaining a smooth room response in small- to medium-sized rooms is, by definition, the limited dimension in height, length, and width available to address bass nodes. Achieving the tolerances for proper room response curves shown in Section 3.1.2 can be challenging, especially in the low-frequency areas.

Use of an RTA is highly recommended to identify anomalies in the room response. Often, architectural or acoustical treatment of the room and/or adjustments to the loudspeaker and listening positions can solve many of the irregularities.

Equalization may be necessary when the linearity of the operational room response curve cannot be achieved by architectural means.

To avoid degrading the quality of reproduction, electrical equalization should be used carefully, and if possible, in the low-frequency range only.

4.3 Delay

The surround and front-channel signals should arrive at the listening position at the same time (coincident arrival); therefore,

- If the front loudspeakers need to be placed on a straight line base (moving the center speaker closer to the reference position), compensating time delay must be introduced in the signal feed of the center loudspeaker.
- If the surround loudspeakers are placed closer to the reference position than the front speakers, compensating time delays must be introduced in the signal feed of the Ls and Rs loudspeakers.

When necessary, delay should be added according to the following rule:

For each meter (foot) short of the reference distance, add 2.94 ms (.9 ms) of delay.

4.4 Bass Management

Note: Always check your mix using a bass-managed system.

In cinematic sound, explosions, earthquakes, and other high-energy, low-frequency special effects dictate the need for a dedicated subwoofer fed by the sixth, “.1” or LFE signal channel (see Section 2.2). Multichannel music may or may not make use of the LFE channel. Some music engineers feel that there is no need for the LFE channel at all. Because many of the popular consumer 5.1-channel speaker packages direct all of the bass from the main channels to the sub, however, combining it with any LFE information that may be present, it is important to at least check the effect bass management has on a mix while still in the studio. For example, in one known case, the combined bass from the main speakers was out of phase with the LFE channel, virtually eliminating the low end of the program when played back through a consumer, bass-managed system.

The goal of bass management is accurate bass reproduction in a given room—hearing all the bass frequencies in a smooth response in combination with the high-frequency drivers—and must be approached with caution and complete dedication to accurate calibration and monitoring.

As mentioned in Section 3.4.3, a low bass-management crossover frequency (80 Hz) gives more flexibility in subwoofer placement and helps keep its location audibly invisible.

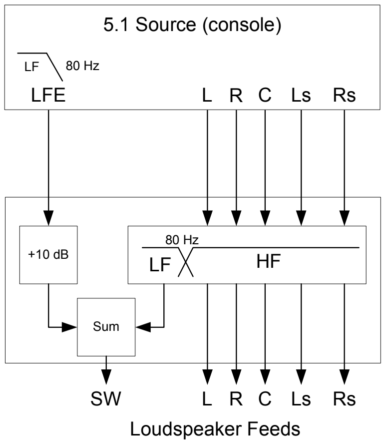


Figure 4-5 Bass Management

Bass-management allows the user to redirect low-frequency information from any of the five main speakers to the subwoofer. This is important since the five main speakers in home theater systems (satellite/sub speaker arrangements) are typically not designed to reproduce frequencies much below 80 Hz . Even though bass management is not required when monitoring in a studio with full-range speakers and a subwoofer, it is useful for checking how redirected low frequencies from any of the main channels may interact with the LFE-channel information. Remember that the consumer is likely to use some form of bass management, so proper bass management is necessary to emulate a consumer home theater system.

Historically, the LFE channel for cinema applications has a range that extends up to 120 Hz. Some less expensive consumer receivers offer only a fixed bass-management crossover frequency (often at 80 Hz) using one filter after the summation of the LFE and the main channel information (see Figure 4-6). This results in a situation where bass content between 120 and 80 Hz in the LFE channel is lost.

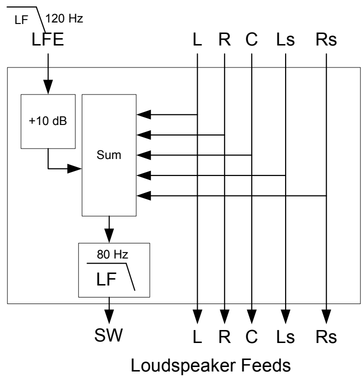


Figure 4-6 Worst-Case Consumer Bass Management

For this reason, it is recommended that the LFE channel be rolled off at 80 Hz during 5.1-channel music mixing.

The bass management feature on some consumer processors offers other combinations of redirection. For example, when the consumer does not have a subwoofer in the system, the LFE content can often be sent to the L and R main speakers. The 5.1-channel music producer may want to monitor the effects of these processes while still in the studio to ensure a desirable result.

Again, the goal in production and home playback is to hear all the bass accurately, regardless of which speaker reproduces it. When using full-range speakers with good bass response, there is theoretically no need for bass management, even when using a sub for monitoring the LFE channel. However, the track should still be listened to at some point with bass management engaged, to emulate what the consumer will hear at home.

4.5 Downmixing

The performance of a multichannel system under the conditions of two-channel playback should be tested using a reference downmix. Although the use of a fixed downmix may seem restricting, it covers the worst-case real-world scenarios that may be encountered.

In addition, time and budget may limit the ability to create a separate stereo mix, leaving the downmixed 5.1-channel as the only stereo option. In some circumstances, only a downmix is available (portable players, broadcasting, and so on); therefore, it is important to monitor the results while still in the studio. The equations for the reference downmix [10] are:

$$Lo = 1.00\,L + 0.71\,C + 0.71\,Ls$$
$$Ro = 1.00\,R + 0.71\,C + 0.71\,Rs$$

These downmix equations should not be confused with the more flexible systems offered by particular music formats, but as a quality check, the mix should be monitored as Lo, Ro.

Please note that since the equations are additive, it may be necessary to lower the overall level of the resultant downmix. Also, notice that the LFE channel may be disregarded in some circumstances, which should be kept in mind when deciding what content to include in the LFE channel during mixing.

4.6 Timecode

Timecode plays two important roles in preparing multichannel mixes. First, it is common to use some form of SMPTE or MIDI timecode for synchronizing recording machines and digital editors while recording and mixing material. It is important to know at the beginning of a project what the final timecode delivery format will be. Working in that format will save time later and prevent possible errors in frame rate and synchronization. This is especially true when working with video. With the introduction of high-definition video formats worldwide, even more varying frame rates and timecode modes (including drop-frame) now exist.

Second, if the material is going to accompany video at some point in the future, timecode is needed to enable a time stamp that is used to synchronize audio with the video, essential for proper audio/video synchronization.

Having the correct timecode that matches the picture is essential to proper audio and video synchronization. In addition, it is also very important that the timecode be stable and uninterrupted. Always use the timecode generated from a digital source such as a digital VTR. If unsure of the timecode source, generate clean timecode from a synchronizer or use a quality timecode regenerator.

In all cases, the associated audio and video equipment should all be clocked to a single master clock (see Section 5.3) to avoid timing drifts, mutes, and clicks and pops in the signal path and resulting master.

4.7 Consoles

When deciding on a console, it is wise to consider both current project demands and future 5.1-channel production needs. The requirements of 5.1-channel mixing consoles differ significantly from those of two-channel stereo. Fortunately, with the increasing flexibility of analog and digital consoles, there are now many options for surround mixing. Film-style four-bus (Left, Center, Right, and Surround) consoles have been in production for many years.

The fundamental requirement, however, of a 5.1-channel mixing console is a minimum of six discrete output buses (Left, Center, Right, Left Surround, Right

Surround, and LFE) per input/output channel. As with four-channel production, the six-bus console must also provide a means of panning audio. A console with film-style panning between the five main channels (L, C, R, Ls, and Rs) and routing to an LFE channel offers the greatest flexibility of sound placement in the surround field. While most manufacturers provide channel bus and pan features within the console, third-party developers have created add-on outboard devices with mixing controls to properly route multichannel audio for 5.1-channel mixes. In addition, whether onboard or third-party, console parameter automation is an asset when completing complex multichannel mixes.

Analog mixing consoles with six bus outputs and automation can offer a capable solution for multichannel mixing but require an analog-to-digital conversion stage, since 5.1-channel audio is delivered in the digital domain

In contrast, digital mixing consoles offer a direct path to today's multichannel delivery systems. Many digital consoles provide format conversion and internal signal processing. This all-in-one design delivers greater efficiency and flexibility during production. However, unwanted signal delays created by linking and chaining digital effects in digital audio consoles can cause timing errors between output channels. For instance, Left and Right audio channels may be configured to pass with relatively little processing along their signal paths. If the Center channel signal were to undergo processing such as digital EQ or dynamic compression, a measurable output signal delay could occur between the Center and Left/Right channels. Acknowledging this, digital console designers and manufacturers have created a variety of solutions to these timing problems.

4.8 Small Format Consoles

There has been an explosion in the marketplace of small professional consoles with enough buses to handle 5.1-channel mixing. While the implementation of multi-bus panning may be different, the functionality and setup of analog consoles and the newer consumer digital consoles are basically the same. Instead of panning in stereo, consoles with four or more auxiliary sends (in addition to the stereo bus prefader) can be used to place sounds anywhere in the 5.1-channel soundfield.

Currently, there are several excellent small digital consoles on the market. These consoles handle five-channel panning in different ways, with different features on the various models. Key features of some of these consoles include:

- Center mix level: allows adjustable panning through the center channel.
- Surround pattern editor: allows the path of the surround pan to be changed. Select the size and shape of a circle, arc, or line.
- Jog wheel speed manipulation: allows use of the wheel to change the speed of a pan.
- Multiple surround formats: 2+2, 3+1, 3+2+1.

- Master fader can be made into six-channel ganged fader.
- Divergence control on L/C/R and Front/Rear: these controls focus or spread the sound by controlling the bleed to adjacent channels during a pan.
- Surround Bus Assignment: in 5.1-channel mode.
- LFE level: controls the amount of signal going into the LFE channel.
- Surround Bus Isolation: Surround buses 1–8 are automatically solo-isolated so that they are not muted when any (surround-pan enabled) channel is in solo mode.
- Software control includes morph function that can interpolate between two points.
- On-screen editing of surround panning.
- Copy control of surround pans between channels.

Most console types put the surround mix through either the output bus path, or, in some cases, through auxiliary bus paths without separate multichannel monitoring paths. To have constant levels to the multitrack recorder as well as adjustable monitoring volume, it is necessary to bring the outputs of the multitrack back onto six grouped faders on the console and out a separate path to the amps and speakers. For the monitor outputs, use either auxiliary sends or a separate I/O card.

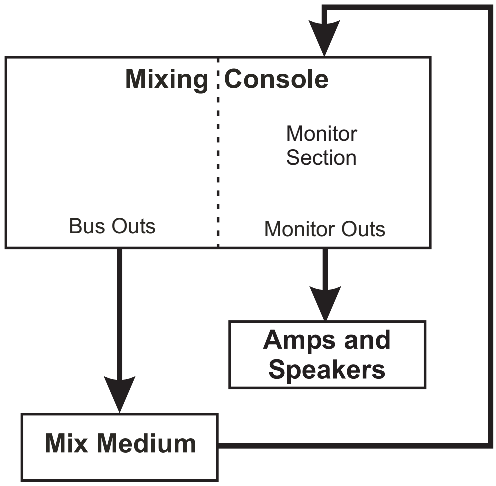


Figure 4-7 Console Interconnect Example

In Figure 4-7, arrows refer to six-channel buses, and should be connected using the following channel assignments, if possible: (1) Left, (2) Right, (3) Center, (4) LFE, (5) Left Surround, and (6) Right Surround. Choice of mix outputs (Bus, Aux, Monitor, and so on) depends on the console.

Currently, mixing-surface remotes that act as interfaces for software driven systems are getting more popular (Figure 4-8). Given the streamlining of this technique, there is less need to have additional digital I/O, sample rate converters, and so forth, while maintaining data integrity and audio fidelity in one host workstation.

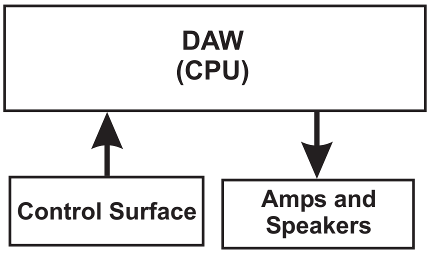


Figure 4-8 Digital Audio Workstation Interconnect Example

Note that some interface boxes for these types of digital audio workstations (DAWs), as well as some consoles, offer outputs at both +4 dBu and –10 dBV. Make sure that all of the monitor outputs are of the same level.

Chapter 5
Tips for Mixing 5.1-Channel Music

5.1 About 5.1-Channel Mixing

5.1-channel mixing offers an unparalleled level of creative flexibility. Use of the center and rear speakers, however, continues to be a topic of debate among artists, producers, engineers, and other serious listeners.

It is expected that genres of perspectives will develop over time and that mixes may someday be grouped into categories such as “in the band,” “in the audience,” or other real or imaginary listening perspectives.

Knowing the history of 5.1-channel audio will help in the future creation of a successful mix.

5.1.1 The Center Channel

If you want an image in the front center, use the Center channel. This is not to imply that hard panning or bussing to the Center channel only is recommended, but rather that it has been demonstrated [13] that a phantom center image delivers less clarity and a frequency response different from that delivered by a hard Center speaker. Using a combination of phantom and hard Center generally delivers the smoothest frontal image. However, when mixing to a combination of speakers, consider the phase relationships imposed by signal processors in use in some channels but not in others. If these signals are ever to be downmixed, unintended signal cancellations or comb filtering may result.

5.1.2 The LFE Channel

Since the reproduction of the LFE channel can sometimes be considered optional, essential low-frequency information should not be mixed exclusively to the LFE channel. In fact, in most downmixing situations, the LFE is completely disregarded. Conversely, in some rare cases, there may be a consumer playback system with small speakers, incapable of deep bass reproduction, yet lacking the bass management to direct the main channel bass to the subwoofer. In such cases, the content of the LFE channel is all the bass that will be heard.

When mixing to the LFE channel, it is important to band-limit the content at 100 Hz (see Section 4.4).

5.1.3 The Surround Channels

Even though it is recommended that five identical speakers be used for 5.1-channel music production, there are many installed consumer systems that use dipoles for the surround channels. Dipoles should be set up, calibrated, and used to listen to mixes, just to experience what consumers with these systems hear.

5.2 Metadata

Most new audio delivery formats allow the inclusion of information that describes, and in some cases, controls many aspects of the reproduction. The following are just some of the recommended fields that should be noted during various stages of production and, if possible, included in the metadata channel of the particular delivery format.

5.2.1 Informational Metadata

Informational metadata describes the content. Just a small sampling may include:

- The reference mixing level
- Room type/size
- Copyright information
- Artistic and production credits
- Identifying numbers (International Standard Recording Code [ISRC], etc.)
- Album title, track title
- Track number, disc number
- Label
- Genre
- Beats per minute

5.2.2 Control Metadata

Control metadata contains parameters that can be acted upon by a system capable of reading the information. Control metadata allows the production team to take control of and optimize how its audio program will be reproduced in different home listening configurations and environments. Examples of control metadata include:

- **Level normalization**: This is a measurement of the average loudness over time of a typical section of the program and can be specified as a dB level below full-scale digital. The film industry has extensive experience with this parameter. It is newer to the broadcast and music industries, but should be noted as a tool for matching program levels.

- **Downmixing**: As mentioned in Section 4.5, many formats enable automatic creation of downmixed versions of 5.1-channel material based on parameters set by the producer. Some systems offer a wide range of controls, including flags that limit downmixing capability.
- **Dynamic range**: For circumstances that require playback with varying amounts of limited dynamic range, this parameter offers the producer a chance to specify how the music should be controlled.

It is important to monitor the effects of various control metadata values while still in the studio to confirm the desired effect.

Obviously, each delivery format has its own metadata system and capabilities, but it is highly recommended that metadata of both informational and control natures is captured and monitored as early as possible—ideally, at the mixing stage—to help enable a more accurate and efficient production flow.

5.3 Master Clock

In all instances of multichannel digital audio production, every piece of gear should be synchronized to a master clock source. The absence of a master clock during sampling/recording, playback, editing, transferring, and so forth can lead to a number of disasters including phase issues, jitter and clocking errors, noises, pops, and ticks.

5.4 Sample Rate Conversion

Currently, music is released in a wide variety of formats including Red Book CD, DVD-Video, DVD-Audio, and low bandwidth codecs. These formats support a wide variety of sample rates and bit depths including 48/96/192 kHz and 44.1/88.2/176.4 kHz at 16-, 20-, or 24-bit resolution. Given that a number of projects require sample rate conversion, it is recommended that, in the interest of data integrity and fidelity, integer-based sample rate conversion be utilized when SRC is necessary. For a higher resolution project that is also slated for release on CD, the lowest common denominator is CD at 44.1 kHz at 16 bits. Ideally, the master recording exists at 176.4 kHz (24 bits) or 88.2 kHz (24 bits) allowing for precision integer down sampling to 44.1 kHz (for example, 88.2 kHz ÷ 2 = 44.1 kHz in conjunction with dithering 24-bit to 16-bit audio). Additionally, the same 176.4-kHz or 88.2-kHz master can be further down sampled to 44.1-kHz or 22.050-kHz and dithered as needed for low bandwidth codec delivery.

The converse, however, is discouraged. That is, because a number of 44.1-kHz and 48-kHz sampled digital masters currently exist, it is tempting to zero pad odd samples and integer upsample to be able to claim 88.2-kHz/176.4-kHz or 96-kHz/192-kHz output, respectively. Because the resultant audio stream does not contain additional audio data and is not of higher quality then the original, it is recommended that the music simply be released in its original sample rate (fs) and bit depth.

There are special circumstances where upsampling does produce a higher quality end result. For instance, upsampling a multitrack **before** digitally mixing has the advantage that any processing and/or effects added in the digital domain will be at the higher resolution. In this case, the end result **is** better then the original and is encouraged.

5.5 Documentation

Complete, clear, and accurate documentation should always accompany the source delivery master. This information is important not only when the master is in use but also as a reference, once it is archived. Dolby has created *Mix Data* and *Mastering Information* sheets to facilitate proper documentation or to use as a guide for creating similar documents. These sheets are available in Appendix A and at www.dolby.com under Technical Information. The *Mix Data* sheet provides concise information about the source media to all the engineers on a project. Typically, it includes information on sampling frequency, bit resolution, timecode, track assignment, titles, and program start and stop times. The *Mastering Information* sheet provides documentation relevant to the mastering engineer or authoring facility on source media, timing, and encoder settings, as well as general notes.

5.6 Test Signals

A 30-second, 1-kHz alignment signal at –20 dBFS, and a 30-second, 100 Hz alignment signal also at –20 dBFS should appear on all channels at the beginning of the source delivery master prior to program start. Two minutes of wideband pink noise, also at –20 dBFS (see Section 4.1.2) should be included on all channels to allow for the reference listening level alignment. The finished master should contain at least 30 seconds of digital black after the alignment signals and before each subsequent program. If appropriate, each title should begin with at least two seconds of encoded digital black.

For proper transport control in DVD-V and DVD-A, identification of intended index points/start times of the song for interpretation in the DVD authoring software (I and P frames) is required. See Appendix A.

Channel IDs and a polarity check mechanism would also be beneficial.

5.6.1 Mix Data Sheet

The *Mix Data* sheet provides production and mastering engineers with concise yet thorough media layout information. The information contained in the *Mix Data* sheet should be created during or after the multichannel master media is mixed, prior to mastering and encoding.

The *Mix Data* sheet should be used in conjunction with the *Mastering Information* sheet to provide necessary parameters prior to encoding for final delivery medium (that is, DVD-Video, DVD-Audio, DTV broadcast, etc.). All mix data information should be duplicated and placed on the master media as well. While recording and production media types may vary, accurate labeling of the media with *Mix Data* sheet information provides additional engineers with the proper origin knowledge.

5.6.2 Mastering Information Sheet

The *Mastering Information* sheet provides the mastering engineer or the digital authoring specialist technical information on media layout, timing information, and encoder specifics for each of the delivery mediums (that is, DVD-Video, DVD-Audio, DTV broadcast, etc.). If there is more than one delivery medium to address, each should receive a *Mastering Information* sheet. The information contained in the *Mastering Information* sheet is created both before the mastering process (RECOMMENDATION status indicated) and duplicated during the authoring/mastering/creation (FINAL MASTER status indicated) of the final delivery medium, (for example, Dolby® Digital or MLP Lossless™ bitstream for use on DVD-Audio) to confirm final selection of parameters.

Additional documentation, such as production notes, is invaluable in completing a project. Notes provide engineers with an explanation for key actions with relation to time, level, error, artistic consideration, and downmix-specific parameters. In addition to hard copy for each output medium, all documentation should be duplicated and affixed to the appropriate master delivery media (i.e., DLT tape, DVD-R, etc.).

5.7 Program Interchange

5.7.1 Channel-to-Track Allocation

Dolby encourages the adoption of channel-to-track allocation described in ITU-R BR.1384 Recommendation, *Parameters for International Exchange of Multi-channel Sound Recordings* and SMPTE 320M-1999, *Channel assignments and levels on multichannel audio media(Standard Assignment A).*

Track layouts depend on channel complement, although tracks 1, 2, and 3 are always channels Left (L), Right (R), and Center (C), respectively. Table 5-1 shows possible configurations. When the LFE channel is not used, track 4 may contain a mono Surround (MS) signal. Additionally, tracks 7 and 8 can be utilized for corresponding Lt/Rt or Lo/Ro stereo material. Alternative practices exist within various industries, so it is imperative to check the source and accompanying documentation.

Table 5-1 Channel/Track Allocation

	Track							
Format (channels)	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**
3/1 (four-track)	L	R	C	Ms				
3/1 (alt)	L	R	C		Ms (–3dB)	Ms (–3dB)		
3/2.1 (5.1)	L	R	C	LFE	Ls	Rs		
3/2.1 (5.1) + 2/0 [14]	L	R	C	LFE	Ls	Rs	Lt or Lo	Rt or Ro

Where:

L	=	Left channel	A	=	2/0 Left channel
R	=	Right channel	B	=	2/0 Right channel
C	=	Center channel	Lt	=	Matrixed Left
LFE	=	Low-Frequency channel	Rt	=	Matrixed Right
Ls	=	Left Surround channel	Lo	=	Left only
Rs	=	Right Surround channel	Ro	=	Right only
Ms	=	Mono Surround			

When recording additional multichannel programs onto a carrier with more than eight tracks (such as a 24-track recorder), groups of eight (that is, tracks 9–16 or tracks 17–24) should be preserved, as shown in Table 5.1.

Appendix A
Mix and Mastering Data Sheets

Mix Data

Date ____/____/________

Project ______________________ **Project #** ______________________

Client ______________________ **Producer** ______________________

Studio ______________________ **Engineer** ______________________

Sampling Frequency ☐ 44.1 kHz ☐ 48 kHz ☐ ×2 ☐ ×4

Bit Resolution ☐ 16-bit ☐ 20-bit ☐ 24-bit

Timecode Format ☐ 25 fps ☐ 29.97 fps ☐ 30 fps ☐ DF ☐ NDF

Media Format ☐ ______________

Surround Level SPL Calibration ☐ Equal to Front ☐ –3 dB to Front ______ dB SPL/channel

Tones ☐ 1 kHz @_______dBFS ☐ 100 Hz @____dBFS

Downmix data ☐ See Notes ☐ 100 Hz @____dBFS

Channel Configuration

CH1	CH2	CH3	CH4	CH5	CH6	CH7	CH8

Timecode	Program
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	

Notes:

Mastering Information

Date ____/____/________

Project ______________________ **Project #** ______________________

Client ______________________ **Producer** ______________________

Studio ______________________ **Engineer** ______________________

Mastering Status	☐ Recommendation	☐ Final Master			
Sampling Frequency (fs)	☐ 44.1 kHz	☐ 48 kHz	☐ ×2	☐ ×4	
Bit Resolution	☐ 16-bit	☐ 20-bit	☐ 24-bit		
Timecode Format	☐ 25 fps	☐ 29.97 fps	☐ 30 fps	☐ DF	☐NDF
Media Format	☐ ___________				

Tones	☐ 1 kHz @ ______dBFS		☐ 100 Hz @_______dBFS
Coding	☐ MLP Lossless™		☐ Dolby® Digital
Timecode Control	☐ No	☐ Yes	☐ Notes: In/Out/Duration
Downmix data	☐ See Notes		

Dolby Digital Encoding Information	Bitstream Information	Processing
Metadata Parameters LFE Filter ☐ ON ☐ OFF Dialogue Level: _____ Mix Level: _____ Data Rate: _____ Channel Mode: _______ Encoder Used: ________ Software Version: _____	Bitstream Mode: ___________ RF Overmod Protection: _____ ☐ Copyright Bit Center Downmix Level:_____ Surround Downmix Level: _____ ☐ Extended Bitstream ☐ Dolby Surround EX Preferred Downmix ________ Lt/Rt C Downmix Lvl: _______ Lt/Rt S Downmix Lvl: ________ Lo/Ro C Downmix Lvl: ________ Lo/Ro S Downmix Lvl: ________ ☐ HDCD converter	☐ Digital De-Emphasis ☐ DC Highpass Filter ☐ Bandwidth Lowpass ☐ LFE Lowpass Filter
Surround Channel Processing ☐ 90-Degree Phase Shift ☐ 3 dB Attenuation	**Line Mode/RF Mode Preferences** ☐ None ☐ Speech Film: ☐ Standard ☐ Light Music: ☐ Standard ☐ Light	

Channel Configuration

Mode	CH1	CH2	CH3	CH4	CH5	CH6	CH7	CH8

Timecode	Program
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	

Appendix B
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