Setting up for Surround Sound on Commercial Speaker Layouts

In this lecture we focus on preparation for surround mixing. For this, we need to consider the loudspeaker layout, the media we are mixing for, calibration of loudspeakers, and mix preparation in a sequencer.

First…. A little bit about Codecs!

Before we talk about loudspeaker layouts it is worthy to introduce the concept of codecs, since the choice of codec has a direct bearing on the reproduction layout and vice versa. One of the major problems with putting multichannel audio onto a storage medium is the amount of space available to accommodate high resolution multiple audio streams. There are four main ways to overcome this problem whilst maintaining an acceptable level of quality in the audio. These are:

1. **Matrixed Audio:** Here the multichannel audio is reduced down to two channels via a matrix encoder and stored onto a medium. On playback, a matrix decoder decodes an estimate of the original audio. This method typically uses sum and difference techniques to encode and decode the audio. Side effects include cross talk in the channels, reduced stereo imaging and reduced localisation accuracy. Matrixing techniques were originally developed during the quadraphonic era, but are still in use in Dolby Surround, Dolby Pro Logic and Pro Logic II systems for example.

2. **Perceptual (Lossy) Encoders and Decoders:** In the same way that MP3 is a perceptual encoder that allows for a reduced data rate, DTS and Dolby Digital (AC-3) are multichannel encoders of the same fashion. Perceptual encoders divide the audio spectrum of each channel into narrow frequency bands of different sizes optimized with respect to the frequency selectivity of human hearing. This makes it possible to sharply filter coding noise so that it is forced to stay very close in frequency to the frequency components of the audio signal being coded. By reducing or eliminating coding noise wherever there are no audio signals to mask it, the sound quality of the original signal can be subjectively preserved. In this key respect, a coding system like AC-3 is essentially a form of very selective and powerful noise reduction.

3. **Lossless Encoders and Decoders:** These codecs use compression algorithms that preserve, bit-for-bit the original multichannel audio. Many use numerical coding techniques similar to the schemes found in Zip file compression. MLP (Meridian Lossless Packing) is an example of a lossless encoding scheme and is used in DVD-Audio discs.
4. **PE/MA Hybrids**: In loudspeaker layouts that have more than 5.1 channels, we often find codecs that use a combination of Perceptual Coding and matrix encoding. The former is typically used to convey the 5.1 channels and the latter to obtain the extra surround channels. Dolby Pro Logic IIx and DTS-ES Matrix are both codecs of this type, for use in 6.1 surround sound.

**Loudspeaker Layouts**

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

- **3-1 ch**: This method is based on a two-channel system (L, R), and adds a center channel (C) and surround channel (S). Although there may be two surround speakers, one each at left and right, the rear playback is monaural. The “3” in “3-1” indicates L, C, and R, and the “–1” indicates S. Note that if “3-1” is expressed as “3.1,” this means “L, C, R” + “LFE”.

- **5.1 ch**: This method is based on the 3-1 ch system, but changes the surround to stereo (LS, RS) and adds an LFE (Low Frequency Effect) channel for low-frequency effects. The LFE channel is played back through a dedicated subwoofer designed for low-frequency playback.

- **6.1 ch**: This method is based on the 5.1 ch system, and adds a new back-surround channel (BS). If two speakers are provided to play back the back-surround channel, these are sometimes called BSL and BSr, but the signal that is played back is a monaural signal where BSL = BSr.

- **Other**: 3-2 (or 5.0, without LFE) and 2-2 (or Quad, without C and LFE), which are based on 5.1ch but do not use specific channel(s) of them.
7.1 Layouts and Codecs:

Disc mediums such as HD-DVD and Blu-ray accommodate 8 discrete channels of audio, allowing for commercial 7.1 playback. Thus commercial 7.1 systems have found their way into the domestic market. Codecs which use 7.1 layouts include:

**Dolby ProLogic IIx**: uses matrix logic processing within the decoder to stereoize the back channels (LB, RB in diagram below).

**Dolby Digital Plus** delivers discrete 7.1 AC3 channels.

**Dolby True HD** delivers discrete 7.1 PCM (with Meridian Lossless Packing).

![Diagram showing 7.1 speaker layout]

**DTS-HD Master Audio**: **DTS-HD Master Audio** is a lossless audio codec. It is an extension of DTS which, when played back on devices which do not support the Master Audio extension, degrades to a 1.5 Mbit/s "core" track which is lossy. DTS (Digital Theatre Systems) recommend several different listener configurations for DTS-HD listening. These are shown below:
Traditional

Full Rear Surround

Side High
SDDS:

Another speaker arrangement in current use is the theatrical 7.1 format used by Sony SDDS™. This places five full-range speakers behind the screen — left, left center (“left extra”), center, right center (“right extra”), and right — along with two arrays of rear surround speakers, plus a subwoofer. The main advantage of this system is improved localization, particularly when applied to dialog. For example, two "talking heads" onscreen might be assigned to the left center and right center speakers, with the soundtrack panned hard left and right and sound effects routed to the center channel.

Other Interesting Codecs:

Dolby Digital Live:

Dolby Digital Live is designed for gaming technology and converts any audio signal into a Dolby Digital bitstream for transport and playback through a home theater system. With it, a PC or games console can be hooked up to your Dolby Digital-equipped audio/video receiver or loudspeaker system via a single digital connection (usually Optical or HDMI).

Systems using Dolby Digital Live technology can provide Dolby Digital (5.1-channel surround sound) during gameplay, immersing players in high-quality surround sound that puts them at the center of the action.

DSD:

Although not really a codec, DSD takes a completely different approach to reproducing multichannel audio than existing PCM based systems. It is the lossless technology used to record and produce audio content on the SACD. DSD is a 1-bit representation of the
audio waveform with 2.8224Mhz of sampling. This allows SACD to achieve very high resolution audio quality.

**What are you mixing?**

What we are mixing will partially dictate what the loudspeaker layout and codec employed is. For instance, if we are mixing music for film, we may want to do it over a 7.1 system with DTS-HD or Dolby Digital True HD. For mixing games, we might employ Dolby Pro Logic II on a 5.1 system. The reference table below shows what layout, codec and media type could be used for music, game and film.

<table>
<thead>
<tr>
<th>Type</th>
<th>Layout</th>
<th>Codec</th>
<th>Media</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cinema</td>
<td>2.0</td>
<td>Dolby Stereo</td>
<td>35mm Film</td>
</tr>
<tr>
<td></td>
<td>5.1</td>
<td>Dolby DIGITAL, DTS</td>
<td></td>
</tr>
<tr>
<td></td>
<td>7.1</td>
<td>Dolby TrueHD, DTS-HD, SDDS</td>
<td></td>
</tr>
<tr>
<td>DVD-Video</td>
<td>2.0</td>
<td>PCM, Dolby Pro Logic</td>
<td>DVD</td>
</tr>
<tr>
<td></td>
<td>3-1</td>
<td>Dolby Pro Logic</td>
<td></td>
</tr>
<tr>
<td></td>
<td>5.1</td>
<td>Dolby DIGITAL, DTS</td>
<td></td>
</tr>
<tr>
<td></td>
<td>6.1</td>
<td>Dolby Digital EX, DTS-ES</td>
<td></td>
</tr>
<tr>
<td>HD-DVD Video</td>
<td>7.1</td>
<td>Dolby TrueHD, DTS-HD, SDDS</td>
<td>Blu-Ray, HD-DVD</td>
</tr>
<tr>
<td>Music</td>
<td>2.0</td>
<td>PCM</td>
<td>CD, DVD, dual disc</td>
</tr>
<tr>
<td></td>
<td>3-1</td>
<td>PCM, Dolby Pro Logic</td>
<td>DVD</td>
</tr>
<tr>
<td></td>
<td>5.1</td>
<td>PCM, Dolby Pro Logic 2, Dolby DIGITAL, DTS</td>
<td></td>
</tr>
<tr>
<td></td>
<td>6.1</td>
<td>PCM, Dolby Digital EX, DTS-ES</td>
<td></td>
</tr>
<tr>
<td></td>
<td>7.1</td>
<td>PCM, Dolby True HD, DTS-HD</td>
<td>HD-DVD, Blu-Ray</td>
</tr>
<tr>
<td>SACD Music</td>
<td>2.0</td>
<td>DSD</td>
<td>SACD</td>
</tr>
<tr>
<td></td>
<td>5.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Gaming</td>
<td>2.0</td>
<td>Dolby Stereo, PCM</td>
<td>DVD</td>
</tr>
<tr>
<td></td>
<td>3-1</td>
<td>Dolby Pro Logic</td>
<td></td>
</tr>
<tr>
<td></td>
<td>5.1</td>
<td>Pro Logic II, Dolby Digital Live (DDL)</td>
<td></td>
</tr>
<tr>
<td></td>
<td>7.1</td>
<td>Dolby True HD</td>
<td>Blu-Ray, HD-DVD</td>
</tr>
</tbody>
</table>
Configuring a loudspeaker setup for 5.1 (Recommendation ITU-R BS. 775-1):

The main features of the ITU-R placement are as follows.

1. L/R angle of separation = 60°: This emphasizes compatibility with conventional audio listening environments (an equilateral triangle consisting of L<>R<>listener).

2. Surround speakers (LS, RS) placement angle = 110° ±10° (with C located at 0° in the plane)

3. Height of each speaker = 1.2 m (listener ear height): The surround speakers (LS, RS) may be placed higher than L, C, and R as long as the elevation angle is within 15°. Surround speakers (LS, RS) are placed at the sides rather than at the rear. It is one of the most effective placements in order to supply lateral information that is lacking in conventional L/R two-channel playback.

Subwoofer placement

Although Rec. ITU-R BS. 775-1 mentions systems that include an added LFE system (optional), it does not specify the placement of the sub-woofer speaker for playback. However, the playback bandwidth is specified as 20 Hz--120 Hz.
When placing the sub-woofer, we must take the acoustics of the room into account. For example, placing the sub-woofer in the corner of the room may produce problems in the frequency response due to disruptions caused by standing waves. The figure below shows an example of the measured relationship between the sub-woofer location in a listening room and the frequency response. It can be seen that the frequency response changes in various ways depending on the location of the sub-woofer.

**Frequency responses**

![Graph showing frequency response](image)

In some cases, placing two sub-woofers in appropriate locations can stabilize the playback environment. The figure below shows an example of calculations performed to simulate the differences in sound pressure distribution between one and two subwoofer sources. You can see that playback using two sources produces less variance of sound pressure distribution across the width (W-axis) of the room than a case in which only one source is used. If two sub-woofers are placed across the width of the room in this way, changes in sound pressure level will be mainly limited to the depth (D-axis) of the room.

**Room plan**

![Room plan diagram](image)

**Monitor alignment**

![Monitor alignment diagram](image)
In two-channel playback, placing the L and R speakers at the left and right of the listening point and playing them back at the same power will allow both L and R to be received at the same power and the same timing. However for multi-channel playback, simply powering all speakers equally is not usually enough to ensure that all channels including LFE are received at the appropriate level balance and the same timing. In most cases, a process of measurement and adjustment is required when setting up a multi-channel listening environment. To measure the monitor response, we need pink noise as a source signal (pink noise is filtered to give equal power per 1/3 octave), and a sound level meter to measure the playback sound pressure level of the speakers.

Method: Play back pink noise from the main speakers, and adjust the gain of each amp so that the sound pressure level of each speaker is 85 dBC at the listening point. The sound level meter must be set to “slow” response and the “C-weighted” frequency curve.

Note that because we are using the C-Weighting, the sound pressure level indicated by the SPL meter is the “all-pass level” or the sum of all the levels of all frequency bands.

![SPL meter A and C frequency responses](image)

Side note: Remember when adding SPLs use the following equation:

$$L_\Sigma = 10 \cdot \log_{10} \left( 10^{L_1/10} + 10^{L_2/10} + \cdots + 10^{L_n/10} \right) \text{ dB}$$

So note,

80dB + 80dB is not 160dB, its 83dB!!

So on the meter, if the all-pass level is 85 dBC, the 1/3 octave band level will be approximately 71 dB for each of the 31 frequency bands. Note that adjustment based on the all-pass level (85 dBC) using only a sound level meter is an easy method that is possible only if all speakers have the same playback response and the room acoustics are sufficiently good. In other measurements, it is desirable that you check not only the all-pass level (85 dBC), but also use a real time spectrum analyzer to check the band levels (71 dB, 1/3 octave).
**LFE Level:**

*In Theory.......*

A 5.1 playback environment for DVD-Audio or Super Audio CD differs from the 5.1 playback environment for DVD-Video in the playback level of the LFE channel. For DVD-Audio or Super Audio CD, the LFE channel is treated exactly the same as other channels. The frequency bandwidth of the LFE in DVD-Audio differs from DVD-Video in that since an LPF is not applied during encoding, full-range recording and playback is possible. However as stated in the “DVD-Audio Software Production Guidebook (Supplemented Edition),” it is desirable that a LPF be applied during production to the LFE master source in order to maintain compatibility for a variety of end-user playback environments.

Thus for DVD-Audio and Super Audio CD, note that the LFE playback level must be the same as the other channels. **However with DVD-Video, it must be 10dB higher!**

Attention must be paid to the playback level of the LFE signal particularly when producing DVD-Audio and DVD-Video hybrid multichannel discs. For example, in order for an LFE signal produced in a DVD-Audio environment to be converted for use with DVD-Video, the LFE master signal must be recorded at a level 10 dB lower.

![Loudspeaker response for DVD-Audio](image)

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

Note that it is a mistake to adjust the amp gain so that the pink noise playback level shown by the sound level meter is 95 dBC (=85dBC+10dB), because we are not using the full spectrum in the subwoofer (only 20-120Hz).

The most reliable way is to use a Real Time Spectrum Analyzer, and make adjustments so that the 1/3 octave band levels are approximately 81 dB. In this case, the all-pass level indicated by the sound level meter will be approximately 89 dBC, not 95 dB (=85dBC+10dB).

In cases such as DVD-Audio or Super Audio CD, where you set the band level of the LFE channel to the same level as the band level of the main channels (±0 dB), the all-pass level shown by the sound level meter will be approximately 79 dBC (if we assume the LFE playback bandwidth to be 20 - 120 Hz).

**IMPORTANT!**

**And in practice…..**

The above theory has caused massive confusion in the audio industry in terms of handling of the LFE channel. It has been my experience, (and the experience of more well seasoned mastering engineers) that in the case of dual DVD-Video/DVD-Audio discs you must pay special attention to LFE handling. Most DVD player manufacturers do not provide automatic adjustment of the LFE on DVD-Audio discs, and so quite often LFE playback will be 10dB too loud! The best solution to this is to remember that the
other channels are FULL RANGE (0 – 20,000 Hz). Most domestic surround sound systems have bass management which route a portion of the low frequency audio from all the surround channels into the subwoofer anyway (More on this later!). So, for DVD-Audio, ask yourself the question “Do I really need to use the LFE channel here?” In most cases, you probably don’t.

**Time Alignment:**

It is crucial that all surround signals arrive at the listening point at the same time. In situations where equidistant speaker placement is impossible it is necessary to delay compensate the loudspeakers. The electrical delay time of the nearest speakers should be such that the sound from the farther speakers arrives at the listener at the same time. The propagation time $T_i$ can be calculated as

$$T_i = \left( \frac{D}{C} \right) \text{ Secs}$$

where $D$ is the distance from the speaker to the listener in meters and $C$ is the speed of sound.

So for example, if we have a 5 loudspeaker setup with 4 loudspeakers at 2 meters away and 1 loudspeaker at 2.2 meters away, then the other four loudspeakers must be delayed to accommodate synchronous arrival of all 5 sources at the listener position. In this case the electrical delay time $\delta T_i$ would be

$$\delta T_i = \left( \frac{D_S - D_i}{C} \right)$$

where $T_s$ and $D_S$ are the arrival time and propagation distance of the farthest loudspeaker.

**Configuring a console for surround sound:**

When using a multitrack device/sequencer with multiple outputs, the mixing console must be configured to route the audio to the speakers. There are two ways of accomplishing this.

1. **Via Output busses:**
   In the same fashion that you would route multiple microphone signals to a multitrack recorder via the mix busses, you can also route your tape returns (sequencer outputs) to the speakers via the mix busses. Patch the loudspeakers into busses 1-6 (in a 5.1 setup). Then route the corresponding channels from your sequencer to these mix busses,

2. **Via Auxiliary Channels:**
   In the same fashion that you would route tape return signals to hardware effects processors, you can route the individual 5.1 channels through auxiliary sends to the surround speakers.
It is also important to be aware of the correct LFE monitoring level. Further more, if a digital mixing console is used, the appropriate calibration gain and delays can be implemented digitally.

**Setting up Cubase/Nuendo busses for surround sound**

**Output bus configuration**

Before you can start working with surround sound in your sequencer, you have to configure a surround output bus, through which all the speaker channels of the chosen surround format are routed.

1. Open the VST Connections window from the Devices menu.
2. Click the “Outputs” tab.
3. Click the “Add Bus” button and select one of the preset formats from the Configuration pop-up. The new bus appears with the ports visible.

4. By clicking in the Device Port column you can now route the speaker channels to the desired outputs of your audio hardware.
5. If you like, rename the output bus by clicking its name and typing in a new one. This name will appear in the mixer and on routing drop-downs. The following surround configurations are included:
<table>
<thead>
<tr>
<th>Format</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>LRCS</td>
<td>LRCS refers to Left Right Center Surround, where the surround speaker is center-rear positioned. This is the original surround format that first appeared as Dolby Stereo in cinema and later as the home cinema format Dolby ProLogic.</td>
</tr>
<tr>
<td>5.0</td>
<td>This is the same as 5.1 (see below) but without the LFE channel. The LFE channel is optional in 5.1 and if you don’t plan to use it, you might find this option more convenient.</td>
</tr>
<tr>
<td>5.1</td>
<td>This format is one of the most popular in cinema and DVD. In its various cinema and DVD encoding implementations (established by different manufacturers) it is referred to as Dolby Digital, AC-3, DTS and MPEG 2 Multichannel. 5.1 has one center speaker (mainly used for speech) and four surround speakers (for music and sound effects). Additionally a sub-channel (LFE - Low Frequency Effects) with lower bandwidth is used for special low frequency effects.</td>
</tr>
<tr>
<td>LRC</td>
<td>Same as LRCS, but without the surround speaker channel.</td>
</tr>
<tr>
<td>LRS</td>
<td>Left-Right-Surround, with the surround speaker positioned at center-rear.</td>
</tr>
<tr>
<td>LRC+Lfe</td>
<td>Same as LRC but with an Lfe sub-channel added.</td>
</tr>
<tr>
<td>LRS+Lfe</td>
<td>Same as LRS but with an Lfe sub-channel added.</td>
</tr>
</tbody>
</table>