5.1-Channel Production Guidelines

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Issue 1 S00/12957

Table of Contents

List of Figures

Chapter 1
Introduction

These guidelines provide starting points for producing 5.1-channel audio content by explaining terms, highlighting areas where there are alternative courses of action, and clarifying outcomes that may not be immediately apparent. The multichannel audio concept originated in the film world; therefore some cinematic terms are used.

1.1 Historical Perspective

5.1-channel audio was first developed for cinema applications. Unlike any other recording and playback format intended for a consumer audience, film sound is mixed in the same environment in which it is reproduced. All aspects have been standardized and calibrated so that what the mixers create on the dubbing stage is what is heard in the cinema. These aspects include the recording levels on the film soundtrack and the overall loudness during playback.

To improve the cinema sound years ago, level and calibration standards were established to ensure uniformity across various playback environments. Even though the continuous evolution of power amplifier and loudspeaker technologies made it possible to reproduce a higher quality of sound in the cinema, it was still not easy to deliver or reproduce strong bass. The best soundtracks of the day (70 mm magnetic) had reached their maximum recording capability, so it was impossible to increase the bass without causing overload. Even today, the main screen speakers used in cinemas do not reproduce below 30 Hz, so if the soundtrack carried more bass to the amplifiers, it would not necessarily be reproduced.

To increase low frequency playback capabilities, subwoofers were installed in the cinema. To direct bass signals to the subwoofers, a separate channel was added to the soundtrack. Known as the LFE channel (for Low Frequency Effects), it handles bass created specifically for subwoofer *boom* effects and may also carry low frequency information from the other channels in order to enrich the overall soundtrack.

A consumer delivery format like a CD is significantly different from a cinema system. CD loudness is not calibrated—the consumer decides where to set the volume control. Nor is the CD recording level calibrated—a music producer may add more bass to a recording by adjusting the overall levels thus ensuring that the additional bass will not cause overload.

In a similar manner, each channel in the Dolby Digital system can 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 and without alteration to the home video format. This does not mean the LFE channel should not be used. It suggests that the LFE channel may not be the only, or the best way to provide loud, deep bass. This becomes more apparent when one actually mixes multichannel audio using a properly configured and calibrated studio monitor system.

1.2 Dolby Digital and 5.1-Channel Audio

Dolby Digital (AC-3) is a perceptual audio coding system developed in 1992 to allow 35 mm theatrical film prints to carry multichannel digital audio in addition to the standard analog optical soundtrack. The system has since been adopted for use with laser disc, ATSC high definition (HDTV) and DVB/ATSC standard definition (SDTV) digital television, digital cable television, digital satellite broadcast, DVD-Video, DVD-Audio, DVD-ROM, and Internet audio distribution.

Dolby Digital divides the audio spectrum into narrow frequency bands using mathematical models derived from the characteristics of the ear, and analyzes each band to determine the audibility of those signals. To maximize data efficiency, the greatest number of bits represent the most audible signals; fewer bits represent less audible signals. In determining the audibility of signals, the system performs what is known as *masking*. Masking refers to the phenomenon that the ear cannot detect low-level sounds when there are higher level sounds at nearby frequencies. When this occurs, the high level sound masks the low level one, rendering it inaudible. Exploiting this phenomenon allows audio to be encoded more efficiently than in other audio coding systems. This makes Dolby Digital an excellent choice for systems where high audio quality is desired, but bandwidth or storage space is restricted. This is especially true for multichannel soundtracks since Dolby Digital's compact bitstream allows 5.1-channel audio to occupy less space than a single channel of Red Book PCM audio.

5.1-channel audio typically consists of five discrete, full range main channels (Left, Center, Right, Left Surround, and Right Surround) plus an optional band-limited Low Frequency Effects (LFE) channel for added bass (the .1). Dolby Digital bitstreams deliver full frequency bandwidth main channels, from 3 Hz to 20 kHz, and a limited frequency bandwidth LFE channel, from 3 Hz to 120 Hz. Current Dolby Digital encoders accept word lengths of 16, 18, or 20 bits at sampling rates of 32, 44.1, or 48 kHz. The Dolby Digital algorithm provides 24-bit resolution, and future versions may extend sampling rates to 96 kHz. All multichannel programs carried within a Dolby Digital bitstream, can be downmixed (see Section 2.2) for compatibility with Dolby Surround, stereo, or mono systems.

Chapter 2
Getting Started

2.1 Dolby Digital Encoder and Decoder

The discrete 5.1-channel mix will encode and decode in a Dolby Digital 5.1-channel system and remain discrete 5.1. When mixing and monitoring in 5.1, it is important to have monitors set up and calibrated correctly so the mix will play properly when decoded by the consumer.

Monitoring through an encoder and decoder is important in regards to downmixing and Dynamic Range Control (DRC). Using a Dolby Digital encoder and decoder in a monitor chain will allow quick examination of the variety of ways a consumer may hear a mix. Dolby Digital offers many features to maintain backward compatibility as well as to allow consumers the ability to customize their listening environments. For best results, features such as downmixing, DRC, and bass management need to be checked during content creation and delivery to see if they meet the intent of the content provider as well as the needs of the consumer. Encoding with a Dolby Laboratories Model DP569 Dolby Digital encoder and monitoring with a Model DP562 Professional Reference Decoder provides monitoring capabilities for these parameters in addition to being able to simulate almost any listening environment.

2.2 Downmixing

Downmixing has two frequently interrelated applications: format compatibility and channel redirection, as described below.

2.2.1 Format Compatibility

Dolby Surround-compatible, stereo, and mono mixes are often created when multichannel material is downmixed to fewer channels. It is important to check a number of aspects of each downmix to confirm that it translates as closely as possible to the original intent of the mix.

There are many consumers who will listen to Dolby Digital sources such as DVD or DTV without having a full 5.1-channel Dolby Digital playback system. These consumers will hear the two-channel analog or PCM outputs of their DVD players or DTV set-top boxes through existing stereo or Dolby Surround Pro Logic systems. All DVD-video players and DTV set-top boxes have the ability to create and deliver a Dolby Surround compatible or stereo downmix from the two-channel analog or PCM outputs. The DP562 Professional Reference Decoder can simulate what the consumer will hear while listening in these modes.

Example 1: Using a properly calibrated 5.1-channel monitoring system (incorporating appropriate bass management) set the DP562 to **Dolby Digital** and **Full**. In this configuration, a 5.1-channel bitstream will reproduce all channels as a consumer with a

Dolby Digital 5.1-channel system will hear it. Pressing **Pro Logic** on the DP562 downmixes the five main channels (discarding the LFE channel) to a Dolby Surround-compatible bitstream. The downmix is then Dolby Surround Pro Logic decoded resulting in Left, Center, Right, and mono Surround channels at the outputs. Monitoring in this mode simulates how a consumer will hear the 5.1-channel bitstream when downmixed and then reproduced through a Dolby Surround Pro Logic system.

Example 2: With **Pro Logic** still engaged, select **Stereo** instead of **Full** in the **Listening Mode** section. This mode provides for monitoring the way a consumer will hear the 5.1-channel bitstream when downmixed and then reproduced through a stereo system.

Example 3: In addition to Dolby Surround (Lt/Rt) compatible downmixes, Lo/Ro (Left only/Right only) downmixes can also be checked. Selecting **Stereo** mode without **Pro Logic** engaged will create an Lo/Ro downmix at the outputs.

2.2.2 Channel Redirection

The ability to redirect channel information provides a means to account for the design and number of speakers in the listening environment.

There will be consumers who may not have or can not use all 5.1 speakers with their Dolby Digital decoder. Dolby Digital consumer decoders have the ability to redirect or downmix decoded multichannel information such as a 5.1-channel soundtrack.

Example 1: Using a properly calibrated 5.1-channel monitoring system (incorporating appropriate bass management), set the DP562 to **Dolby Digital/Full**. In this configuration, a 5.1-channel bitstream will reproduce all channels the way a consumer with a Dolby Digital 5.1-channel system will hear them. Pressing any of the other **Listening Modes** causes the DP562 to redirect audio to the outputs of the selected speaker configuration.

Example 2: Select **3 Stereo** instead of **Full** in the **Listening Mode** section and the Surround channel information is redirected to the Left and Right speakers to simulate a monitoring system with no Surround speakers.

Example 3: Select **Phantom Center** and the Center channel information is appropriately attenuated and redirected to the Left and Right speakers to simulate a multichannel monitoring system with no Center speaker.

***Technical Note**: Summing multiple channels of audio, which occurs while downmixing, has the potential for overloading the decoder outputs. Please refer to the DP562 operation manual for more information.*

2.3 Dynamic Range Control (DRC)

Dolby Digital incorporates both dynamic range compression and protection against the decoder overload, which can result from downmixing. Some consumer products allow the user to choose full or reduced dynamic range when listening to a Dolby Digital multichannel soundtrack. When downmixing is in use, overload protection is applied automatically. The DP562 has the capability to monitor the dynamic range compression information encoded into the Dolby Digital bitstream. For further information on Dynamic Range Control (DRC), please consult the *Dolby Digital Professional Encoding Manual*.

Example: Selecting **Line** from the **Dynamic Compression** section implements both dynamic range compression and Dialog Normalization. This would simulate consumer listening conditions such as late night viewing of programs with a wide dynamic range or dynamic range compression required during downmixing. **RF** mode implements dynamic range compression, overload protection, and dialog normalization to simulate the case when the audio signal must follow an RF remodulation path for playback or for late night listening, when the least amount of dynamic range (lower signal peaks relative to dialogue level) is desired. **Custom** mode offers the same options for monitoring dynamic range compression as **Line** along with the ability to defeat Dialog Normalization (consumer products do not offer this mode). Caution must be used however. Depending upon how **Custom** mode is set up, it may not represent either the level balance or dynamic range heard in a consumer decoder. **None** is strictly a professional mode that defeats both dynamic range compression and Dialog Normalization. This mode (**None**) is used to hear the full level and dynamics in the program material being encoded. It is never allowed in a consumer product and may not represent either the level balance or dynamic range heard in a consumer decoder. Encoding judgments based on this compression mode should not be made.

2.4 Bass Management

Bass management allows the user to redirect low-frequency information from any of the five main speakers to the subwoofer; conversely, if there is no subwoofer, the LFE information can be redirected to the left and right speakers. This is important as the vast majority of consumer home theater speaker systems require some degree of bass management since typically none of the five main speakers is designed to reproduce frequencies below 80 Hz (i.e., satellite/sub speaker arrangements). The DP562 provides the same bass management functions as a consumer Dolby Digital decoder. Even when monitoring with full-range main speakers that require no bass management, this function 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 will most likely use some form of bass management. Accordingly, proper bass management is necessary to emulate a consumer home theater system. See Section 3.3 for more details.

2.5 Compatibility with Existing Dolby Surround Equipment

In the world of 5.1 digital multichannel audio such as Dolby Digital, it is important to remember that stereo delivery in the form of broadcast, VHS, and CD will continue to exist. However, with Dolby Surround, stereo audio formats have delivered multichannel audio since 1982. Dolby Surround is a passive matrix four-channel (Left, Center, Right, Surround: LCRS) to two-channel encoding system that is delivered via stereo media, Figure 2-1. The stereo-compatible Dolby Surround encoded soundtrack is referred to as Left total, Right total or Lt/Rt. In both the analog and digital world, Dolby Surround exists on media, e.g., VHS Hi-Fi, broadcast, CD, laser disc, etc. Dolby Digital has backward compatibility with Dolby Surround encoded material.

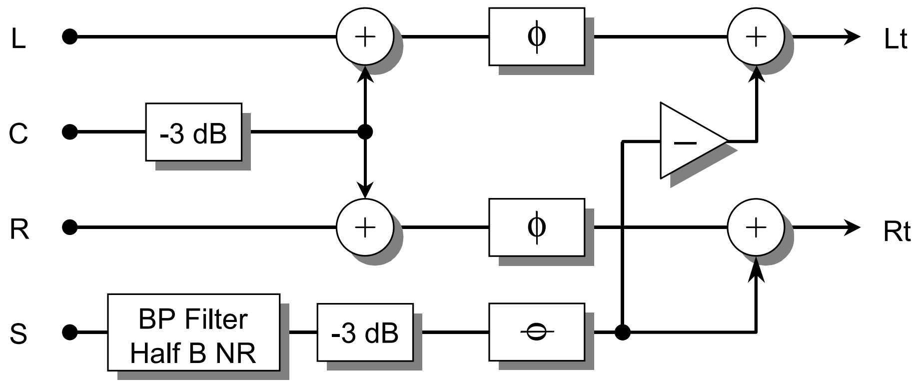

Figure 2-1 Dolby Surround Encoder

Every multichannel Dolby Digital decoder, such as an A/V receiver, contains a digital implementation of a Dolby Surround Pro Logic decoder. Dolby Digital decoders will allow stereo material encoded with Dolby Surround (Lt/Rt) from digital sources (e.g., laser disc, DVD, DBS, digital cable, etc.) to be decoded back to a four-channel output (LCRS).

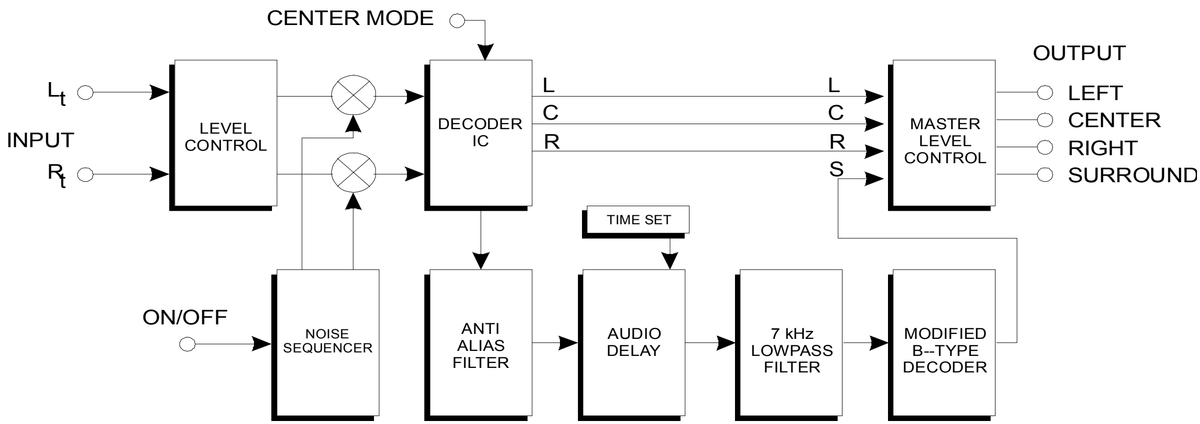

Figure 2-2 Dolby Surround Pro Logic Decoder

In the case of producing a Dolby Surround encoded mix (Lt/Rt) for a Dolby Digital encoded project, it is important to create the Dolby Surround mix through a Dolby Surround encoder (SEU4 or DP563) and a Dolby Surround decoder (SDU4) or Dolby Digital multichannel decoder (DP562).

For further information on the theory of operation of Dolby Surround and how to mix in Dolby Surround, please refer to the *Dolby Surround Mixing Manual* (part number 91536) available at www.dolby.com.

Chapter 3
Production Environment

The advent of 5.1-channel audio production poses unique challenges not only for artists and audio engineers, but also for the designers and builders of audio facilities.

3.1 Room Layout/Design

Significant, although not necessarily profound, differences exist between stereo and multichannel production environments. Basic factors to consider when creating any critical production environment include equipment needs, room size and geometry, construction methods, wiring, HVAC, lighting, AC power requirements, acoustics, and ergonomics. The addition and placement of equipment necessary for multichannel production will often affect room acoustics. Whether designing a new facility or planning to retrofit a studio, consulting a professional acoustician and architect familiar with building critical audio monitoring environments is always recommended.

3.1.1 Room Size and Shape

Depending on the application, room size and shape will vary considerably. In multichannel audio production for broadcast, limited space in a remote truck often makes speaker placement difficult. Similarly, in a production facility designed to accommodate additional listeners such as clients, producers, and staff, increased furniture and monitoring requirements may become factors.

3.1.2 Acoustics

The advantage of building a new facility is the opportunity to design specifically for multichannel production. When retrofitting a studio, it is important that additional equipment needed for multichannel production does not compromise the room acoustics. Simply relocating this equipment or modifying acoustic treatment can reduce or even eliminate undesirable acoustic anomalies.

3.1.3 Examples/Diagrams

A variety of room layouts for 5.1-channel audio production exist. All examples require a space that can allow for placement of five full-range (20 Hz–20 kHz) speakers around the engineering position with placement of an additional speaker capable of reproducing the LFE (Low-Frequency Effects) channel (3 Hz–120 Hz) in the designated space. A typical room layout for producing multichannel audio will have Left, Center, and Right speakers placed in front, while two or more surround speakers are needed behind the engineer,

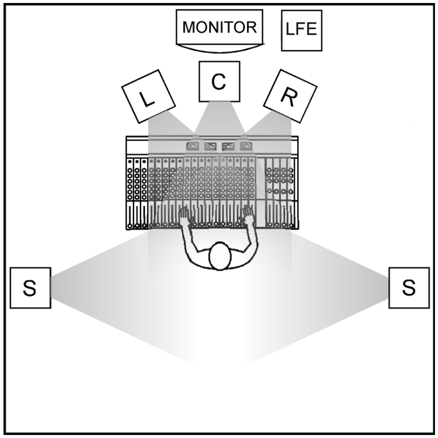

Figure 3-1 Typical 5.1-Channel Room Layout

Alternative room layouts for multichannel 5.1 production have been suggested by the International Telecommunication Union, as in Figure 3-4 ITU-R Recommended Listening Room.

For larger listening spaces with secondary listening areas, additional surround speakers are needed to accommodate the additional space, as show in Figure 3-2 Extended Room Layout. In this example it is important to optimize the engineer's listening position when compromise between two arrays of surround speakers is required for area coverage.

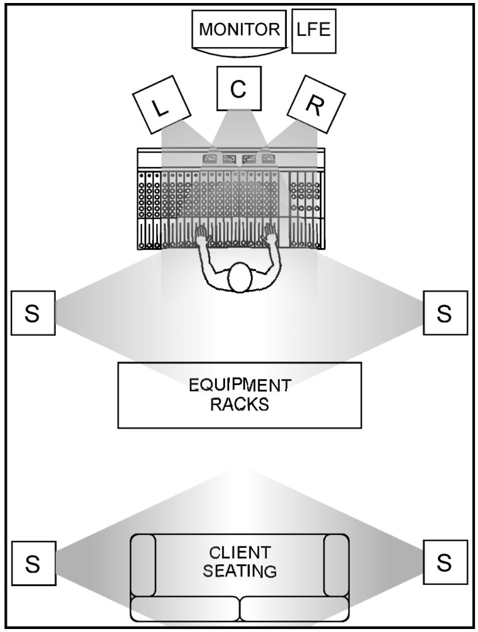

Figure 3-2 Extended Room Layout

For larger critical listening environments, such as dubbing stages, several surround speakers may be used to simulate the playback environment of a movie theatre.

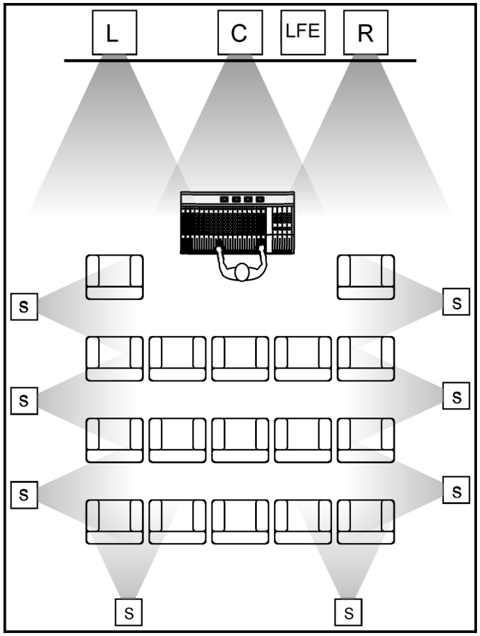

Figure 3-3 Large Listening Room Layout

All multichannel room design considerations should bear in mind requirements for producing Dolby Digital 5.1, as well as Dolby Surround. For more information on Dolby Surround and Dolby-suggested production room layouts, please review the *Dolby Surround Mixing Manual,* Part Number 91536, available at www.dolby.com.

3.2 Monitoring

Some aspects of multichannel studio monitor setup are understood and accepted. The best approach to others is still open to debate. The following guidelines offer commonly accepted practices for setting up multichannel audio monitoring systems.

3.2.1 Front Speakers

Multichannel sound systems add a center speaker to the Left/Right pair used in stereo systems. To promote good imaging, all three should be identical, just as conventional L and R stereo speakers must be matched. If all three cannot be the same model, the center speaker may be a smaller model from the same product line.

The front speakers should be equidistant from the listener, with their acoustic centers in the horizontal plane; that is, on-axis to the ear. The center speaker may need 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 horizontal straight line as possible. This may require either an inverted or lateral orientation of the center speaker, as well as rotating the tweeter (when possible) to maintain the proper dispersion characteristic.

If the center speaker is not equidistant with the L/R pair, signal delay may be used to obtain coincident arrivals. Please refer to Section 3.4, Level Calibration.

The front speakers must exhibit the same acoustic polarity. It is highly recommended that electronic signal polarity be maintained throughout the entire monitoring system.

3.2.2 Surround Speakers

Whenever possible, use the same speakers all around to achieve uniformity. If this is not feasible, the surround speakers may be smaller than the front speakers but should maintain the same character; i.e. they might be smaller speakers from the same manufacturer.

The front and surround speakers should be equidistant from the listener, with their acoustic centers in the horizontal plane that is on-axis to the ear. Refer to Figure 3-5.

The surround speakers should achieve coincident arrival with the front speakers either as a result of equal path lengths or through alignment with signal delays. Please refer to Section 3.4, Level Calibration.

The Surround speakers must exhibit the same acoustic polarity as the front speakers. It is highly recommended that electronic signal polarity be maintained throughout the entire monitoring system.

3.2.3 Subwoofer(s)

The LFE channel requires the use of at least one subwoofer in the monitor system. It is equally important to include one or more subwoofers and bass management when some or all of the speakers may not cover the deepest bass in soundtracks or music recordings. The bass from any channel that is not reproduced in the main speaker for that channel must be redirected to the subwoofer(s). There are now various products that handle bass management (crossover filters, bass mixing, and combining with the LFE channel in the proper mixing ratio) that can help achieve a proper monitor setup in the studio. It is essential to integrate the subwoofer(s) with the main speakers correctly to ensure a wide, smooth, and uniform frequency response from all five main channels. In addition, it is critical to have the LFE channel reproduced in the proper relation to the other channels.

Positioning the subwoofer(s) can often be an arduous task and the relative location(s) will not be the same for all rooms. A certain amount of experimentation should be expected particularly when retrofitting an existing production room. Initially, place the subwoofer(s) near the listening position. Play program material with significant low frequency content and listen at likely subwoofer locations in the room. Locations delivering the smoothest bass response are apt to be the best choice for final subwoofer placement.

3.2.4 Room Layout

The ITU-R[1] has specifications for a listening room layout designed for the critical evaluation of multichannel programs. These recommendations are a good starting point for a mixing room setup as well. Aside from signal alignment, a specific geometry is described. With the Center speaker directly in front, position the L/R speakers 30° from center (forming a 60° angle) and the Surround speakers 110° off center. Please refer to Figure 3-4.

[1] The ITU, International Telecommunication Union, headquartered in Geneva, Switzerland, is an international organization within which governments and the private sector coordinate global telecom networks and services. ITU-R refers to the Radiocommunication Sector.

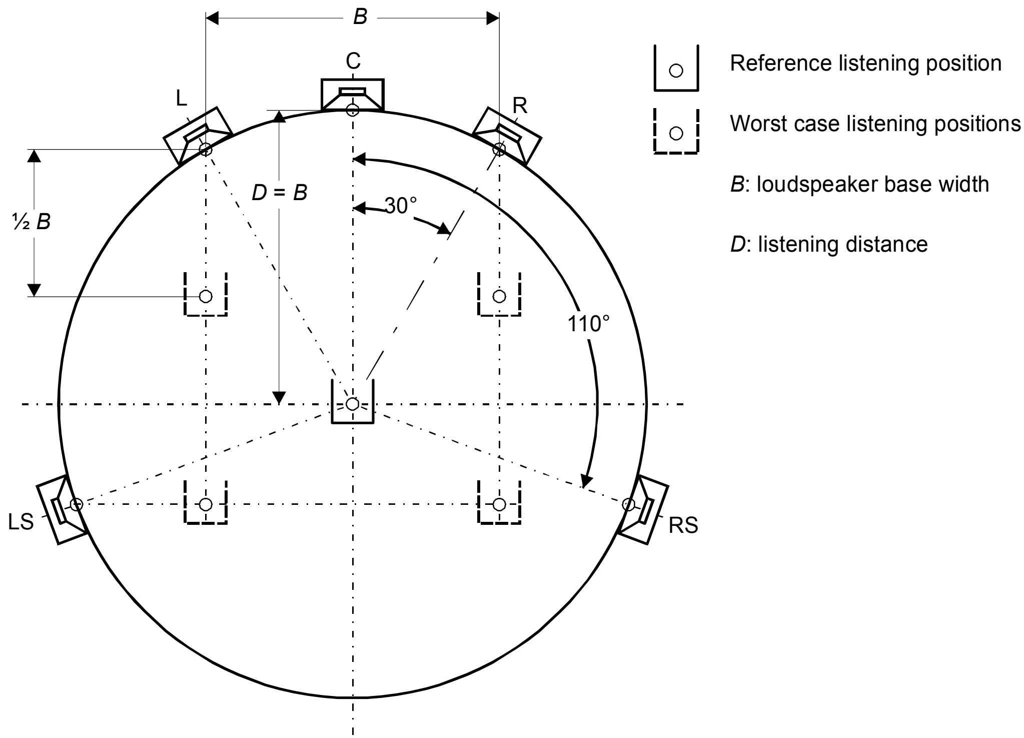

Figure 3-4 ITU-R Recommended Listening Room

3.3 Bass Management

Stereo requires the reproduction of signals from 20 Hz to 20 kHz. This is done with multi-way speaker systems, which utilize a combination of woofers and tweeters to achieve full range response. These speakers are connected via a crossover network to route the appropriate frequencies to the various speakers in the system. This may be a two-way, three-way, four-way, or even-five way system, but in each case, the goal is to reproduce 20 Hz to 20 kHz evenly.

The introduction of Dolby Surround home theater systems added three speakers to the stereo system, a full range center speaker and two limited-range surround speakers. The bandwidth of the surround channel is 100 Hz to 7 kHz. This results in the ability to use small bookshelf speakers for the surround channel.

Although the center channel is full bandwidth, most consumer applications will not allow a full-sized center speaker as would be typical for the left and right speakers. Because of this, a bass management system is included in the Dolby Surround Pro Logic consumer decoder to cross over the center channel low frequencies below 100 Hz and redirect them to the left and right speakers. The bass signal is now reproduced by speakers capable of handling the information without overloading the smaller, center speaker.

Dolby Digital consumer decoders also include a bass management system. Just as with the stereo and Dolby Surround systems, the goal is to be able to reproduce all frequencies within the system. In this case, there are more options, as the five main channels are full range and an extra LFE channel is added.

Possible combinations of speakers include five full range main speakers and a subwoofer for the LFE; five small speakers for the main channels and a subwoofer for both the LFE and all five main channels; and various combinations of the above examples.

As with consumer 5.1 applications, studios must be able to reproduce all reasonable frequencies from each full bandwidth channel. Crossovers, subwoofers, and main speakers should work together to give flat response for each of the five main channels. In addition, larger rooms may dictate the need for more than one subwoofer to achieve adequate bass response. Many manufacturers of near-field monitors make complementary subwoofers to complete the system.

When utilizing the LFE channel in a mixing situation, it is important to band-limit the information for this channel. In the Dolby Digital encoding process, the encoder will *brickwall* filter the LFE signal at 120 Hz. To properly hear the LFE content, a sixth or seventh order 120-Hz low-pass filter must be included in the monitor chain. It is advisable to include this filter in the console output before the monitor such that both the recorded information and the heard information are band-limited. Failure to include this filter will result in hearing substantial bass information above 120 Hz in the mix that will not be present in the Dolby Digital encoded version. 120 Hz is the proper crossover frequency for theatrical film applications.

For consumer applications such as DVD and Digital TV, the consumer decoders add a slight twist to the equation. Consumer decoders take the LFE signal and add any channels in need of bass management, as determined either by product design or user selection. The five main channels are then high-pass filtered at either a fixed frequency of 80 Hz or a selectable frequency of 80, 100, or 120 Hz. The summation of the LFE and any other channels is low-pass filtered at the same frequency. If the crossover frequency is fixed at 80 Hz, as is standard in lower priced decoders, information in the LFE channel between 80 Hz and 120 Hz will be reproduced at a lower level than it is recorded at. To replicate what the consumer will hear, a third order 80 Hz filter in the LFE audio signal path to the recorder is recommended.

While the Dolby Digital Encoder and Decoder together will handle bass management in decoding, it is not feasible to use it in this way when mixing, due to the delay through the encoding and decoding process. Because of this, it is necessary to have a separate crossover system in place to handle the bass management. Many manufacturers now offer such devices for this purpose. Again, for DVD and other consumer applications, a crossover frequency of 80 Hz is required.

Once the ability to reproduce all frequencies in each channel has been met, the room must be calibrated. For each of the five main channels, pink noise is adjusted for 85 dB C-weighted slow. Various EQ curves will come into play, depending on room size and application. If in doubt, additional information is available in the *Dolby Surround Mixing Manual*, *Technical Guidelines for Theater Applications* and other publications from Dolby Laboratories.

The LFE channel is calibrated such that each 1/3 octave band between 20 and 120 Hz is 10 dB higher than the equivalent 1/3 octave bands for any of the full-range speakers, assuming that the full-range speaker is ideally flat. This level is read from a real-time analyzer (RTA), rather than a Sound Pressure Level (SPL) meter. If an RTA is not available, an SPL meter may be used to approximate the level. When the level is correct , most meters will read around 90–91 dB SPL C-weighted slow for the LFE channel. The difference in level is because there is no energy being reproduced for the frequencies above 120 Hz (80 Hz for consumer applications).

A properly calibrated room will result in mixes that will sound correct when played back in a consumer environment. An improperly tuned room will result in mixes that will sound fine in the mixing facility, but will be incorrect in other situations. Using the guidelines above will result in a properly tuned room. The room alignment has little to do with the release format. A discrete six-track mix or a Dolby Digital encoded DVD will reproduce equally well when these guidelines are followed.

3.4 Level Calibration

It is important to have a properly calibrated audio monitoring system to ensure accurate encoding and decoding. Proper calibration requires the use of an SPL meter and an RTA to measure relative and specific playback levels (in decibels) of all six channels.

3.4.1 Playback Levels

Use an SPL meter and a Pink Noise Generator to set your system’s playback level to a particular reference level.

There are three options to adjust monitor system playback levels:

- Amplifier gain trim controls
- Mixer’s group outputs (one for each of the L, C, R, Ls, Rs, and Sw channels)
- Decoder output level trim controls

The best option is to use your amplifier gain controls to set proper playback levels. This option allows you to maintain optimum signal-to-noise performance from the decoder and console. Using either the group outputs from the console or the output level trim controls on the decoder may sacrifice signal-to-noise ratio.

Pink noise readings depend on the type of meter used to set level (and, strictly speaking, on the bandwidth of the pink noise signal). In film practice, pink noise level is read with a true VU meter, or a meter display with a VU characteristic. If reference level is specified as "0 VU," where "0 VU" corresponds to -20 dBFS for a digital recording medium, then pink noise should be set to “0 VU” on the console meter or meter display, and the SPL set accordingly.

Note: *Many, if not most, modern consoles have peak-reading meters or meter displays. Pink noise that reads at reference level using a true VU meter will read from 8–12 dB higher on a peak reading meter or meter display. If your console has switchable meter characteristics, be sure to select the "VU" mode when setting pink noise levels.*

For film work, pink noise at reference level should produce a sound pressure level (SPL) of 85 dBC for each of the front channels (left, center, and right). Each surround channel should produce a sound pressure level of 82 dBC (the lower surround level is specific to film-style mixing rooms).

For television work, pink noise at reference level is typically set to produce an SPL ranging from 79 dBC to 82 dBC for each of the main five channels. The lower reference level for television is due to the lower average listening levels used by the consumer (typically 70–75 dBC). Since the reference listening level used can dramatically effect the balance and intelligibility of the mix, it is important to consider both the level at which a program was mixed, as well as the typical listening level in the home for the same program.

For music mixing, each speaker channel should be set to the same SPL (the same as television mixing). There is no standard practice for reference levels for music mixing. Some engineers prefer to mix louder than others do, but if the levels between channels are correct, the overall level is not as important. However, as stated previously, consideration of the typical consumer listening level is always good practice.

When mixing for television or music in small mixing rooms (e.g., remote recording trucks), the surround channel is generally set 2 dB lower than the front channels (for example, 80 dB in front and 78 dB in back). This takes into account the short distance to the surround speakers.

Experience has shown that this setting makes the sound in the home environment very close to the sound heard by the mixer.

3.4.2 Sound Pressure Level Meter

To properly calibrate speaker levels, an SPL meter is necessary. A suitable and relatively inexpensive meter is readily available from Radio Shack ® (Tandy Electronics outside of North America). Since the relative level between channels is more important than absolute level, the accuracy of this meter is sufficient for channel balancing. For greater accuracy, more expensive meters may be used. It is recommended that an inexpensive meter be left in the control room for quick level checks. Note that this meter is not generally suitable for calibrating the LFE channel.

Please refer to the subwoofer calibration section.

3.4.3 Taking a First Measurement

1. Before you turn on the Pink Noise Generator, be sure that your playback system has been set to a moderate listening level. Adjust your amplifiers, self-powered speakers, or mixer only.

WARNING: BEWARE THAT IF THE PLAYBACK LEVEL IS VERY HIGH, YOU MAY RISK DAMAGING YOUR SPEAKERS OR POSSIBLY EVEN YOUR HEARING.

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2. An internal test noise generator will typically cycle pink noise automatically between the Left, Center, Right, Right Surround, and Left Surround channels—

remaining approximately two seconds at each output before moving on to the next. If you do not hear pink noise in any of the five main channels as the noise cycles through that channel, check your system connections and settings.

3. Ensure that you are sitting in your normal, proper reference listening or mixing position. Set the SPL meter to “C” weighting and “slow” response.
4. Facing the front speakers, hold the SPL meter at chest level, with the microphone facing up at an angle of approximately 45 degrees to the center speaker. Keep the meter at arm’s length to prevent measuring audio that may reflect from your body. You should be able to take SPL readings as you look down at the meter.
5. Keep the SPL meter in this position. Make sure that the meter is aimed at the center speaker as you take readings for the left and right speakers.
6. When taking the SPL readings for the left surround or right surround speakers, keep the meter at the same angle and position as you did for the front speakers. Turn your body 90 degrees from the center speaker towards the wall closest to the surround speaker you are measuring. This will minimize “shadowing” or obscuring the meter with your body.

3.4.4 Bass Redirection

Prior to calibration, decide whether the monitoring system requires bass redirection of low frequencies. Many decoders are capable of redirecting low-frequency information below a selected frequency (typically 80, 100, or 120 Hz) to the channels capable of reproducing them. This is similar to the normal/wide mode for the Center channel in a consumer decoder with Dolby Surround Pro Logic.

For example, if the monitoring system consists of five satellite speakers and a subwoofer, redirect the low frequencies from the five main channels to the subwoofer output. If using small Center and Surround speakers, direct the low frequencies from those channels to the Left, Right, or subwoofer outputs. If no subwoofer is available, redirect the LFE channel to the Left and Right channel outputs. Bass redirection can be accomplished in either a Dolby Digital decoder or by using an external dedicated bass management box.

3.4.5 Individual Level Calibration

There are typically two types of pink noise available for speaker/channel calibration: pink and filtered. For setting relative levels, filtered noise is recommended, as any room anomalies will not be factored into the level calibration process. For spectral balance of each channel, pink noise is used with an RTA to adjust the system for proper frequency response.

3.4.6 Subwoofer Calibration

Ideally, the test noise used for subwoofer calibration should be band-limited pink noise, low-pass filtered at 120 Hz. To properly calibrate the subwoofer, an RTA is required. If an RTA is not available, you can approximate the settings with an SPL meter.

When using an RTA, proper calibration requires setting the LFE channel signal sent to the subwoofer, within its typical bandwidth of 25–120 Hz, 10 dB higher than the main channels. Refer to Figure 3-5.

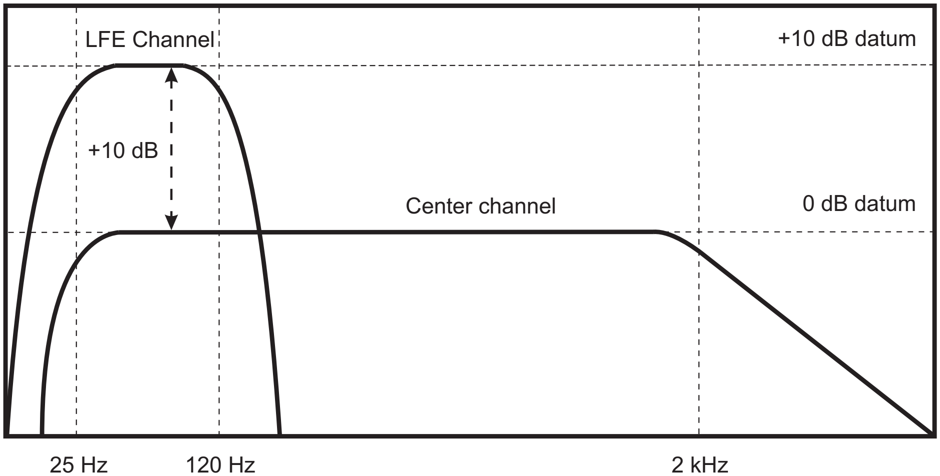

Figure 3-5 Real-Time Analyzer (RTA) Display

If an RTA is not available, setting the LFE channel higher (e.g., ~ 90 to 91 dBc for the subwoofer channel when the Center channel measures 85 dBc), can give an approximate level with an SPL meter. This level varies with the quality of the meter being used.

For future reference, if calibrating the subwoofer with an RTA, measure the level with an SPL meter and note the meter reading for the proper calibration. Use this measured value for quick checks of the system calibration in the future.

3.5 Signal Delay

In addition to setting the proper monitoring levels, it is important that the sound from each speaker arrives at the listening position at the correct time.

In an ideal room setup, the five main speakers would be equidistant from the mixing position. In this case, the listener would hear common sounds emanating from two or more speakers at the same time. If the speakers are not and cannot be equidistant, signal delay will be required to achieve the intended result. The speaker position furthest from the listener determines which of the remaining speakers will require signal delay. Relative delay times are derived from the difference in distance between the furthest speaker and each individual speaker. The required delay time is approximately 1 ms per foot or 3 ms per meter. With proper compensation, the entire system should exhibit coincident signal arrival at the primary listening position.

To achieve this, the delay times of the Center and Surround channels may need to be adjusted. Channel delay is determined by calculating the distances from each speaker to the listener, as seen in Figure 3-6.

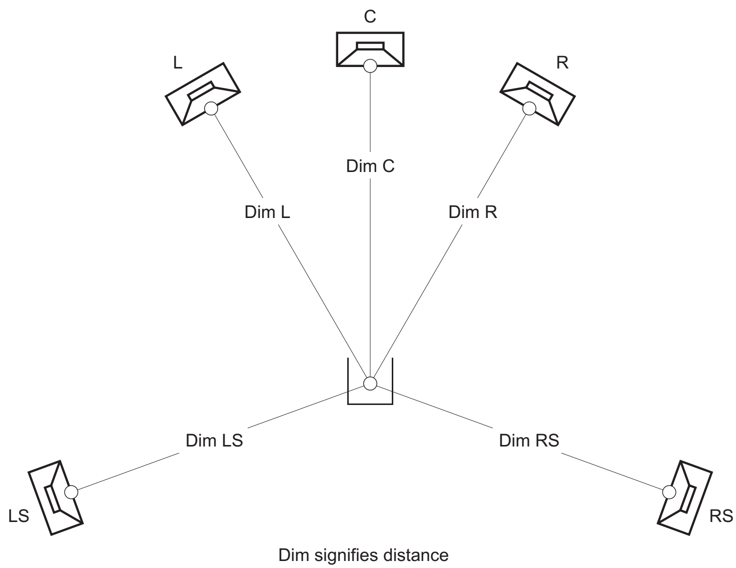

Figure 3-6 Speaker-to-Listener Distances

Since Dolby Digital is a discrete system, there is no crosstalk between channels. Although related, the delay times required for Dolby Digital are different from those required for Dolby Surround.

In decoders, Surround delays may be set up separately for Dolby Surround and Dolby Digital, with the Center channel delay time being the same for both modes.

3.5.1 Surround Delay for Dolby Digital

Determining channel delays for Dolby Digital is similar in concept to those for Dolby Surround Pro Logic, with important additional considerations. Since Dolby Digital delivers discrete signals for each channel, there is no leakage or crosstalk between channels. There is, therefore, no need to delay the Surround channels to take advantage of the Haas effect (precedence effect, which states that if two similar sounds arrive at our ears at slightly different times, the brain tends to focus on the sound arriving first and ignore the second). Dolby Surround takes advantage of this effect to reduce the perceived crosstalk between the front and rear channels. A signal arrival difference of 10–20 ms is adequate to make the Haas effect work. The Surround and front channel signals should arrive at the listening position at the same time (coincident arrival). Consequently, the Dolby Digital mode uses approximately 15 ms less delay than the Dolby Surround Pro Logic mode for the same speaker/seating arrangement.

To calculate the Surround delay for Dolby Digital, measure the distance from the listening position to each of the three speakers:

- Left speaker (L) or Right speaker (R)
- Center speaker (C)
- Nearest Surround speaker (LS or RS)

All of these measurements must be made in feet. If measuring in meters, multiply the metric measurements by three to get the approximate equivalent in feet. Once these two measurements have been made, calculate the required delay settings for the room. Use the following formula to calculate the delays.

(Distance from L) – (Distance from S) = Surround delay for Dolby Digital

L = Left or Right speaker

S = Left Surround or Right Surround speaker

In the example below, the listening position is 15 feet from the Left (or Right) speaker and 10 feet from the Surround speakers, so the delay setting for the Surround channels is five milliseconds for Dolby Digital. This is the correct delay in milliseconds to program into the decoder.

15 (number of feet from L) – 10 (number of feet from S) = 5 (Surround delay for Dolby Digital, in milliseconds)

3.5.2 Center Channel Delay

The concept of coincident arrival can also be applied to the adjustment of the Center channel in Dolby Digital. No delay is required if the Center speaker can be placed at the same distance from the listener as the Left and Right speakers (in an arc). Thus, distance C = distance L or R. If the Center speaker is placed closer to or farther from the listening position than the Left and Right speakers, delay can be added to the Center channel signal to bring it into acoustic alignment with the Left and Right speakers. This will electronically move the speaker to its proper position in the room.

For example, if distance C is two feet less than distance L or R, Center delay is set to 2 ms. If distance C is two feet more than distance L or R, Center delay is set to -2 ms.

To make the Center delay negative, the decoder actually sets C delay to zero and adds delay to the Left, Right, Left Surround, and Right Surround outputs of up to 3 ms. This ensures coincident arrival of the Surround and front channel signals.

3.5.3 Compatibility with Dolby Surround Monitoring

When monitoring Dolby Surround signals, a decoder adds additional delay (typically 15 ms) to the Surround channel output. When using a Dolby SDU4 or Dolby DP562 for Dolby Surround decoding, the minimum delay setting for these units will allow them to be used with 5.1-channel monitoring systems.

Chapter 4
Equipment

Much of the equipment used for stereo and Dolby Surround production can also be used for 5.1-channel production. There are, however, some needs specific to 5.1-channel production.

4.1 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, Right, Center, 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 (Left, Center, Right, Left Surround, and Right Surround) and routing to an LFE channel will offer 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. However, since 5.1-channel audio is delivered in the digital domain, using an analog mixing console requires an additional analog-to-digital conversion stage.

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. Be aware though, that 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 using digital EQ, dynamic compression, etc., a measurable output signal delay would 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.

4.1.1 Small Format Console

There has been an explosion of small professional consoles with enough buses to handle 5.1 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.

The semi-professional eight-bus analog consoles on the market today do not include surround panners or built-in master surround panners. Panning between buses on these consoles usually entails using auxiliary sends. There are also excellent retrofitting solutions available such as PicMix from Otari, (www.otari.com) and the OPS-1 Surround Sound Panner from Omnisound (www.omnisound.com). These panning systems include all the features of a large format surround sound console using external rack-mounted systems and joystick panners.

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

Yamaha (Current models are 02R and 03D)

1. Center mix level–allows adjustable panning through the center channel.

2. Surround pattern editor–allows the path of the surround pan to be changed. Select the size and shape of a circle, arc, or line.
3. Jog wheel speed manipulation–allows use of the wheel to change the speed of a pan.
4. Multiple surround formats: 2+2, 3+1, 3+2+1.
5. Master fader can be made into six-channel ganged fader.
6. Software editing available.

Tascam (Current models TMD8000 and TMD4000)

1. Divergence control on L/R and F/S–use these controls to focus or spread the sound by controlling the bleed to adjacent channels during a pan.
2. Surround Bus Assignment–in 5.1 mode, set this to Type 4 to comply with current L, R, C, LFE, LS, RS assignments.
3. LFE level–controls the amount of signal going into the LFE channel. This also may be set to **Off**.

Mackie (Current Model D8B)

1. 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.
2. Software control includes morph function that can interpolate between an **a** and **b** point (currently a straight line).
3. On-screen editing of surround panning with Mackie OS2 Ramsa (Panasonic) (Current Model WR-DA7).

4. Surround pattern editor–allows the path of the surround pan to be changed. Select the size and shape of a circle, arc, or line.
5. Copy control of surround pans between channels.
6. Jog wheel speed manipulation–allows use of the wheel to change the speed of a pan.
7. Software editing available.

Each of these console types puts the surround mix through either the output bus path, or, in some cases, through auxiliary bus paths without separate multichannel monitoring paths. This means that 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 use auxiliary sends, or a separate I/O card.

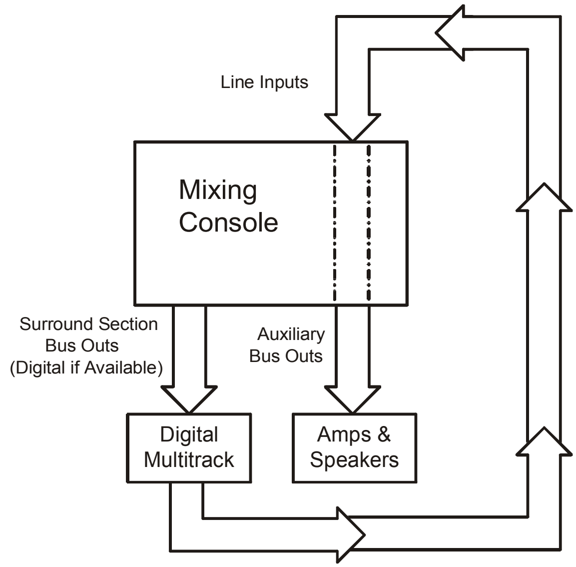

Figure 4-1 Interconnect Example

In Figure 4-1, 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, etc.) will depend on the console.

4.1.2 Digital Audio Workstations

Digital audio workstations (DAWs) can be employed to great effect for non-linear editing and mixing of multichannel audio. Though the vast majority of systems currently available are designed with stereo in mind, it is possible to use these workstations for multichannel applications with the help of some simple workarounds. The minimum requirement for mixing 5.1-channel audio is a stereo bus with four or more auxiliary sends that can be used for discrete placement of audio elements.

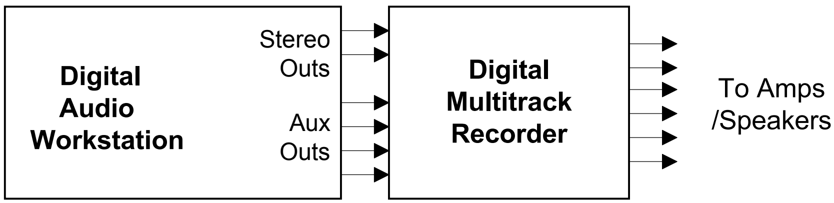

Figure 4-2 Simple DAW Interconnect

Currently, there are numerous DAWs on the market from a wide range of manufacturers, including Digidesign, Solid State Logic, Sonic Solutions, SaDie, Digital Audio Labs, and Fairlight.

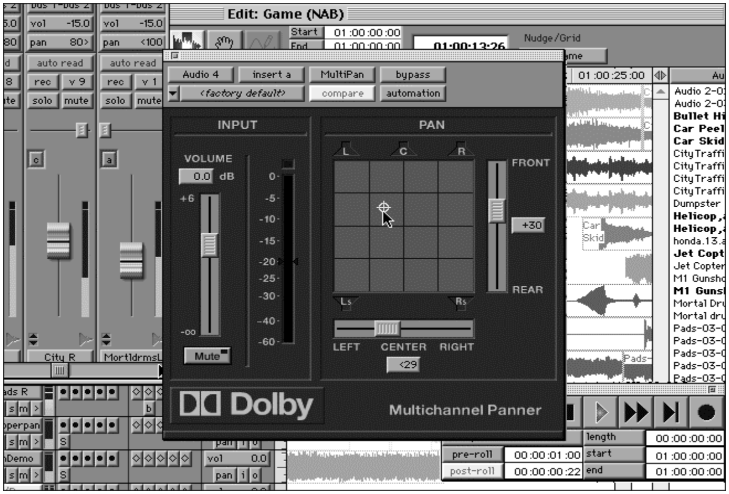

Figure 4-3 Typical DAW with Multichannel Panner

Each of these workstations positions the surround mix through either the output bus path or the auxiliary buses. These outputs are typically routed to a multichannel digital mastering deck (e.g., ADAT or DA-88, etc.). However, in order to have consistent levels feeding the multitrack recorder that do not affect your overall monitoring/listening levels, it may be desirable to route the outputs of the multitrack recorder back through another set of grouped input faders on the console and route them to a separate set of output buses to your monitoring system. See Figure 4-4.

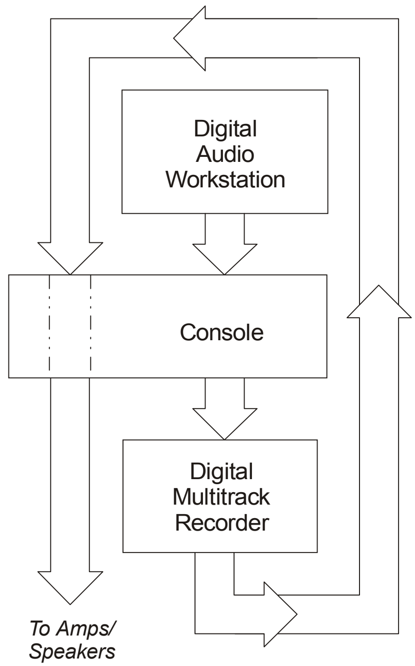

Figure 4-4 Typical Digital Multichannel Audio Workstation Interconnect

4.1.3 Audio Processing Equipment

Audio processors can supplement fundamental sound design and enhance image placement in 5.1-channel productions. Processors are available for every phase of production, from conception and recording to mastering and duplication. Factors to consider when purchasing audio processors for multichannel production include specifications and multichannel functionality.

Multichannel processors are being developed to accommodate the production demands of film, music, and broadcast. Products that offer true multichannel processing and feature sets (e.g., signal alignment and channel linking options) will become increasingly important as multichannel production grows.

4.1.4 Routing/Switching

Routing and switching multichannel material builds on the techniques used in mono and stereo productions. The majority of audio signal routing during production will be done to mono sources through the bus assignment functions of the audio console. Stereo sources can also be routed via the bus assignments, either as grouped stereo or split mono signals depending on the particular console. Switching multichannel programs requires multichannel group functionality. To provide the adequate resources for additional channels of material in both the production and mastering stages of audio creation, both routing and switching capabilities will need to be expanded from existing mono or stereo configurations.

For additional information on multichannel routing and switching as it applies to broadcast production and delivery, please refer to the *Dolby Digital Broadcast Implementations Guidelines* available on the Dolby website, www.dolby.com.

4.2 Recorder/Storage

The evolution of non-linear multitrack digital audio recording and editing have made multichannel production for 5.1 commercial releases attainable by a large community of audio production engineers. While conceptualizing a multichannel project of whatever media type (digital audio workstations or modular digital audio recorders), in both recording and mixing a multichannel audio project it is important to budget the appropriate amount of audio tracks. An audio track may not be limited to fixed audio tape tracks, but rather may be in reference to hard disk space or RAM. For a 5.1-channel audio mixdown, a good starting point is allocating six to eight tracks of audio (or corresponding disk space and/or RAM) for the final delivery tracks at the appropriate bit resolution and sampling rate. For work pertaining to DVD video, it is important to remember the 48-kHz sampling rate for the relationship to video. When audio is used with a visual application such as laser disc, time code is also required. Please review Section 6.4, Time Code for more information on time code as it applies to production.

Common audio formats currently being used include those of digital audio workstations such as Digidesign’s ProTools, Studio Audio’s SaDie, Digital Audio Labs V8 with Minnetonka’s MX51, as well as modular digital audio recorders such as Tascam’s DA88/DA98, Alesis’ ADAT, and Fairlight’s Digital Audio Dubber. While new media formats continue to be introduced into the market, the most common current delivery is on eight-track modular digital multitrack tape such as the Tascam DA88 format. Frequently, times-two mixes will exist on the same delivery medium to support both a discrete 5.1-channel mix, as well as a matrix-encoded Dolby Surround (Lt, Rt) on the remaining two tracks of the eight-track media.

For more information on documentation for mixing and mastering, please review the material in Chapter 6.

Chapter 5
Production Techniques

Many traditional two-channel techniques are applicable to 5.1-channel production. Nonetheless, there is an opportunity to create more convincing and engaging listening experiences.

5.1 Microphone Techniques

Prior to beginning a 5.1-channel production, consider the type and size of the environment to be created. A variety of close, distant, coincident, and spaced microphone techniques can be used to produce a natural-sounding environment. Processing such as artificial delay and reverb may also be used to enhance the listening experience or compensate for compromises made during the recording. Sources can be recorded individually, with or without spatial information, for placement at any point in the 5.1-channel soundfield during mixing. Carefully combining microphone technique, processing, and mixing allows one to place the listener almost anywhere in the desired setting. Although the listener's perspective will most often be the same as that of an audience member, numerous possibilities exist for both realistic and unrealistic environments.

5.2 Recording

Multitrack audio recording technology has undergone great advancements in recent decades. Today, audio professionals have an increasing number of options for recording, editing, and delivering content. The specific allocation of tracks for a 5.1-channel production depends on the project and the mix. If the final product will have no panning movement, such as documenting an acoustic performance, one might simply allot separate tracks for stereo surround ambience. If directional surround placement of prominent elements is desired, more preproduction and additional tracks may be necessary.

It is important to remember that in 5.1-channel audio, the multichannel source delivery master requires three times the amount of data storage as required in stereo production. Regardless of the type of multichannel project, reserve adequate hard drive space or linear tracks for additional channel content.

Refer to Chapter 6 for information on correct formatting and track layout.

5.3 Mixing

With the addition of the Center, Surround, and LFE channels, 5.1-channel mixing presents both interesting choices and unique challenges not present with traditional stereo.

5.3.1 Center/Front Channels

In a stereo program there is only one way to obtain a centrally placed sound image: mix the signal equally to the L/R channels. In a multichannel system, there are three ways:

- Create a phantom center just as with stereo.
- Use the center channel alone.
- Use all three front channels equally or in varied proportion.

Each approach offers advantages and drawbacks. In use since stereo began, the phantom center is well understood. The primary disadvantage of this technique is that the listener must be equidistant from the L/R speakers to achieve proper center imaging. This is rarely the case in the home, and is nearly impossible in a car. Another disadvantage is that due to cross-cancellation effects, the timbre is not the same as from a direct speaker source.

Using the center speaker alone creates a stable center image for every listener no matter where they sit. To prevent the image from sounding too focused or narrow, reverb from the center channel can be spread to the L/R channels.

Distributing the center image amongst the three front speakers allows control of the range of spatial depth and width. A phantom center can be reinforced by some additional signal in the center channel, or a center channel signal can be enhanced with some additional signal spread into the L/R pair. The more channels that are used to carry the same signal, however, the more likely it is that side effects may occur.

Signals might interact with each other causing the phantom image to conflict with the true center image. In systems using dissimilar speakers, or in cases where the listeners are seated off the central axis, the sound arrivals from all three speakers may not blend well. Differences in arrival time can cause a comb-filtering effect, shifts in tonal color, or a smearing of the image. Consider all of these effects when placing the exact same signal in all three front channels. To counteract these effects, process the additional signals first to change their spatial character, timbre, or prominence relative to the main center signal.

5.3.2 Surround Channels

Whereas center-image signals were always part of mixing for stereo, surround channels offer a completely new sonic dimension to consider. Using stereo surrounds is already well established in the film industry; however there is still room for experimentation in the music, multimedia, and broadcast industries. It can certainly be said that the use of surround channels can enhance the sense of depth and space over conventional stereo.

For example, the ambience and room reflections of a concert hall delivered from the surround speakers can drastically change the listener's perspective. Imagine the difference between peering through a window and sitting in a concert hall. Popular music can often benefit as well from a creative use of the surrounds, whether with background singers, instruments, or effects. But as with any new tool or effect, it can be overdone and become tiresome if used to excess. The principle that has served the film industry so well also applies to multichannel mixing: Use the surround channels to enhance rather than distract from the overall experience.

5.3.3 LFE Channel

What is the difference between the LFE channel and the subwoofer signal? The LFE channel is a separate, limited frequency bandwidth signal created by the mixing engineer and delivered alongside the main channels in the mix.

The subwoofer signal is created in the decoder as needed for the particular speaker complement in use, using crossover filters. This signal is created using bass management, and all Dolby Digital decoders perform this function. Through bass management, a subwoofer signal may comprise bass from any channel or combination of channels—typically bass frequencies from channels being replayed on small speakers are directed to the subwoofer speaker. If no subwoofer is present, the bass (including the LFE channel, if it exists) is redirected to the speaker(s) best able to reproduce it, usually the main stereo pair.

Even though an eighth-order, 120-Hz brickwall filter can be applied to the LFE channel by a Dolby Digital encoder, a low-pass filter must be inserted into the LFE signal path during the mix process to ensure proper monitoring. Furthermore, the filter must be applied to the signal being recorded so that the results will be consistent, whether delivered by Dolby Digital or Linear PCM. A maximum cutoff of 80 Hz is suggested when using a typical filter with a gradual slope as compared to the steep, low-pass filter in a Dolby Digital encoder.

In theatres, the LFE channel is used in conjunction with subwoofers to supplement the capabilities of the screen speakers. In most music productions, it is unlikely there will be a technical need to use the LFE channel. Since the overall program level may be adjusted to render any proportion of bass perfectly, the LFE channel might be an advantage only in situations similar to the famous cannon shots in Tchaikovsky's 1812 Overture. In such a case, the overall program level might have to be reduced several dB just so the last few minutes can make the desired impact without overload. By using the LFE channel, the orchestra can be recorded at a normal level, with some of the loudest, deepest bass of the cannons carried in the LFE channel. Of course, the main channels will still carry the cannon shots so that they will be heard from the appropriate locations and in a downmix.

Another benefit to using the LFE channel when carrying explosive bass signals is that smaller stereo systems may not be able to handle such high levels of deep bass without significant stress. Since the Dolby Digital downmix process discards the LFE signal, these low-frequency signals will not present any difficulty for these smaller systems. The remaining portions of the bass frequencies delivered by the main channels will convey the essential aspects of the performance when listening to the downmix.

While it may be of no particular consequence for effects, filters like those used in generating the LFE signal may interfere with the ability to seamlessly blend the LFE channel with the other channels. The best way to ensure a cohesive audio signal across the entire audible spectrum is to maintain its integrity in the main channel(s).

5.3.4 Downmix - Dolby Surround Compatibility

Always check the downmix for Dolby Surround compatibility. Dolby Surround is comprised of mono surround information; therefore stereo surround information in the 5.1-channel mix will be summed and reduced in level to become compatible in the Dolby Surround downmix. While the LFE channel is used primarily for supplemental high impact effects (e.g., explosions, crashes, storms, aerial fly-overs, etc.), delivering crucial low-frequency material exclusively in the LFE channel will produce a Dolby Surround downmix lacking in low-frequency content.

When the 5.1-channel mix is completed, it is often compared with the Dolby Surround mix, if one exists. Several audible differences may be noticed.

- The panning and location of elements within the mix may have changed. Because the 5.1-channel mix is discrete, positioning of elements is easier than through the Dolby Surround matrix.
- Center channel buildup will not be present in the 5.1-channel mix. Mono sounds in the Left and Right channels will stay there, and not appear in the center.
- The 5.1-channel mix increases the potential acoustical dynamic range since there are now five full-range channels instead of two.
- In the 5.1-channel mix the frequency bandwidth of the Surround channel is now full-range, not limited as with Dolby Surround. Also, there are now two Surround channels, not one (stereo, not mono).

If the possibility of delivering the original Lt/Rt exists, this is the preferred method. If it is not possible, then mix compromises may be necessary to compensate for unwanted artifacts caused by the downmix process.

5.3.5 Downmix-Stereo Compatibility

Even with the popularity of Dolby Digital and Dolby Surround systems, there will always be a need to address stereo reproduction. There are three basic ways to accomplish this.

- Prepare a new stereo mix from the original multitrack elements.
- Prepare a studio-adjusted downmix from the multichannel mix.
- Let the decoder derive a stereo downmix.

Created from Multitrack Elements

This option is no different from today's conventional stereo mixing sessions. The stereo mix is carried as PCM or as a separate two-channel Dolby Digital bitstream.

Created from a Multichannel Mix

The second option takes advantage of all the work that has gone into the mixing of the 5.1-channel version, and allows the mixing engineer to quickly derive a stereo version while retaining flexibility in the exact proportions of each channel represented in the final stereo mix.

Created by Decoder

This option does not create a separate mix. In this case, the decoder derives a stereo downmix based on preset formulas. When applicable, consumer decoders will apply dynamic range reduction during the downmix process to prevent overload. All downmix and dynamic range control options can be previewed in the production studio. A range of adjustments is possible and the resultant effect can be checked in advance. For additional information on the downmix process in the decoder, please refer to the *Dolby Digital Professional Encoding Manual*.

5.3.6 Upmixing

Upmixing describes the process where additional new channels of audio information are created from an original soundtrack, for example making a 5.1-channel version from a stereo master.

The term “5.1” has strong appeal to many people. Consumers and professionals alike can sometimes be drawn by the idea that a soundtrack available in 5.1 is somehow better than a stereo or mono soundtrack. However, there are many instances where this may not be true. For example, an old black-and-white movie may sound incongruous with a 5.1 soundtrack, and mono may be the better format in this case. Conversely, there are instances where there is a legitimate need for a multichannel version when the master is not available in the 5.1 format. The question is how to deal with this latter situation.

The most obvious and best solution is the remix. If original elements such as a multitrack source tape or individual *stems* have been archived, then this could provide the source material to create a new 5.1 version. Of course the original mix should be given due consideration, preferably by enlisting the help of the original artist or producer.

A difficulty arises when these elements do not exist and the only source material is the final release master. There is much debate about how best to upmix to create extra channels from a given source. Most approaches rely on the use of phase relationships or timing between signals, based on the simple idea that in-phase material should go to the center channel and out-of-phase material should go to surrounds. See below for a discussion of some of the benefits and consequences of upmixing using this method.

Two-Track Master

It is usually best to transmit two-channel material, be it matrix-encoded or conventional stereo, to the consumer in its existing two-channel format, rather than decoding it in the studio to derive an *artificial* discrete multichannel mix. The results of upmixing will vary from the intent of the original producer/director, especially in the case where extra processing has been added on top of surround decoding in the studio, perhaps to generate an LFE channel or stereo surrounds. Consequently, the sound may fail to meet listener expectations for true 5.1-channel material. Upmixing in this way can never create a truly discrete mix and the effects of the upmixing system will always be audible.

In addition, many consumers do not yet have a full 5.1-channel Dolby Digital playback system, and instead connect their DTV set-top boxes' and DVD players' two-channel outputs to a stereo or Dolby Surround Pro Logic system. Under these conditions, the multichannel soundtracks will be downmixed to mono, stereo, or Dolby Surround matrix-encoded stereo. If the multichannel soundtrack was derived by phase decoding and upmixing a two-track master, there may be downmix quality problems due to signal addition and cancellation as the various channels are combined in the downmixing process. Decoding the downmixed 5.1 content with Dolby Surround Pro Logic will make these problems even more obvious. In nearly every case, the original stereo or Dolby Surround mix will give the best results. If upmixing is absolutely necessary, it is best to use a decoding system without matrix steering or even a simple Hafler matrix. Any steering logic, including that used by Dolby Pro Logic, adds some degree of image instability that is unwelcome, particularly for pure music applications. When using a decoder to upmix, it is important to turn off any surround channel time delays as they can introduce problems when the sound track is downmixed.

Sometimes additional processing is used on top of the surround decoding in the studio, perhaps to generate an LFE channel or stereo surrounds. When this is done, the sound may fail to meet the listener’s expectations for true 5.1-channel material; upmixing in this way can never create a truly discrete mix and the effects of the upmixing system will always be audible.

Other Formats

Dolby analog, or Dolby Stereo matrix-encoded film soundtracks were originally produced from discrete four-track master tapes. While the four tracks on these masters corresponded to L, C, R, and S, during the matrix-encoding process the results were monitored through a matrix *de*coder (4:2:4 monitoring) to anticipate what would actually be heard in the cinema. The final sound heard in the cinema, therefore, can be quite different from that of the four channels played discretely, due to the matrix encoding and decoding. It is therefore a good idea to compare the discrete four-track mix to the decoded Lt/Rt mix, and adjust accordingly. Alternatively, you may simply use a Dolby Surround matrix encoder to create a surround-encoded two-channel mix for DTV or DVD from the four-channel master, and flag the bitstream as matrix surround-encoded.

A few films were released in a six-track format based on magnetic tracks recorded on 70 mm film. These masters require special consideration; for example, there were two different channel configurations available to filmmakers. They first used the conventional LCRS channels and added both Left-Extra and Right-Extra channels to convey extra low-frequency information. The other format had a channel identification similar to the 5.1 seen today.

5.3.7 Translation to Consumer Systems

One of the most important considerations when dealing with multichannel audio is the translation to the final listening environment. Everyone is familiar with the requirements for mono compatibility in stereo mixes; taking this one stage further, thought must be given to how our track will sound in stereo, mono and Dolby Pro Logic, as well as 5.1. Specific attention must also be paid to the available options in dynamic range settings accessible to listeners. The key tool in the checking process will be the professional decoder, as it can mimic all of the possible listening configurations quickly and easily. All of the methods used today to check the mix in a variety of situations are as relevant with Dolby Digital as they are with conventional stereo work. Taking a copy home, listening in the car, or on a small speaker to mimic the sound of radio receivers is crucial. It may also be worth considering the installation of a small consumer-style surround system in the mixing room for quick and easy compatibility checks.

5.3.8 Production Tips

1. More discrete sources available for the mix will result in more control of the soundfield creation. Record as many discrete sources as possible.
2. Echo and reverb effects may need to be rethought for 5.1 mixes. In particular, elements present in the center channel which have echo or reverb effects added will usually require some of the effect to be present in the center channel for a more natural sound. Failure to include the effects for a vocal track in the center channel will usually produce a *naked* sounding vocal. There is much debate over this matter and probably will be for some time.
3. Remember that some listeners will hear a downmix of the 5.1-channel mix. It may be decoded with a Dolby Surround Pro Logic decoder or heard in stereo. It is

important to check the downmixes and to listen for center channel buildup or unwanted surround information from the Pro Logic decoding process. Some effects that work well to create a 5.1 mix do not work with Pro Logic decoding. The 5.1-channel system is discrete, i.e., where a sound is put is where it will stay.

4. Never decode a Dolby Surround track to make it *discrete*. The resulting track will not properly downmix for the consumer. Use the Dolby Digital 2/0 mode with the surround flag set to ON instead.
5. Remember that listeners prefer natural soundfields. Sometimes effects that sound exciting at first can become distracting or tedious with repeated listening.
6. Be sure to document calibration levels, mixing levels, and other small, but important, information for later reference. The 3-dB difference in surround level between film-style rooms and video-style rooms will have a big impact on the mix when set wrong during the encoding process (see 6.2 for more information). Several projects have gone into production with improper surround levels, resulting in questionable mixes and element balances.

Chapter 6
Preparing the Source Delivery Master

When preparing the source delivery master, adhere to accepted standards and practices to ensure proper Dolby Digital encoding.

One of the most common source delivery formats for Dolby Digital encoding is the Hi-8 mm tape used in many popular Modular Digital Multitracks (MDMs). Digital Audio Workstations (DAWs), open reel digital multitracks, and other formats are also used for this application, although infrequently.

6.1 Channel-to-Track Allocation

Whenever possible, Dolby encourages the adoption of channel-to-track allocation described in the forthcoming ITU-R recommendation, *Parameters for Multichannel Sound Recording*. Track layouts depend on channel complement, although tracks 1, 2, and 3 are always channels Left (L), Right (R), and Center (C) respectively. Table 6-1 shows one possible configuration. Since inclusion of the LFE channel is optional and the listener determines its reproduction from a decoder, essential low frequency information should not be mixed exclusively to the LFE channel. When the LFE channel is not used, track 4 may contain a mono Surround (S) signal. Alternative practices exist within various industries, so it is imperative to check the source and accompanying documentation.

Table 6-1 **Channel-to-Track Layout Example**

Channel	L	R	C	LFE	LS	RS	Lt	Rt
Track	1	2	3	4	5	6	7	8

6.2 Channel Levels

Relative channel levels should assume each speaker delivers identical acoustic sound pressure levels to the listener, excluding the LFE channel which is intended for reproduction at +10 dB SPL (with respect to the main channels within the same 3–120 Hz passband). Assuming that a Surround (S) signal is delivered to a single speaker and two Surround signals (LS, RS) are each delivered to individual speakers, Surround levels should be identical to the front channels.

In film sound practice, stereo Surround channel levels are typically recorded +3 dB relative to the front channels. This is done to compensate for the -3 dB Surround levels (relative to the front channels) encountered in cinema monitor systems. In such cases, select the **3 dB Attenuation** option for the LS and RS channels in the Dolby Digital encoder.

When the Surround channel is mono, allocate it to both tracks 5 and 6 with 3 dB attenuation applied to each signal. Use the following formula:

Track 5 = Track 6 = 0.707 * S

Follow this recommendation even when track 4 also contains the S signal, which should always be at normal level on this track. Indicate on the tape label clearly that tracks 5 and 6 each contain the S signal at -3 dB relative to their normal levels.

6.3 Reference Levels

The standard reference level is -20 dBFS for digital recorders (0 VU for analog recorders). This level is typically +4 dBu from professional consoles and -10 dBV from semi-professional consoles. When transferring from 35 mm magnetic film (analog), attenuation and/or peak limiting may be needed to avoid digital clipping. If used, these processes will affect what value is selected for the Dolby Digital Dialog Normalization parameter setting during encoding and will require complementary gain recovery in the reproduction chain.

A 30-second, 1-kHz alignment signal at -20 dBFS should appear on all channels at the beginning of the source delivery master prior to program start. The finished master should contain at least 30 seconds of *digital black* after the alignment signal and before each subsequent program. Each title should begin with at least two seconds of encoded digital black.

6.4 Time Code

Time code plays two important roles in preparing mixes for Dolby Digital encoding.

First, it is common to use some form of SMPTE or MIDI time code for synchronizing recording machines (DA88, ADAT) and digital editors (Pro Tools or MIDI sequencing program) while recording and mixing material. It is important to be aware at the beginning of a project what the final time code 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. Even with the introduction of high-definition video formats worldwide, varying frame rates and time code modes (including drop-frame) will be around for some time to come.

Second, if the material is going to be encoded using Dolby Digital for either DVD or Digital Television, it might be necessary to add a time code stamp to the Dolby Digital bitstream at the time of encoding. This is done by sending the time code from the source material into the Dolby Digital encoder **Time Code** input and selecting **Enable Time Stamp** from the setup section of the Dolby Digital encoder. Dolby Digital has the ability to add a time code stamp to the associated Dolby Digital frame. This stamp can be used to align the Dolby Digital audio file with a compressed video file containing a corresponding time code stamp. This process can be done with either a DVD-authoring station or DTV encoder.

Having the correct time code that matches the picture is essential to proper audio and video synchronization. In addition, it is also very important that the time code source feeding the Dolby Digital encoder be stable and uninterrupted. Always use the time code generated from a digital source such as a DA88 or digital VTR. If unsure of the time code source, it is best to generate clean time code from a synchronizer or use a quality time code regenerator before routing the time code into the Dolby Digital encoder.

6.5 Documentation

Complete, clear, and accurate documentation should always accompany the source delivery master used for Dolby Digital encoding. 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 of this document and on the Dolby website under the *Technical Information* heading at www.dolby.com. The *Mix Data Sheet* provides concise information about the source media to all the engineers on a project. Typically, it will include information on sampling frequency, bit resolution, time code, 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.

6.5.1 Mix Data Sheet

The purpose of a *Mix Data Sheet,* Appendix A is to provide all production engineers with thorough and concise media layout information.

The information contained in the *Mix Data Sheet* should be distributed for technical parameters prior to encoding for final delivery, i.e., in production or postproduction PRIOR to the output distribution (DVD or Digital Broadcast). An important point to note is that all mix data information should be duplicated and should be placed on the master media as well. While size of the recording and production media may be on a smaller scale, accurate labeling of the media with *Mix Data Sheet* information will provide additional engineers with the proper insight of origin.

6.5.2 Mastering Information Sheet

The purpose of a *Mastering Information Sheet*, Appendix A, is to provide the mastering engineer or the digital authoring specialist technical information with respect to media layout, timing information, and encoder-specific information.

This information is created during authoring/creation of the final delivery medium, e.g., Dolby Digital encoding of an AC-3 stream for a movie on DVD.

Additional documentation such as production notes will be invaluable in completing a project. The purpose of **Notes** is to provide engineers with any explanation for key actions with relationship to time, level, error, or artistic consideration. In addition to hard copy, all documentation should be duplicated and affixed to the delivery media.

Chapter 7
Miscellaneous Information

7.1 Technical Assistance

Dolby Laboratories provides technical support to content creators and encoder users in a variety of ways. Many technical documents are available for viewing or downloading on the Dolby website at www.dolby.com. Printed copies of documents may also be obtained by sending an email request to info@dolby.com with a description of the desired documents, and a complete mailing address.

Dolby has a staff of engineers who can assist with soundtrack production, encoding, and trademark usage questions. Dolby engineers are also available to provide on-site assistance with room configuration and calibration, soundtrack production, and encoding. Telephone support is available free of charge, and local on-site support can often be provided without cost. However, in situations where extensive on-site support or long distance travel is required, standard engineering rates may apply.

For technical support, please contact the nearest Dolby office at any of the locations listed on page ii of this document.

7.2 Contacting Dolby Laboratories

In addition to its headquarters in San Francisco, Dolby has several other offices around the world. All offices can provide information on soundtrack production and encoding.

Contact Dolby from anywhere in the world via the following e-mail addresses:

Address	Use to --
tsa@dolby.com	apply for a Dolby trademark agreement (TSA).
hdtv@dolby.com	ask questions on HDTV audio production, encoding, and implementation issues.
tvaudio@dolby.com	ask questions about TV broadcast audio technologies, equipment, and implementation.
dvd@dolby.com	ask questions on audio encoding for DVD.
multimedia@dolby.com	ask questions on multimedia applications.
info@dolby.com	request general information and make inquiries.

7.3 General information and Inquiries

A wide variety of technical and trademark information can be found on Dolby's website at www.dolby.com.

Please contact the nearest office, as listed in this document, for direct assistance.

7.4 Trademark Usage

Dolby Laboratories encourages use of the Dolby Digital trademark to identify soundtracks that are encoded in Dolby Digital. This is an effective way to inform listeners of the soundtrack format, and the use of a standard logo promotes easy recognition in the marketplace. As with any trademark, the Dolby Digital logo may not be used without permission. Dolby Laboratories provides a royalty-free Trademark and Standardization Agreement (TSA) for companies who wish to use Dolby trademarks. The company that owns the program material being produced must sign this agreement. Recording studios or production facilities that provide audio production, encoding, or manufacturing services for outside clients generally do not require a trademark license. However, we do ask that these facilities refer their clients to us for trademark licensing information.

To use the Dolby Digital logo, apply for a Dolby Trademark and Standardization Agreement (TSA) by sending email to tsa@dolby.com or by contacting Dolby Laboratories at any of the locations given in this document. When sending written requests, please indicate whether you would like a Dolby Digital or Dolby Surround trademark license, or both; also include your name, company name, mailing address, and the type of media your soundtracks, music recording, etc., will be distributed on (such as DVD, DVD-ROM, DTV broadcast, etc.).

For detailed information on Dolby trademark licensing, please refer to the document *Use of Dolby Trademarks on Audio and Video Media*, available on the Dolby website at www.dolby.com. A licensing application form is also available through our website.

If you are already a Dolby licensee and would like more information on trademark usage, please contact Dolby Laboratories. We are always happy to review artwork and assist with the proper use of our trademarks.

Appendix A
Mix and Mastering Data Sheets

Dolby

Mix Data Sheet

Date ______ / ______ / ______

Project ____________________ **Project #** ____________________

Client ____________________ **Producer** ____________________

Studio ____________________ **Engineer** ____________________

Sampling Frequency	☐ 32 kHz	☐ 44.1 kHz	☐ 48 kHz	☐ 96 kHz
Bit Resolution	☐ 16-bit	☐ 18-bit	☐ 20-bit	☐ 24-bit
Time Code Format	☐ 25 fps	☐ 29.97 NDF		
Tape Format	☐ ADAT	☐ DA-88	☐ ½" Digital	☐ Data Cartridge (JAZ)
Surround Level SPL Calibration	☐ Equal to Front	☐ -3 dB to Front		

CH1	CH2	CH3	CH4	CH5	CH6	CH7	CH8

Time Code	Program
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	

Notes: ____________________

Dolby Mastering Data Sheet

Date ____ / ____ / ________

Project		**Project #**	
Client		**Producer**	
Studio		**Engineer**	

Sampling Frequency	☐ 32 kHz	☐ 44.1 kHz	☐ 48 kHz	☐ 96 kHz
Bit Resolution	☐ 16-bit	☐ 18-bit	☐ 20-bit	☐ 24-bit
Time Code Format	☐ 25 fps	☐ 29.97 NDF		
Tape Format	☐ ADAT	☐ DA-88	☐ ½" Digital	☐ Data Cartridge (JAZ)

Dolby Digital Encoding Information	Bitstream Information	Processing
Audio Service Configuration	☐ Default	☐ Digital De-Emphasis
LFE Filter ☐ ON ☐ OFF	☐ Copyright Bit	☐ DC High-Pass Filter
Dial Norm Setting:	Center Mix Level:	☐ Bandwidth Low-Pass
Mix Level:	Surround Mix Level:	☐ LFE Low-Pass Filter
Data Rate:		

Surround Channel Processing	Dynamic Range Compression			
☐ 90-Degree Phase Shift	☐ None	☐ Speech	Film:	Music:
☐ 3 dB Attenuation			☐ Std.	☐ Std.
			☐ Light	☐ Light

Mode	CH1	CH2	CH3	CH4	CH5	CH6	CH7	CH8

Time Code	Program
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	
: : ;	