GENELEC®

# LOUDSPEAKER SET-UP FOR MULTICHANNEL MONITORING

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# INTRODUCTION

A modern audio production facility must be able to serve productions in a large number of different formats. The change from mono and stereo to multichannel reproduction has produced many problems, both in converting existing production facilities to multichannel format and in new installations.

The audio formats that must be handled by a modern production facility include currently:

- Mono, stereo
- Matrix four channels
- Five channels (5.0 systems)
- Five channels with an LFE channel (5.1 systems)
- Six channels (with either discrete or matrixed Rear Centre channel) with an LFE channel (6.1 systems)
- Other multichannel formats such as 7.1, 10.2 and more

This paper discusses multiple practical questions about the monitoring loudspeakers, their set-up and possible sources of problems, which should be avoided. A brief overview of the current multichannel formats and dedicated sections on LFE Channel and Bass Management are also included.

# A INTERACTION BETWEEN LOUDSPEAKERS AND ROOMS

## 1 Radiation Space in a Room

Probably the majority of the audible problems in monitoring are due to the effects of the room on the sound radiated by the loudspeaker. Therefore, the placement of the loudspeaker in a room is critical. Since a loudspeaker and its subwoofer radiate very long wavelengths at low frequencies, the various cancellation effects and standing waves in the room will affect the loudspeaker/subwoofer performance.

Because the space has a strong effect on the radiation, it is important to understand the conditions. The radiation space is typically characterised by a rough estimate of the solid angle (part of a sphere) into which the loudspeaker is radiating (see Figure 1). As the loudspeaker/subwoofer is driven, it creates a certain volume flow, which naturally spreads into all directions. As we limit the space seen by the loudspeaker and at the same time we keep the total power identical the energy density (intensity) in the limited radiation space increases. Hence, *reducing the radiation space increases Sound Pressure Level (SPL).* Every halving of the radiation space doubles the SPL.

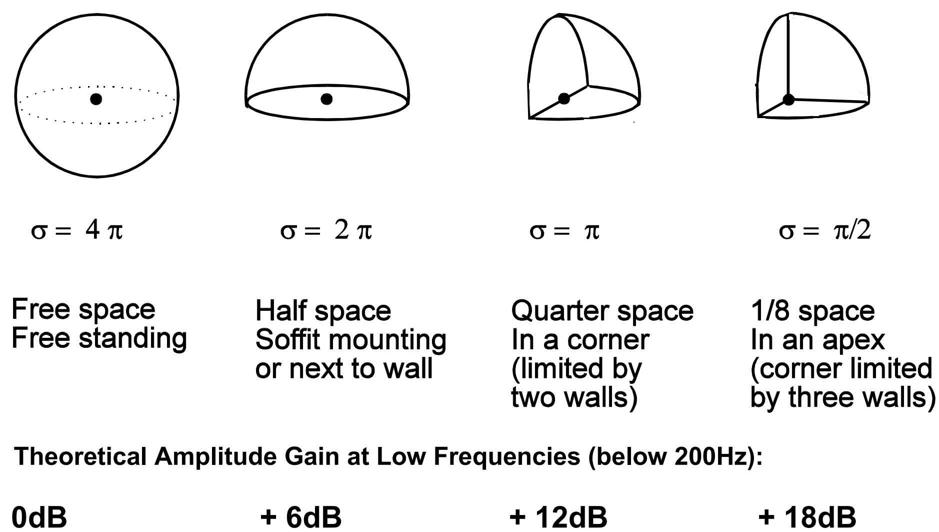


**Figure 1 - Different values of solid angle, respective amplitude gain and loudspeaker alignment.**

Concerning the usual placement of a loudspeaker/subwoofer, we notice that:

- Against one solid boundary, which is large compared to the wavelength, the radiation space is $2\pi$, and the theoretical amplitude gain is +6dB at low frequencies. This applies typically to flush mounting.

- A subwoofer is typically placed on the floor and against a wall, hence we get two boundaries, and the radiation is now into solid angle of $\pi$, and the amplitude gain is +12dB.

- If a loudspeaker or a subwoofer is located in a corner, on the floor, the radiation space is again halved and the amplitude gain is then +18dB.

It is essential to note that in this context the words like ”large”, ”close”, etc. refer always to the wavelength. ”Large” compared for example to 10 m (34 Hz) is different from ”large” compared to 3.4 m (100 Hz) or 3.4 cm (10 kHz). This means that when a loudspeaker is placed ”far” from boundaries at, say, 150 Hz, it probably is very ”close” at 30 Hz. The radiation space depends on frequency and therefore it is important to be able to correct the response of the loudspeaker/subwoofer so that the final acoustic performance stays as flat as possible.

## 2 Wall behind the Loudspeaker Cancellation

The mechanism of cancellation is very simple. When two identical signals are in anti-phase (180 degrees out of phase), they cancel each other. If the loudspeaker is a quarter wavelength away from a reflective wall, the reflected wave comes back to the loudspeaker in anti-phase (phase difference of half a cycle, see Figure 2) and thus cancels the original signal at that frequency. How complete the cancellation is depends on the distance and the reflection coefficient of the wall. Longer distance means that the amplitude of the reflected signal is lower and thus the cancellation is not complete.

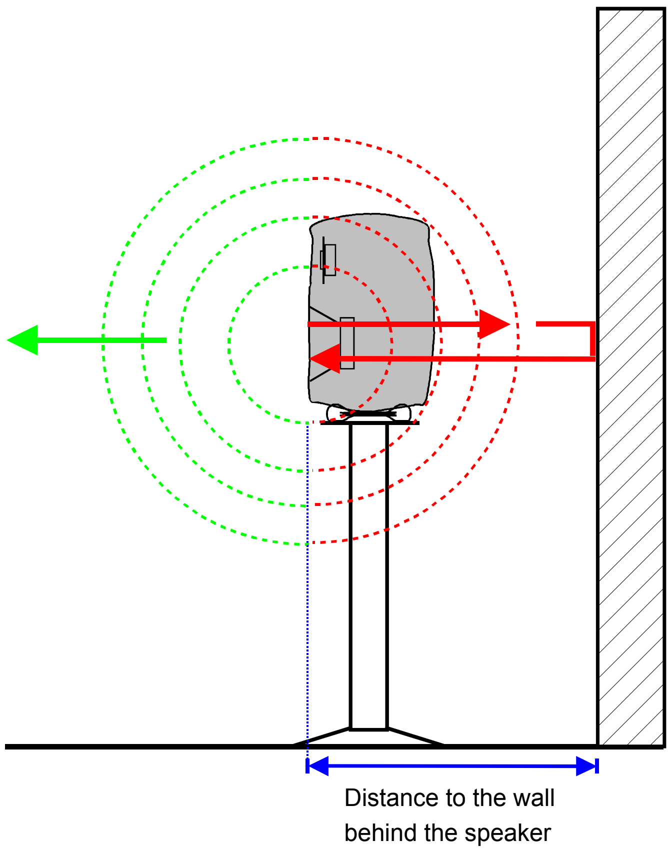


**Figure 2 - Wall behind the Loudspeaker Cancellation Phenomena (Distance to wall = ¼ Wavelength distance)**

In real life the depth and width of the *cancellation dip* varies depending on the level of the reflection, but in most cases it is well audible. No equalisation will cure this situation because it originates from interference; adding amplitude at the dip frequency will also boost the reflection and thus their sum remains the same. This simple case deals with one mode only: reflection from one wall behind the loudspeaker. Table 1 shows various distances between a wall and the front baffle of a loudspeaker with their corresponding dip frequencies.

| Distance from wall, in meters | First cancellation frequency, Hz (D=λ/4) |
|---|---|
| 0.10 | 858 |
| 0.20 | 429 |
| 0.40 | 214 |
| 0.60 | 143 |
| 0.80 | 107 |
| 1.00 | 86 |
| 1.20 | 71 |
| 1.40 | 61 |
| 1.60 | 54 |
| 1.80 | 48 |
| 2.00 | 43 |
| 2.20 | 39 |
| 2.40 | 36 |
| 2.60 | 33 |
| 2.80 | 31 |
| 3.00 | 29 |

**Table 1 - First cancellation frequency at different distances from the wall**

The **first** and best cure for the ‘wall behind the loudspeaker’ cancellation dips is to *flush mount* the loudspeakers in a hard wall – also called ‘infinite baffle’ or ‘soffit mounting’ (which totally eliminates this wall reflection and cancellation) – see Chapter 6.

**Second** best is *placing the loudspeaker very close to the wall*, which raises the cancellation frequency higher. This works well when the loudspeaker is not too small. The risk is, with small loudspeakers which inherently are less directional in mid frequencies, that the dip just moves to the low mid-band and causes even worse coloration. As seen above, the distances between 0 and 20 cm from the wall let the loudspeaker response to be, in most cases, unaltered; i.e. the directivity of the loudspeaker is high enough so that the rear radiation cannot cause a severe cancellation. Additionally, the low frequency boost should be compensated for when the loudspeaker is mounted close to the wall (+6 dB).

Alternatively, the **third** cure is to move the loudspeaker considerably *away from the wall*: the cancellation frequency goes down so far that it is below the low frequency cut-off of the loudspeaker. Thus, the minimum distance ‘loudspeaker/wall behind’ depends on the loudspeaker low frequency performance. However, at low frequencies and for large loudspeakers, the minimum distance becomes very long and impractical. At the same time, the distances to other boundaries in the room become similar to the desired distance to the wall behind the loudspeaker, and the reflections from these other surfaces start to dominate the response.

The **fourth** cure is to make the wall so absorptive that the reflected amplitude is so small that it will not cancel the direct sound. The thickness of a porous absorber has to be one quarter of the wavelength of the frequency to be absorbed to become effective. This is the same distance that determines the frequency of the cancellation dip, therefore, the absorber has to be very thick and, in most cases, this is not done.

## 2.1 Free Standing Loudspeakers

Although flush mounting loudspeakers (see Chapter 6) offers many benefits it is expensive and in most cases, especially in smaller installations, the loudspeakers are left free standing. As stated above reflections from various boundaries will characterise the performance of a free standing loudspeaker.

Genelec 8030A, 8040A and 8050A should be placed so that a minimum gap of 5 cm (2”) is left behind the loudspeaker for amplifier cooling and rear opening reflex port sound radiation. In general, when positioning the loudspeaker’s front baffle further than 30 cm from the wall, a reflection can cause a cancellation in the low frequency response and hence a loss of bass sound quality. For two-way loudspeakers, LF cancellations in the range 40…80 Hz should definitely be avoided (see Figure 4). Low frequency cancellations in the 80…200 Hz range should also be avoided where possible. If this is not possible the sound quality can still be perceived to be good. Translating these frequency ranges into distance recommendations shows that a good response can be achieved at distances from the wall up to 1 m (see Figure 3). Beyond that, 1 m…2.2 m should definitely be avoided. Loudspeakers placed greater than 2.2 m, may suffer from a cancellation in the very low frequency region around the LF cut-off thereby compromising the loudspeaker’s LF extension. So the lower the LF cut-off, the further away the loudspeaker must be placed from the wall.

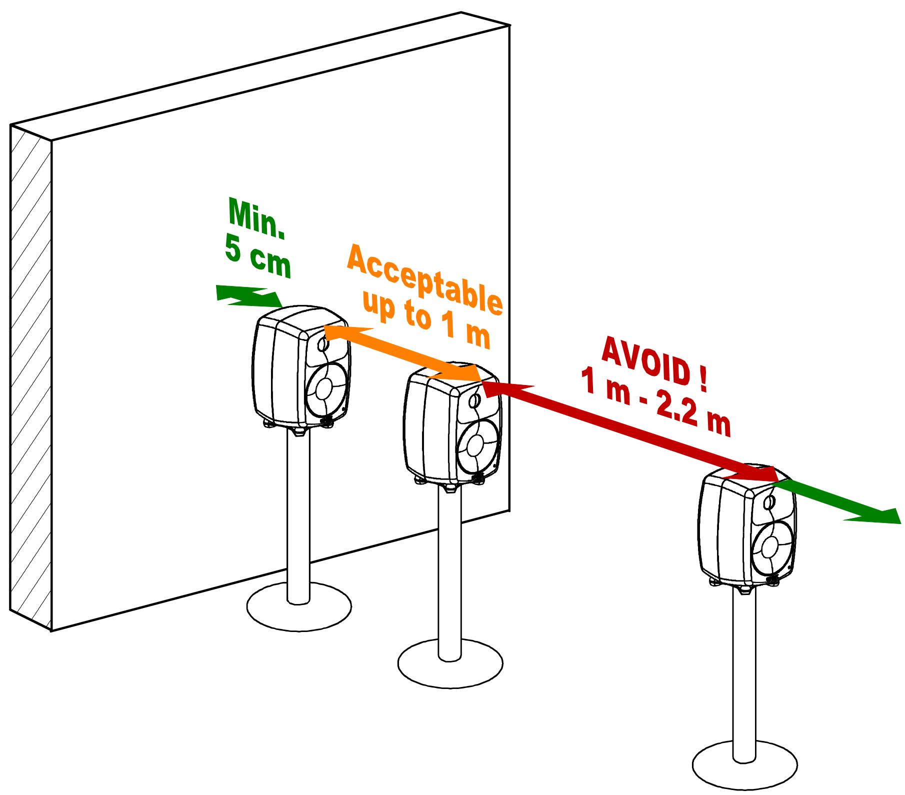


**Figure 3 - Distances from a single wall to the front baffle of free standing loudspeakers. Correct (green), acceptable (orange) and avoid (red).**

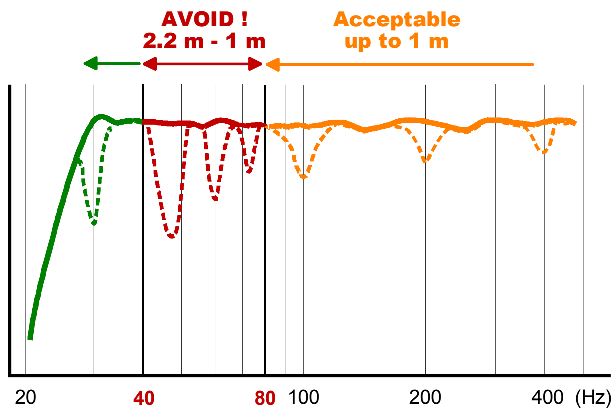


**Figure 4 - Frequency domain notches and distances from the single wall behind a free-standing loudspeaker and its front baffle**

Two observations are immediately apparent:

- For larger loudspeakers placed away from the wall the necessary distance is far too long for any practical rooms.
- In such cases, the distances to ceiling and walls are already smaller than the distance to the wall behind the loudspeaker. Reflections from these surfaces are important and might become more significant.

In practice free standing loudspeakers always suffer from many cancellations induced irregularities of their frequency response.

## 2.2 Free Standing Loudspeakers Combined with Subwoofers

When using subwoofers the additional crossover (at 85 Hz) between the two loudspeaker systems changes the whole monitoring configuration. The subwoofer itself should be placed close to the wall(s) in order to maximise its efficiency – maximum distance of 60 cm (see Figure 5). This also eliminates most of the possible sources of cancellation dips in the subwoofer response, because the subwoofer is acoustically close to the boundaries.

'Satellite' loudspeakers high-passed at 85 Hz do not have to reproduce very low frequencies so they may be placed at a distance where LF notches do not occur in their passband. The guidelines for placing 'satellite' loudspeakers are similar to the ones for free-standing loudspeakers (see Figure 5).

The 'acceptable' distance extends out to 1.1 m due to the fixed LF cut-off of the 'satellite' loudspeakers. From 1.1…2 m, the loudspeakers may be placed without serious compromises from the wall behind the loudspeaker's reflection and corresponding cancellation effects. Although Genelec subwoofers provide accurate phase control facilities at the crossover point, the 'satellite' loudspeakers should not be placed too far (max. 2 m) from the subwoofer. If this is the case, the tonal balance between the satellite loudspeakers and the subwoofer may differ considerably due to excitation of different room modes by the sources.

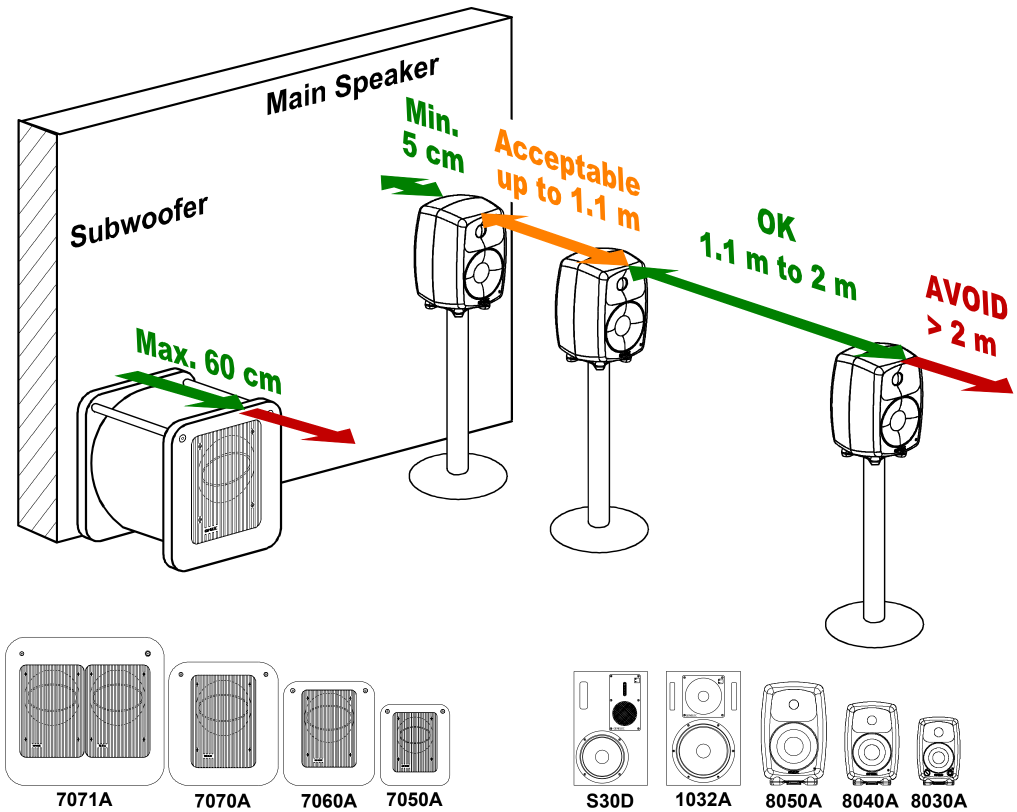


**Figure 5 - Distances from a single wall to the front baffle of the loudspeakers combined with subwoofer(s). Correct (green), acceptable (orange) and avoid (red).**

## 3 Other Reflections

The discussion above concentrated on one reflection only, i.e. the reflection from a single wall behind the loudspeaker. Similar reflections also occur in other directions (side walls, floor or ceiling, console top, equipment racks etc.) and their frequencies again depend on the distances to the relevant boundaries. In general, loudspeakers should be placed as far as possible from reflecting surfaces. This will place the reflection-induced irregularities to as low a frequency as possible, which is also beneficial to the imaging. Although not always practical, one suggestion is that if there are many reflecting surfaces (tables, screens, etc…), loudspeakers could be placed slightly higher than the listening level, and then tilted down towards this listening position.

The critical midrange response of the centre channel is often seriously compromised by strong reflections. Placing this centre channel loudspeaker below the video screen means the loudspeaker cabinet is closer to the floor and the floor reflections will colour strongly the midrange response. So, the centre channel loudspeaker should be placed above the video screen or TV, ensuring that it does not suffer from first order ceiling reflections. Where the ceiling is low and hence might generate strong reflections, some damping material can be applied to absorb these first reflections

It is well known, for example, that the first reflection created by the surface of a large mixing desk induces 3…6 dB ripples starting at about 1kHz – 2kHz and extending up to higher frequencies. The subjective effect of this comb filtering is that the loudspeaker does not sound so ‘open’. Also, the vibration coupling between loudspeaker and console will especially affect small near-field loudspeakers placed on the meter bridge of a console. This could lead to a compromised frequency response particularly in the midrange area. To solve this common problem, Genelec developed a special support called Iso-Pod™ (Isolation Positioner/Decoupler™) for its 8000 series monitors. The rubber based Iso-Pod™ stand provides vibration decoupling between loudspeaker and console meter bridge. It is also designed so that the loudspeaker can be tilted (+/-15 degrees) enabling the acoustical axis to be pointed precisely towards the listening position.

A loading-induced compromise present in the frequency response is a 'bump' in the range 100…250 Hz when a near-field monitor is placed close to a large surface or typically a mixing console (1...3 $m^2$ ). To compensate for this 'bump', the larger 8000 Series monitors offer a 'Desktop Low Frequency' control that provides a reduction of 4 dB @ 160 Hz, so that a flat response may be retrieved in that region as well. A statistic study[1] has shown that this control is required in about 60% of cases where a loudspeaker is mounted in such a way.

## 4 Placement of Equipment in the Room

As the positioning of tables, screens, racks, etc, is critical for stereo installations, it is clear that this position issue must be considered even more carefully when planning, designing and installing a multichannel audio control room.

Early reflections, with high amplitude in relation to the direct sound, can smear the coherence of the sound image and compromise the localisation of the sources in space. To avoid this, all reflecting surfaces between the loudspeakers and the listening position should be minimised. Of course, some equipment are inevitable, however symmetrical positioning of the equipment is essential. Even with symmetry, reflections will remain and everything possible should be done to remove the reflecting surfaces from the vicinity of the acoustic path – see Figure 6. It should also be remembered that the smaller physically the loudspeaker, the less directional it is, and the more influenced it will be by its surroundings.

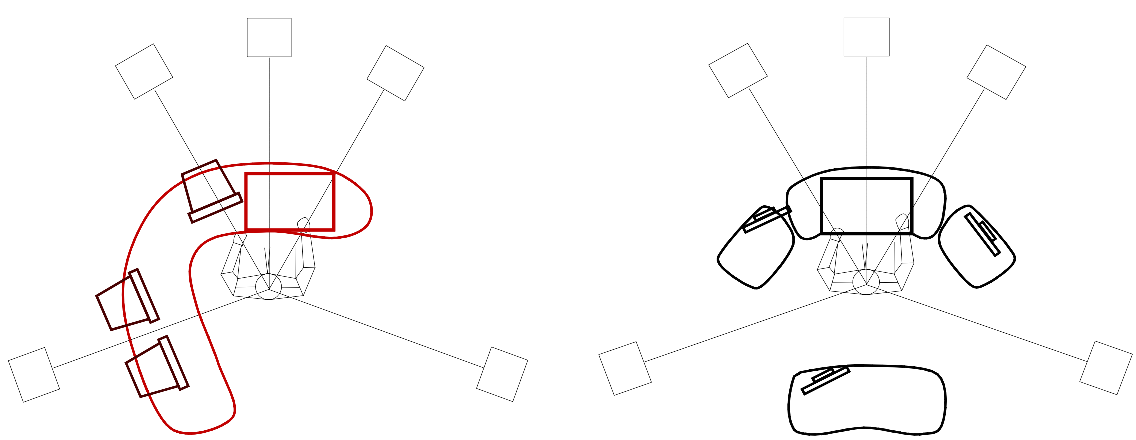

**Figure 6**

**Left: Example of non-symmetrical layout: reflections from computer screens and table surface are totally different. This will create a front-back and left-right localisation smearing.**

**Right: Example of a symmetrical layout: reflection surfaces are minimized, away from the listening position and clearing the acoustical path when possible.**

[1] "*Compensating the Acoustical Loading of Small Loudspeakers mounted near Desktops*", A. Goldberg, A. Mäkivirta, A. Varla, Genelec Oy, Iisalmi, Finland, presented at the 117th AES Convention, San Francisco, U.S.A., October 2004.

# 5 The High Frequency Localization Cues

In a multichannel environment, high frequency information is of utmost importance for the listener to be able to detect subtle movements and variations in the virtual space. Natural sounds around us are not band limited and we feel very comfortable in real space. If there is some bandwidth limitation and loss of HF in the monitoring system, the multichannel virtual space will not be as well defined as the real one we live in and consequently know best. Dynamic panning creates movement that we need to be able to perceive exactly. If the reflections in the room are too high compared to the direct sound, these effects will be smeared and undefined.

As we know from current practice, the typical multichannel room is often not originally designed as a multichannel room. In addition, as standard practices are still not well defined, the installations vary from one room to the next. Clearly, loudspeakers used for multichannel work should have a well controlled directivity, which leads to a high direct-to-reflected sound level ratio and a reduced interaction with the nearby boundaries. The listeners will be located in the more direct field and will therefore be able to listen to more of the program material and less to the room interaction. The purpose of the Genelec Directivity Control Waveguide (DCW™) is to control the radiation angle of the tweeter and midrange elements such that the detrimental effect of diffractions from the loudspeaker cabinet and the nearby walls and structures is minimized. By increasing the direct-to-reverberant sound ratio, the localization and the uniform quality of the sound is improved, irrespective from the loudspeaker location.

# 6 Flush Mounting Loudspeakers

Flush mounting has been mentioned earlier and this mainly concerns larger loudspeakers. However, even small loudspeakers can be flush mounted with all the same benefits as for the larger ones.

The benefits are multiple:

- Improved low frequency efficiency
- Rear wall cancellation is eliminated
- Cabinet edge diffraction is avoided

It should be remembered that the radiation load changes once the loudspeakers are flush mounted and that adjustment of the loudspeaker's frequency response is required to keep a flat response below 200 Hz.

When flush mounting small active loudspeakers, the amplifier ventilation has to be arranged from the rear. This is best done, either totally from the rear of the wall where the loudspeakers are mounted, or by leaving suitable ducts to the listening space. However, the space between the loudspeaker cabinet edge and the soffit should never be left open without a facing panel. If there is no facing panel, the cavity might form a resonator, which might have adverse effects on the frequency response and perceived sound quality – see Figure 7 for some details.

In order to enhance the low frequency reproduction the soffit wall is made hard. In two-channel work, this is ideal but in multichannel work, the front wall might form a large reflective surface off which the rear channel direct sound will bounce. In some cases, the large loudspeakers themselves offer a reflective surface for mid and high frequencies. Depending on the size of the room these reflections may cause image instability. It is therefore recommended that these wall surfaces should be covered with an absorbent, which is efficient enough at MF/HF but does not absorb LF. A 50 mm sheet of rock wool or similar is suitable for the job - see Figure 7 lower right drawing.

In an ideal case, all five loudspeakers should be flush mounted in a similar way to achieve the best possible results. With all loudspeakers on the same radius, this will lead to new room shapes. Various other issues in room design such as geometry of the control room, ceiling design, sound diffusion, positioning and space for low frequency absorbers, midrange definition, clarity, etc… have to be addressed separately by studio designers.

The very low frequency damping is an issue of its own and does not really vary from standard stereo room design, except that there should be sufficient absorption areas on the sidewalls as well. The typical rear wall absorbers are not sufficient to damp modes between sidewalls, which are excited by the rear loudspeakers.

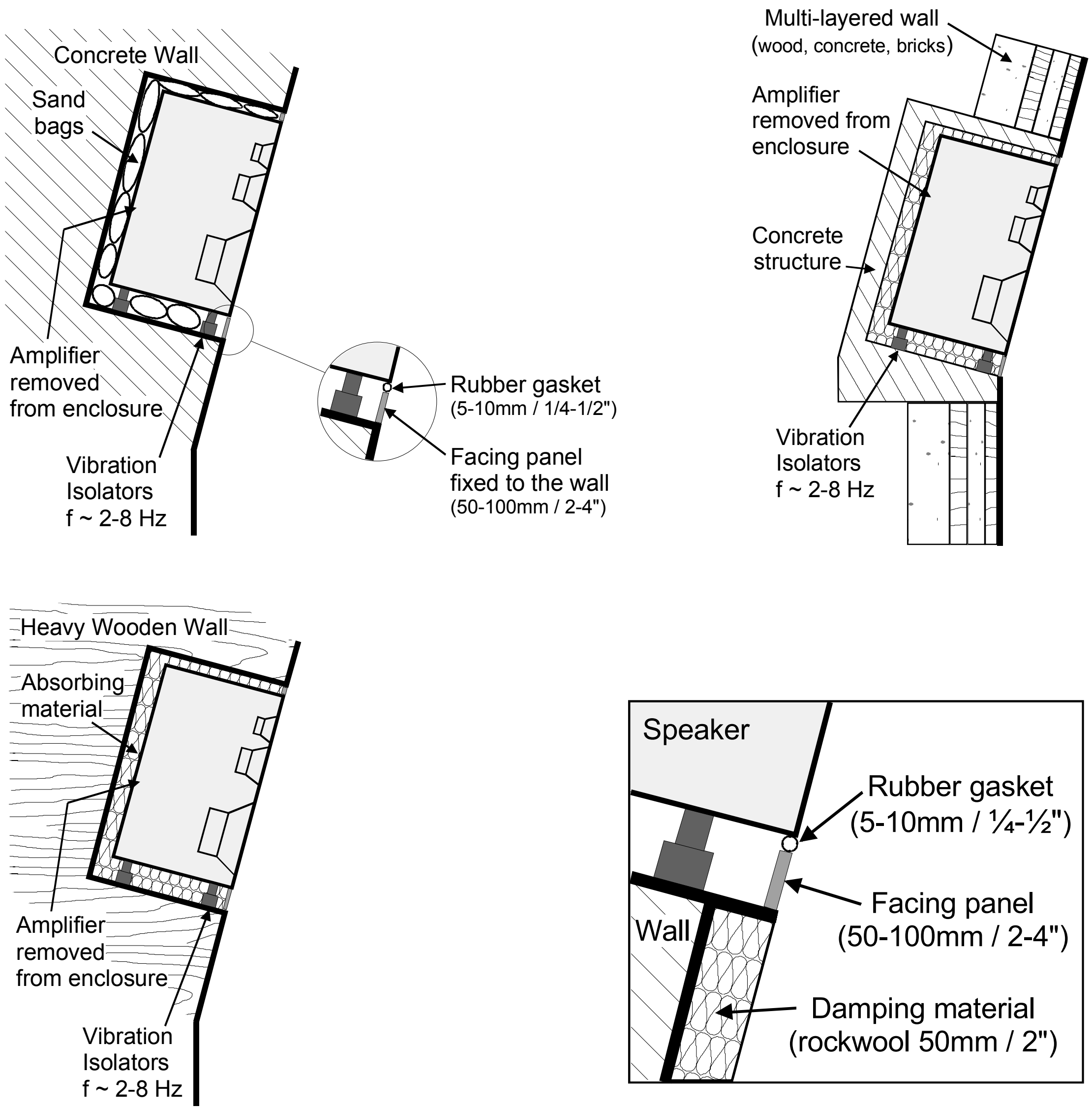


**Figure 7 - Flush Mounting Recommendations: various mounting techniques and materials are illustrated. The facing panel mounting is also essential for smaller loudspeakers. - Lower right drawing: Flush mounting suggestion for multichannel control room design including absorption on the soffit surface.**

# B MULTICHANNEL FORMATS

As the number of audio channel grows in the consumer and professional market, new multichannel formats regularly appear. However, and for obvious reasons, most of the previously accepted multichannel formats continue to be used. This chapter provides a brief overview of the encoding and decoding technologies available today and introduces new developments and ideas.

## 1 Surround Sound Formats

There are two very distinct surround sound technologies; one type is called "discrete" system (5.1 and others), and the other is termed "non-discrete" or "matrix" system.

### 1.1 Non-Discrete Formats

The non-discrete systems are often called "Matrix 4:2:4" formats since the encoder receives two input channels and delivers 3, 4 or more output channels after processing. During this process, it is essential to use the proprietary encoder/decoder to monitor exactly what will be heard by the end listener in each of the extracted channels.

- **Dolby®* Surround**
  Dolby Surround is a matrix encoding process that in essence "folds" Left, Center, Right, and Surround channels onto a stereo soundtrack - Dolby Stereo (Lt/Rt). The ***passive*** Dolby Surround decoder extracts three channels (**L**eft, **R**ight & Mono **S**urround) from two channels Dolby Stereo (Lt/Rt). The left and right channels are full range and the mono surround channel is band limited between 100 Hz and 7 kHz. In practice the surround channel is only used for sound effects. Dolby Stereo encoded soundtracks can be played back in either, mono, stereo or surround format.

- **Dolby Pro Logic**
  Dolby Pro Logic is an ***active*** matrix decoder that decodes the four channels of surround sound that have been encoded onto stereo soundtracks (Lt/Rt). From the "folded" Lt/Rt soundtrack, the Pro Logic decoder "unfolds" four channels (L, C, R, S) on playback (without a Pro Logic decoder, the encoded program plays in regular stereo). As most consumer systems do not have a full range centre loudspeaker, the Pro Logic decoder provides a bass management system that will redirect the centre channel low frequencies below 100 Hz to the left and right loudspeakers. This might avoid overloading of the typical small consumer centre loudspeaker.

- **Dolby Pro Logic II**
  Dolby Pro Logic II is an advanced ***active*** matrix decoder that derives five different channels surround (Left, Center, Right, Left Surround, and Right Surround) from any stereo program material, whether or not it has been specifically Dolby Surround encoded. On encoded material such as movie soundtracks, the sound is similar to a Dolby Digital 5.1 reproduction, while on unencoded stereo material such as music CDs the effect is a wider, more involving soundfield. Note that Pro Logic II provides two full-range surround channels, as opposed to Pro Logic's single, limited-bandwidth surround channel.

- **Dolby Pro Logic IIx**
  The Dolby Pro Logic IIx format expands the five-channel Pro Logic II listening experience to 7.1 channels with the introduction of two additional surround channels. Dolby Pro Logic IIx works with any stereo, Dolby Surround, Dolby Pro Logic II, or Dolby Digital encoded material.

- **DTS® NEO:6™***
  DTS NEO:6 is an **active** matrix system that delivers up to six channels (L, C, R, Ls, Cs, Rs) of audio from any two-channel source, analogue or digital. This format allows anyone with a DTS NEO:6 decoder to simulate a surround sound experience, off of any software material.

- **Logic 7 Matrix**
  Dr David Griesinger and Lexicon®* have developed this decoding system. It can be applied to car audio, consumer decoder, etc. One feature of this matrix decoder is that it can extract 7.1 channels from conventional stereo program material. Two additional side loudspeakers are required to replay this matrixed format.

- **Meridian©* TriField**
  Meridian TriField surround processing accept any conventional two-channel sources, and separate them into multiple channels. The extracted surround components of the original recording are delivered to the rear surround speakers, and side surround speakers, if present. The TriField processing technique allows the listener to adjust the width of the image. Meridian recommends to use this processing option for solo, chamber, or vocal music.

In general, various audio quality problems exist with matrix encoded surround systems. They suffer from a lack of channel separation, in some formats certain channels have a limited frequency response and they are prone to centre channel collapse. So to improve the surround sound experience another group of surround systems were developed: the discrete systems.

## 1.2 Discrete 5.1 Formats

These formats are commonly called discrete systems as all the main channels (**L**eft, **C**entre, **R**ight, **S**urround **L**eft & **S**urround **R**ight) are full bandwidth (20 Hz – 20 kHz) but there is also an **LFE** channel that is band limited. Note that the rear surround information is now in ‘stereo’ as there are two different channels. Here are some of the current 5.1 formats:

- **Dolby Digital**
  The Dolby Digital format is a discrete system that keeps the multiple channels fully separated throughout the encoding and decoding processes. The six original discrete channels are digitally encoded into a single bit-stream using the Dolby AC-3 (Audio Coding 3) encoding system. The codec can encode mono, stereo and 5.1 signals.

- **DTS**
  The DTS system encodes and decodes six discrete channels that are encoded into a DTS proprietary bit-stream using the DTS Coherent Acoustics codec. The Apt-X100 is the codec for movie theatres. This is a perceptually optimised differential sub-band coder.

- **Meridian Loss-less Packing (MLP)**
  This is the official DVD Audio encoding format. MLP takes a PCM data stream, ’packs’ it (PPCM) at one end of the chain and ’unpacks’ it at the other to provide a completely accurate replica of the original. The difference between Dolby Digital & DTS and MLP is that MLP is loss-less, i.e. the original signals are encoded without compression losses.

- **DSD (Direct Stream Digital)**
  The DSD data format is the format used by Sony®* & Philips®* for the SACD. On the SACD, space is reserved for two channel stereo recordings, multichannel recordings and separate

area for additional text and graphics information. The DSD data format is based on a 1-bit sigma-delta modulation and operates with a sampling frequency of 2.82 MHz with a bandwidth extending to 100 kHz. The DSD data format cannot be recorded on conventional PCM digital multi-tracks. Special multi-track recorders that can record DSD data format are necessary.

- **MPEG-2 BC** (ISO/IEC 13818-3)
  This MPEG 2 format is backward compatible to MPEG-1, enabling the coding of five channels plus an LFE channel. From the MPEG 2 bit-stream, two of the encoded signals are an optimised stereo down mix of all input signals that will be coded as an MPEG-1 signal. These signals can be the outputs of a simple decoder that only provides a stereo signal.

- **AAC** (MPEG-2/MPEG-4) (ISO/IEC 13818-7)
  The Motion Pictures Experts Group (MPEG), which includes Dolby, Fraunhofer Institute, AT&T®*, Sony, and Nokia®*, developed the Advanced Audio Coding (AAC) format to provide high-quality audio that compresses more efficiently than earlier formats. This codec provides encoding for up to 48 full range channels, up to 96 kHz sampling rate, with 16 low frequency effects channels, 16 overdub channels (language channels) and 16 data streams. Recent test at the IRT have shown that one third of the bit-rate is required for a similar sound quality to the Dolby Digital and DTS formats, making it ideal for streaming audio over the internet.

- **Dolby AAC**
  Dolby® AAC is an enhanced version of the core MPEG-AAC (MPEG-2 and MPEG-4 AAC) that adds proprietary Dolby intellectual property and supports data rates from 14 kbps (mono) to 128 kbps (stereo) and up, enabling wider frequency range at lower bit rates.

- **Dolby Digital Plus**
  Dolby Digital Plus expands the capability of Dolby Digital technology for future next-generation video coding systems, such as H.264, and extends efficiencies in transmission to allow broadcasters to deliver increased capability and capacity. Dolby Digital Plus is designed to meet the four major qualifications of a next-generation broadcast audio codec: backward compatibility, spectral efficiency, cost savings, as well as compatibility with future formats.

## 1.3 More Multichannel Formats

Motivated by the movie industry as well as the automitive and the video and computer gaming industries, the companies developing multichannel formats have already gone beyond six channels of encoding and replay. Most consumers are already familiar with 6.1 systems but there are many more ideas in development and applications available today.

- **Dolby E**
  This Dolby format encodes eight digital full bandwidth channels into two channels, with a digital audio frame identical to the digital video frame for ease of editing. A 5.1 and stereo mixes can be sent down a single AES-EBU cable with up to ten encode-decode cycles possible before audible degradation. This professional format is not intended for consumer use.

- **Dolby Digital Surround EX™**
  Dolby Digital Surround EX™ provides a third surround channel on Dolby Digital movie soundtracks. The third surround channel can be decoded at the cinema's or home viewer's

option for playback over surround loudspeaker(s) located behind the seating area, while the left and right surround channels are reproduced by surround loudspeakers to the sides. To maintain compatibility, the centre back surround channel is matrix-encoded onto the left and right surround channels of an otherwise conventional 5.1 mix, so no information is lost when the film is played in conventional 5.1.

- **DTS-ES™**
  First, the DTS-ES™ Discrete format is a 6.1 format that uses a 5.1 DTS data stream with additional extension bits that allows the decoder to retrieve an additional discrete Rear Centre channel. The DTS-ES™ Matrix version 'folds' and 'unfolds' the rear centre channel from the matrix encoded surround left and right channels.

- **SDDS (Sony Dynamic Digital Sound)**
  The SDDS encoded data is recorded on either side of the 35mm film and is primarily a 7.1 format that can also be used as 5.1. The extra two front channels are used as Centre Right and Centre Left channels to give a better front stage image for the very wide screens found in large Movie Theatres. There are still two surround channels and a dedicated LFE channel so the format remains compatible with 5.1. It is used mainly in Movie Theatres.

- **Logic 7**
  The Logic 7 decoding matrix can have a stereo input signal or a discrete 5.1 input signal. This matrix will conserve the front L, C, R channels but also extracts four different surround channels; two side channels and two surround channels. Two LFE channels are also extracted for 'better LF envelopment' and replayed over two subwoofers.

- **TMH Sound System**
  TMH Labs has been developing and promoting a multichannel system capable of delivering a minimum of 14 channels for the home and 15 channels for the theatre. Currently the delivery methods includes disc or a link to a proprietary computer server that will download the audio content including metadata.

- **Ambisonic System**
  The Ambisonic surround sound system encodes sound directions and amplitudes and then reproduces them over practical loudspeaker systems in such a way as to fool the ears of listeners into thinking that they are hearing the original sounds correctly located. This can take place over a 360 degree horizontal only soundstage (pantophonic systems) or over the full sphere (periphonic systems). Systems using the so-called 'B' format signals to carry the recorded information require three and four channels respectively for full encoding of sounds to the kind of accuracy achievable with first order microphones (cardioid, figure eight, etc). Practical reproduction requires a minimum of four (pantophonic) or eight (periphonic) loudspeakers. The important thing to note is that there is no need to consider the actual details of the reproduction system when doing the original recording or synthesis, since if the 'B' format specifications are followed and suitable loudspeaker/decoder setups are used, the ambisonics system will work. In all other respects the two parts of the system, encoding and decoding, are completely separate.

- **VBAP**
  Vector Base Amplitude Panning[2] (VBAP) is a method for positioning virtual sources via multiple loudspeakers. The number of loudspeakers can be varying and they can be placed in an arbitrary 2-D or 3-D positioning. Since it uses at one time the minimum number of

[2] The VBAP method has been developed by Dr. Ville Pulkki at Helsinki University of Technology

loudspeakers needed (one, two or three), VBAP produces virtual sources that are as sharp as is possible with current loudspeaker configuration and amplitude panning methods.

This list is not exhaustive as there are other multichannel surround formats, like IMAX for example. Many more multichannel formats will be developed in the future since the interest for multichannel audio and its wide-scale application is growing all the time.

# 2 Multichannel Audio Applications

The interest for multichannel audio is very wide so here is a short list of current application areas:

## 2.1 Broadcasting Applications

- Digital TV broadcasting in either high or single definition TV format. The DVB transmission will occur via either terrestrial or satellite network
- TV and radio commercials and jingles
- Internet: radio & TV specific productions for web diffusion (news, sports, interactive programs)
- Radio: digital audio broadcasting, (DAB) via either terrestrial or satellite network (AOL incorporated Dolby AAC in its proprietary Ultravox streaming media platform)
- The Digital Radio Mondiale system (the digital replacement for radio broadcasting under 30 MHz) has been developped on the MPEG-4 AAC audio coding technology

## 2.2 Other Applications

- Music productions for DVD-Audio & SACD discs
- CD-Rom, corporate presentations, product launches
- Movie Theaters, E-Cinema
- DVD-V (with encoded audio in Dolby Digital / Dolby EX, DTS / DTS-ES, etc...)
- Video Games (PlayStation®* 2, GameCube™*, Xbox™*, PC games)
- 3G Wireless terminals: they will be using AAC encoding system (computers, mobile phones, iPod, iTunes 4)
- Computer based multimedia, industrial and consumner animations and simulations (QuickTime 6 supports AAC format)
- High-definition DVD: to run with HD pictures display and 3 GHz PC
- MP3 Surround: compact spatial side information is embedded into the basic stereo MP3 bitstream in a compatible way (Fraunhofer institute, july 2004)
- Archiving, audio library, disk-based storage and more...
- ...

# C LOUDSPEAKER SET-UP FOR MULTICHANNEL

So far, we have discussed the various phenomena affecting the reproduction of a loudspeaker in a room and had a brief overview of the multichannel formats of today. We will now discuss specific questions about multichannel installation for 5.1 and beyond.

Conventional monitoring rooms intended for stereo reproduction are not very suitable for surround sound use. Likewise, a perfect five channels production is more likely to sound dull and uninteresting once replayed in stereo only.

## 1 Important International Recommendations

Various technical committees have issued papers to try to standardise the specifications for surround sound reproduction. The AES Technical Committee is working in collaboration with the EBU and the SMPTE organisation to produce consensus recommendation and standards. The Multichannel Audio Transmission Group (MCAT), which has members from the EBU, Dolby and DTS amongst others, is establishing standard procedures for multichannel audio transmission between European countries. The German Surround Sound Forum working group is working in the same direction. Another important organisation is the ITU that issues many recommendations for various application areas and two of their recommendations are of particular interest for surround sound reproduction:

- **ITU–R BS.775–1**
  ”*Multichannel stereophonic sound system with and without accompanying picture*” (Geneva, 1992-94)

- **ITU–R BS 1116–1**
  ”*Methods for the subjective assessment of small impairments in audio systems including multichannel sound systems*” (Geneva, 1994-97)

The ITU is also working together with the IEC, SMPTE and ISO/MPEG organisations. Once again, more standards and recommendations are issued on a regular basis and this list is not complete. The following chapters highlight the most important points of these ITU recommendations and discuss some other essential points of multichannel reproduction.

## 2 Placement of Main Loudspeakers for 5.1

### 2.1 Horizontal Loudspeaker Placement

Much research work was been done over the past 50 years to determine the best angle for two loudspeakers reproducing stereo material. As we know, 60 degrees has long been considered to give the optimum stereo width vs. mono phantom image. With multichannel audio, we have a standard recommendation (ITU-R BS 775-1) for the positioning of the five loudspeakers in space (see Figure 8 below).

It has to be noted that this recommendation is subject to discussion by some producers, engineers, etc. and that they use alternative angles for positioning the surround channels.

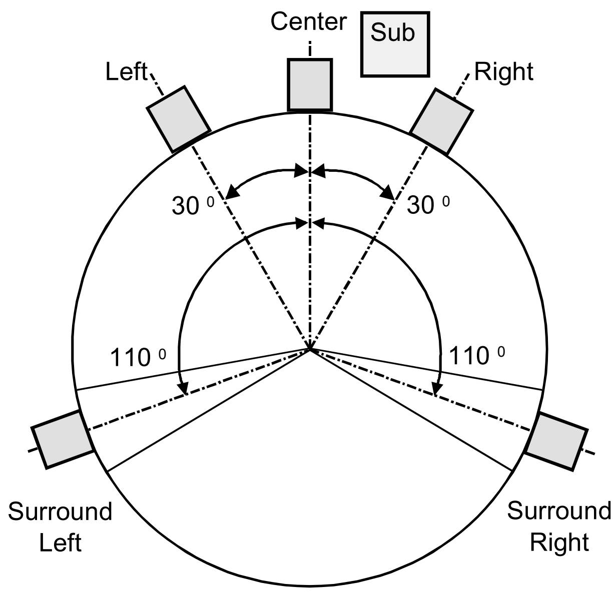


**Figure 8 - Loudspeaker Positioning for 5.1 Multichannel Audio (ITU-R BS 775.1)**

The ITU recommendation specifies quite clearly a number of points. In the front array, the Left & Right loudspeakers are spread 60 to 70 degrees apart, with the Centre loudspeaker in the middle. Ideally, all loudspeakers should be of the same type to achieve coherence in the sound field. In any case, the three frontal L-C-R loudspeakers really have to be of the same type so that no coloration changes occur when panning sounds across the front stage. This part of the practice seems to be clear and most parties agree with it.

Concerning the surround loudspeakers, the ITU recommends a positional window between ±100 degrees and ±120 degrees from the centre line. This is applied where there are two loudspeakers to reproduce the two surround channels. If more than two loudspeakers are used (such as four), an equal number of loudspeakers have to be placed symmetrically on either side of the centre line on a circle between ±60 degrees and ±150 degrees. Despite the fact that most recording engineers choose the ±110 degrees position, there are different views and opinions about these angles. Depending on the source material and the type of surround effects desired, the choice for the positioning of the surround loudspeakers can differ.

It is also very important that there is symmetry in the multichannel installation in relation to the room boundaries. All reflections created by the various nearby boundaries should be identical from left and right in the time domain, so that the spatial information and the panning of the sources stay as stable as possible. Furthermore, we also recommend that the listening position is located in the front half of the room, so that the engineer listens to the maximum of direct-to-reverberant energy. The further away from the direct energy the engineer is, the more reflected energy from the rest of the room he/she will receive.

Once the array is in place, the loudspeakers must be pointed towards the engineer’s listening position to obtain the optimal on-axis reproduction. Since all channels are fully discrete, equal care must be taken for each channel when doing the final multichannel mix down and hence the orientation of the loudspeakers should be very precise. Some installers use lasers to ensure that they have positioned the loudspeakers correctly!

Finally, if a projection screen is required in such a multichannel set-up, the screen type should be acoustically transparent.

## 2.2 Arrival Time

It is firmly recommended that the multichannel set-up places loudspeakers at equal distance from the listening position and is symmetrical relative to room boundaries. In other words, all sources of sound have to have the same 'arrival time' at listening position. For arrival time to be identical, the sources have to be positioned at an equal distance from that listening position, on a circle radius. This places constraints on the room design but it is essential to respect this criterion. As an example, if an audio element is panned from Centre to Left, and the Centre loudspeaker is offset forwards or backwards by 25 mm (1"), one can expect amplitude ripples in the 500 Hz region due to signal arrival delays.

## 2.3 Vertical Loudspeaker Placement and Height

The vertical positioning of the loudspeakers is also important, but less critical that the horizontal. The brain has very high capability to localise information in the horizontal plane (azimuth angle), but it is not so good in the vertical plane (zenith angle). The physical position of our ears explains that fact very clearly.

The ideal positioning of the three front loudspeakers is where the three acoustical axes (as defined in the data sheet for each product) are positioned at exactly the same height. The ITU-R BS 775-1 recommends that all loudspeakers should be ideally placed at the same height.

However, as mentioned above the resolution of the ear/brain in the vertical plane is about 3 degrees above ear level horizon and 3 to 10 degrees below ear level horizon. Precisely because of this behaviour of the ear/brain, our vertical localisation tolerance is about 7 degrees. Within these 7 degrees, there is a localisation blur (inaccuracy in positioning a sound source) in the vertical plane. This allows two sources to be positioned at slightly different heights without the brain noticing the height difference. This useful human hearing limitation can be exploited when planning the centre channel position, for example. Some tolerance in loudspeaker height can be allowed, and this without disturbing the engineer who is doing multichannel work and panning sound sources across the L-C-R loudspeaker array – see Figure 9.

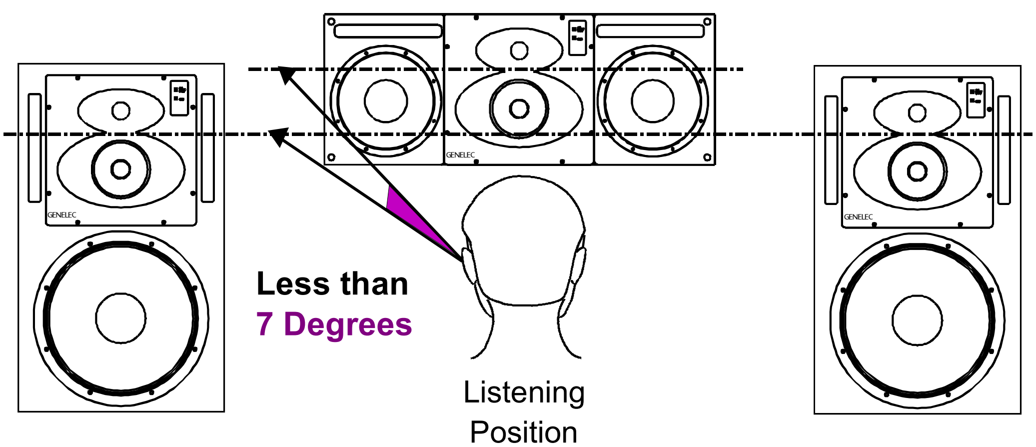


**Figure 9 - Ear/Brain Vertical Localisation Tolerance Applied to Loudspeaker Positioning**

The ITU is quite strict on the identical height positioning for the three front loudspeakers, however it gives some allowances for the surround loudspeakers. They can be placed higher than the front loudspeakers, and tilted down towards the listening position up to an angle of 15 degrees.

Other recommendations, such as the EBU Tech 3276 and the Japanese HDTV Multichannel Sound Forum, specify that all channels have to be placed at the same height, including the surround channels. They can then be all tilted down towards the listening position if required. The Japanese HDTV Multichannel Sound Forum specifies that, "…All loudspeaker axis should be placed at the same height between 1.2…2.0 m, more than 1.2 m is recommended…"

Concerning the overall height of the surround system, the ITU states that, "…The frontal loudspeakers should ideally be placed at a height approximately equal to that of the listener's ears…" This height is specified as 1.2 m from the floor.

This is an important guideline but acoustically when using large loudspeakers the interaction with the floor (below 400 Hz) and the first reflection cancellation (typically between 100 Hz…200 Hz) can be very serious if the loudspeakers are placed too low and close to the floor surface. As seen in stereo control rooms, the overall height of the loudspeakers is dependent on the room geometry and the listening distance. The larger the loudspeakers the further away from the floor they should be placed to reduce cancellation phenomena. Furthermore, in standard stereo control room construction, studio designers always observe the accepted rule that the main loudspeakers should not be placed higher than 15 to 20 degrees from the ear level horizon. For our brain, if this loudspeaker vertical angle is higher than the above recommended values, the auditive and visual coherence becomes inaccurate in our perception.

As in stereo rooms, the relative loudspeaker height in control rooms is dependent upon:

- Loudspeaker size and front baffle surface area
- Listening distance
- Control room geometry as well as design philosophy and layout options
- Equipment and reflections surfaces in the room

# 3 Placement of the Subwoofer in the Room

Psychoacoustics research work has shown over the years that the very lowest frequencies have minimally audible stereophonic effect and hence may be reproduced by a single source. Important work on that matter was reported in two AES papers[3].

However, some recent research (Dr. David Griesinger[4]) suggests that for best results in terms of psycho-acoustical perception (spaciousness and envelopment), there should be two subwoofers placed on either side of the room and driven with the same signal but with a 90 degree phase difference between them.

One misleading 'recommended' place for the subwoofer is in the front, on the floor and in the middle of the room, equidistant from the sidewalls. This position can, in fact, be a serious acoustical compromise, since the subwoofer sits in the first pressure minimum of the lateral standing wave. Hence the frequency response in that location will most likely display serious irregularities.

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[3] "*Perceptibility of Direction and Time Delay Errors in Subwoofer Reproduction*", Juhani Borenius, AES Preprint 2290; and "Loudspeaker Reproduction: Study on the Subwoofer Concept", Christoph Kügler and Günther Theile, AES Preprint 3335.

[4] "*Speaker Placement, Externalisation, and Envelopment in Home Listening Rooms*", D. Griesinger, AES September 1998.

Recommended positions for subwoofers are (subwoofer always on the floor in these cases):

- Close enough to the front wall and slightly offset from the middle of the room to avoid the first pressure minima position.

- In a corner, close to both front and sidewalls. This position will maximise the efficiency due to the corner loading.

Also, it has to be remembered that:

- Adjustment of the gain (input sensitivity) and frequency response (Bass Roll-off) of the subwoofer is necessary since the acoustical loading has changed relative to calibrated anechoic conditions.

- The subwoofer can also be flush mounted in the front wall but the discussion of the position of the source relative to the room remains valid.

- The phase adjustment on the subwoofer at the crossover frequency is important to achieve and keep a flat frequency response in the crossover region.

Sometimes the rear and/or side loudspeakers also need LF extension. There are two possibilities:

- The bass management system can also include these rear/side channels and their LF content will be reproduced through the subwoofer(s) as well. In this case, it is highly important that all 'satellite' loudspeakers are placed on the circle radius so that the arrival time difference between each loudspeaker and the subwoofer is the same. It must be also noted that the acoustical behaviour of the room at low frequency will greatly affect the way the low frequencies are summed and replayed by one single subwoofer. The LF acoustical characteristic of the room should be as identical as possible in the front and rear part of the room.

- A separate subwoofer is used for the rear loudspeakers. The location of this subwoofer should be in the rear part of the room and positioned in a similar fashion to the subwoofer for the front channels. If the room has very different LF acoustical characteristics between the front and the back part, the option of dual subwoofers might provide much better monitoring results than the single subwoofer choice.

In general, regardless of the obvious benefits of low frequency extension, the use of the subwoofer(s) and the LFE channel is currently causing many problems. These should be overcome with proper bass management system and better defined standard practices of the use and content of the LFE channel. As mentioned several times already, we are in an evolving phase of multichannel reproduction and more research and experiments are needed to find the best ways to utilise these available channels.

# 4 Small or Large Monitoring Systems

## 4.1 Placement of Small Loudspeakers for Multichannel Audio

The positioning of the listening point at the centre of an imaginary circle is as important as the positioning angles. It is strongly recommended, once again, that all loudspeakers be of the same type and brand to guarantee similar and coherent responses of the multichannel monitoring system. In a small room, small multichannel systems are suggested and it is a good idea to locate the loudspeakers as close as possible to the walls or boundaries whilst respecting the identical radius from the listening position.

Once the loudspeakers are against a boundary the large baffle created by the walls provides a hemispherical radiation load for the omni-directional low frequencies (below 200 Hz) and hence a level gain of +6 dB will be seen below 200 Hz. Consequently, the bass response of the loudspeakers has to be adjusted so that the frequency response remains flat. The alternative is to reduce the radius of the loudspeakers but this can lead to problems with low frequency cancellation dips from the wall behind the loudspeaker. Optimisation of this was discussed in Chapter A2.

Also, it is important to position small loudspeakers on separate stands as far as possible from any reflective surfaces. Although it is common practice, loudspeakers should not be placed on the meter bridge of a console. Conventional two-way loudspeakers should be placed vertically, not horizontally, unless the design is coaxial. If placed horizontally, lobbing of the polar pattern will destroy imaging when the engineer moves slightly off-axis. In addition, a phase cancellation may be seen at the crossover frequency. For the same reason, two-way centre channel loudspeakers should also be placed vertically.

In three-way loudspeakers the MF/HF section should always be orientated vertically, as shown previously in Figure 9, but the bass drivers can be positioned above, below, to the left or to the right of the DCW™ depending on the room constraints and design. It should be emphasised again that the polar pattern of all front loudspeakers should be identical to achieve stable imaging. In two-way designs, this means that the driver configuration should be identical.

So, summarising there are two recommendations:

- In a small room, the loudspeakers should be placed near walls to avoid cancellations.

- In a large room, the loudspeakers should be placed far away from walls as this gives a better direct-to-reverberant energy ratio, but be careful of cancellations.

## 4.2 Placement of Large Loudspeakers for Multichannel Audio

Large loudspeakers should really be flush mounted to achieve the best possible performance. Detailed recommendations for flush mounting are given in Chapter A6. If this is not possible and the large loudspeakers have to be free standing, they should then be placed against a solid and structurally heavy boundary to avoid low frequency cancellations and the frequency response adjusted accordingly – see Chapter A1. For very large loudspeakers this is not possible, as the depth of the loudspeaker is large enough to put the loudspeaker at a distance from the wall that will cause a cancellation in the bass region. Again, note that the Japanese HDTV Multichannel Sound Forum also specifies that: "*Flush mounting is desirable to avoid reflections from rear walls*".

## 4.3 Loudspeaker Models for Different Applications

The loudspeaker models that are most appropriate for a particular installation depend on a few factors:

- Room size
- Listening distance

- Listening level
- Program spectrum content

By following the three simple steps below and using the selection table (see Table 2), you will be able to select the appropriate loudspeaker models and subwoofer type for every application:

- Calculate the room volume and find the highest row in the table that is not smaller than the room volume.

- Measure the listening distance to the centre of the listening area and find the highest row in the table that is not shorter than the listening distance.

- If there are two different rows selected in the previous two steps, select the models from the row that is lowest in the table, i.e. the larger system of the two if there are two different lines recommended.

Note that these recommendations are for the minimum sized system that can be expected to give a suitable SPL for most professional audio applications. Larger systems offer higher SPL and an increased directivity control, so do not be afraid to select larger models in the range than those indicated. The main point to be concerned about when up rating the system is to keep the whole system in balance, so do not select very large loudspeakers (**1036A's**) for the front channels and small ones for the rear channels (**8030A**'s) together with a small subwoofer (**7070A**) for the LFE channel and the bass management! Additional separate subwoofers can be used on the rear channels too, although it is not detailed here. If space or finances are limited, the rear channel models can be compromised slightly by selecting the next sized model down in the Genelec range. For example, use **8040A**'s instead of **8050A**'s.

**Examples**
1. If the room is 4 m (13') wide, 7 m (23') long and 3 m (10') high, then the room volume is 84 $m^3$ (2990 $ft^3$). This limits the loudspeaker selection to 8040A / 1030A or larger. If the listening distance is then measured to be say 1.9 m (6' 3") then the selected front loudspeakers are confirmed as being 8040A / 1030A or larger.

2. If the room is 6 m (19 1/2') wide, 14 m (46') long and 2.5 m (8') high, then the room volume is 210 $m^3$ (7200 $ft^3$). This limits the loudspeaker selection to 1034B or larger. If the listening distance is then measured to be say 5 m (16'5") then the selected front loudspeakers should then be 1035B or larger as the 1034B should only be used up to 4.5 m (14'9").

**Multiple Subwoofers**
When two or more subwoofers (except the **7050A** as it does not have a sum output) are positioned close to one another mutual coupling is the fortunate by-product. This is due to the long wavelengths, associated with low frequencies, causing constructive superposition. For mutual coupling, the subwoofers must be placed within 1/2 a wavelength of one another (85 Hz upper crossover frequency 1/2 wavelength is approximately 2 m). For example, two subwoofers give a 6dB increase in acoustical output at the listening position - see Table 3.

| Maximum Room Volume, $m^3$ ($ft^3$) | Maximum listening distance, m (ft) | Front Loudspeakers Stereo & L-C-R | Side and Rear Loudspeakers (per channel) | Subwoofers for 2-channel Stereo [1)] | Subwoofers for 5-channel Surround [2)] |
|---|---|---|---|---|---|
| 75 (2,600) | 2.0m (6'7") | 1029A, 8030A | 1029A, 8030A | 7050A | 7060A |
| 75 (2,600) | 2.0m (6'7") | 2029A | 2029A | 7050A | 7060A [3)] |
| 75 (2,600) | 2.0m (6'7") | 2029B | 2029B | 7050A | 7060A [3)] |
| 85 (3,000) | 2.2m (7'3") | 1030A, 8040A | 1030A, 8040A | 7060A | 7070A or 2 x 7060A [6)] |
| 95 (3,400) | 2.3m (7'7") | 1031A | 1031A | 7070A or 2 x 7060A [6)] | 7070A or 2 x 7060A [6)] |
| 100 (3,500) | 2.3m (7'7") | S30D | S30D | 7070A or 2 x 7060A [6)] | 7071A or 2 x 7070A |
| 110 (3,900) | 2.4m (7'10") | 1032A, 8050A | 1032A, 8050A | 7070A or 2 x 7060A [6)] | 7071A or 2 x 7070A [4)] |
| 125 (4,400) | 3.5m (11'6") | 1037C | 1037C | 7071A or 2 x 7070A | 7073A or 2 x 7071A [4)] |
| 170 (6,000) | 4.0m (13'1") | 1038B & 1038BC | 1038B | 7071A or 2 x 7070A | 7073A or 2 x 7071A [4)] |
| 200 (7,000) | 4.5m (14'9") | 1034B & 1034BC | 1038B | 7073A or 2 x 7071A | 2 x 7073A [4)] |
| 240 (8,500) | 4.7m (15'5") | 1039A | 1038B | 7073A or 2 x 7071A | 2 x 7073A [4)] |
| 400 (14,000) | 5.5m (18'1") | 1035B | 1038B | 2 x 7073A | 3 x 7073A [4)] |
| 400 (14,000) | 5.5m (18'1") | 1036A | 1038B | 2 x 7073A [5)] | 3 x 7073A [5)] |

**Table 2 - Active Monitor System Selection Table**

**Table Notes**

1) If the system is planned to be eventually upgraded to surround, it is recommended to select the subwoofer model from the '5-channel surround' column for future SPL compatibility. In addition, selecting the larger of the two subwoofers will give additional headroom and lower distortion in a stereo system.

2) This table assumes that the bass management system built into the subwoofers is used. This is a situation normally encountered in the music industry. Less or smaller subwoofers maybe sufficient if the subwoofer is required to reproduced the LFE channel only. This is a situation normally encountered in the movie industry.

3) When using the digital input, the **2029A** and **2029B** cannot be used with the **LSE™ Series** subwoofer analogue crossover filters. The subwoofer could reproduce the LFE channel only.

4) Additional subwoofers of the same type may be required in a larger room with bass heavy program material.

5) Subwoofers are not necessarily required for a stereo **1036A** installation as these loudspeakers are already full range. For surround systems, use the monitor section in the console to route the LFE signal to the front loudspeakers. Alternatively, subwoofers can be used to reproduce the LFE channel only.

6) The LF extension of the 7060A is reduced from 19 Hz (7070A) to 29 Hz.

### 4.4 Multiple Subwoofers

When two or more subwoofers (except for the **7050A** as it does not have a sum output) are positioned close to one another mutual coupling is the fortunate by-product. This is due to the long wavelengths, associated with low frequencies, causing constructive superimposition. For mutual coupling, the subwoofers must be placed within ½ a wavelength of one another (85 Hz upper crossover frequency ½ wavelength is approximately 2 m). For example, two subwoofers give a 6 dB increase in acoustical output at the listening position – see Table 3.

| Total number of subwoofers | SPL increase compared to a single subwoofer | 7060A | 7070A | 7071A |
|---|---|---|---|---|
| 1 | 0.0 dB | 113 dB | 117 dB | 123 dB |
| 2 | 6.0 dB | 119 dB | 126 dB | 129 dB |
| 3 | 9.5 dB | 122.5 dB | 129.5 dB | 132.5 dB |
| 4 | 12.0 dB | 125 dB | 129 dB | 135 dB |

**Table 3 - Effects of Multiple Subwoofers Table**

## 5 Loudspeaker Placement Beyond 5.1

As mentioned earlier, various formats incorporate more than six channels. Some of these reproduction set-up do not use encoded soundtracks, but discrete channels replayed by a multichannel source. The aim of this chapter is to illustrate the different ideas and set-up arrangements that are in use or proposed today.

### 5.1 Dolby Guidelines

Dolby provides loose guidelines on loudspeaker placement for the different format playback configurations shown in Figure 10. These various playback configurations do not follow the ITU-R BS 775.1 recommendation mentioned earlier. Loudspeakers are not positioned on a circle radius and no angles are specified for the surround left and right channels. Also, the various loudspeaker placements shown are closer to a conventional home theater set-up than a professional control room layout.

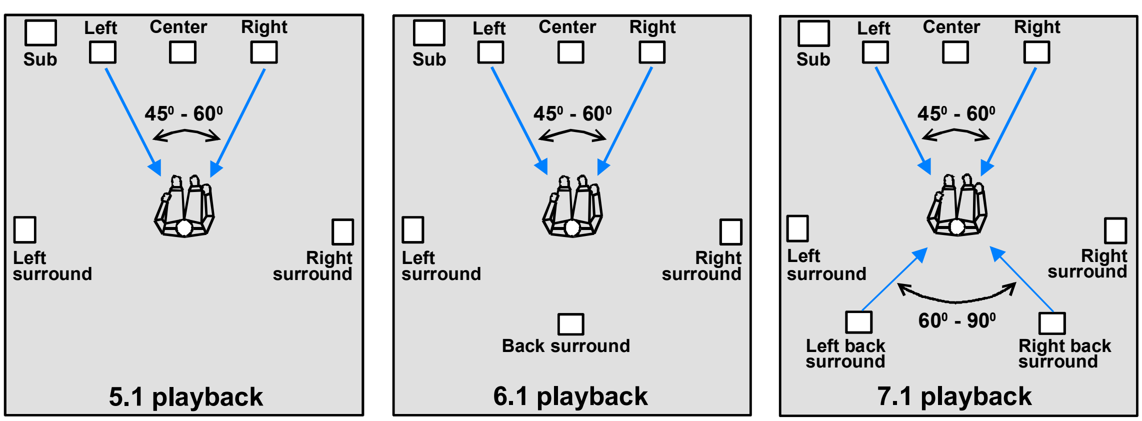


**Figure 10 – Dolby guidelines on loudspeaker placement for various playback configuration**

In the 5.1 playback configuration, the front left-right angles can spread from 45 degrees (typical movie theater set-up) to 60 degrees (ITU set-up). The 6.1 playback configuration displays a single 'back surround' loudspeaker, but a second alternative set-up is possible and illustrated in the following chapter. For the 7.1 playback configuration (used for the various matrixed formats that provide up to 7.1 channels of audio), Dolby suggest angles between ±135…150 degrees from the front centerline for the 'left and right back surround'.

## 5.2 6.1 Loudspeaker Set-up

The conventional five channels ITU set-up is illustrated in grey in Figure 11. The additional sixth channel is the rear centre channel. There does not seem to be an agreement between parties whether a single loudspeaker or two loudspeakers are used to reproduce this rear centre channel. Some guidelines suggest that using two loudspeakers at ±160 degrees from the centre line would provide better envelopment and stable sound-field, especially if the circle radius is large. When the radius is smaller, the use of a single loudspeaker at 180 degrees seems to be sufficient and appropriate. Further experiment and research seems to be needed to agree on a standard set-up.

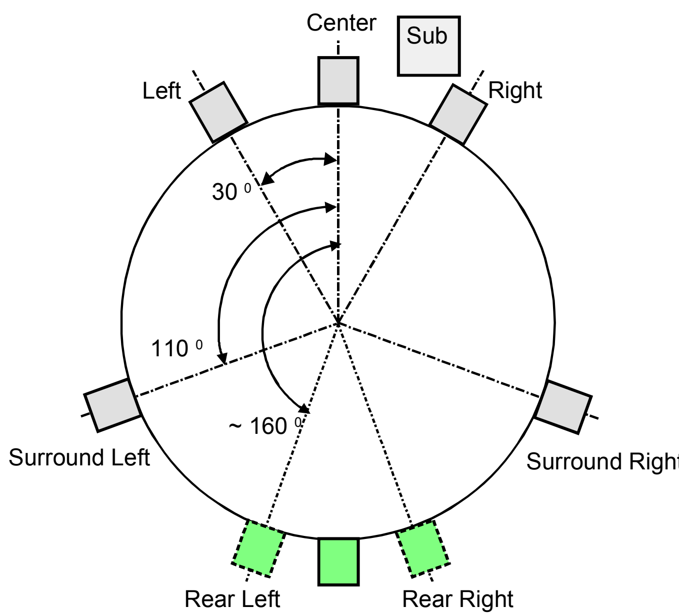


**Figure 11 - 6.1 Loudspeaker Set-up with either two loudspeakers at ±160 degrees or a single loudspeaker at 180 degrees**

## 5.3 Logic 7 Loudspeaker Set-up

Dr. David Griesinger has conducted extensive investigations regarding the positioning of rear channels. The Logic 7 surround sound format is one of the results of his work – see Figure 12. This format uses 2-channel stereo or 5.1 program material as the input signal and extracts, via a matrix system, four different surround channels. Two of these channels are called side channels and are placed at ±90 degrees from the centre line. The two other channels are placed between ±110 degrees (ITU compatible) and about ±140 degrees.

Dr. Griesinger is also advocating for the use of two subwoofers placed on either side of the room (±90 degrees) to provide a better "envelopment effect." The subwoofers have a 90 degree phase shift between them.

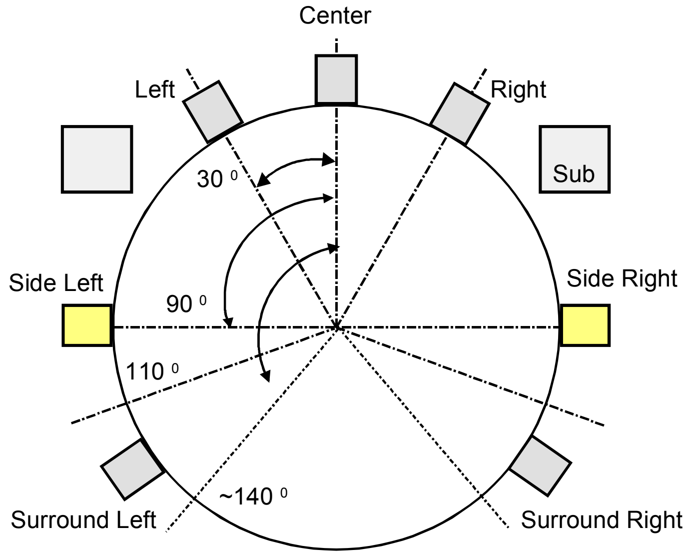


**Figure 12 - Logic 7 Loudspeaker Set-up with four surround channels and two de-correlated subwoofers.**

## 5.4 TMH Sound System Loudspeaker Set-up

Tomlinson Holman is behind the development of this TMH Sound System format – see Figure 13. Designed mainly for movie theatres, it includes the standard 5.1 loudspeaker set-up and placement (ITU set-up in grey), additional Extreme Left and Extreme Right channels at ±60 degree (for a better frontal spread), a Centre Rear channel (for better rear imaging) and two channels at ±45 degrees both in the horizontal and vertical plane (to give some sense of height). This set-up also uses two subwoofers that have a 90 degree phase shift between them.

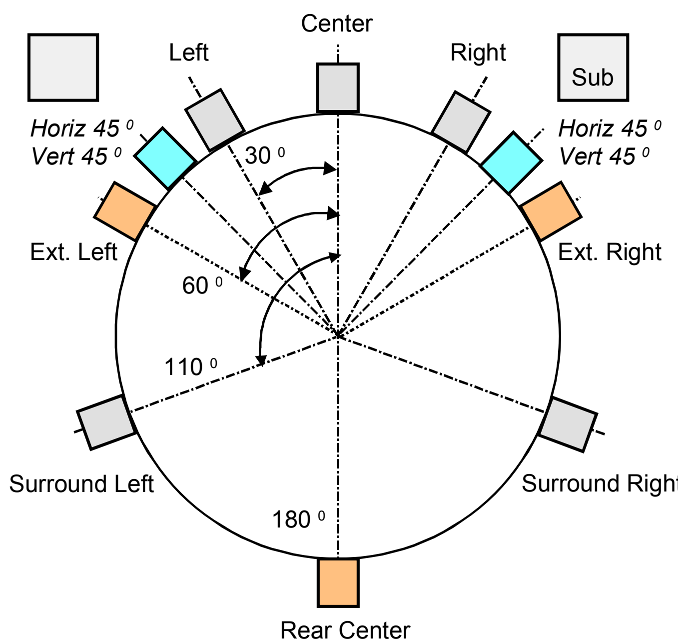


**Figure 13 - TMH Sound System Loudspeaker Set-up, with eight channels in the horizontal plane, two channels at 45 degree and two de-correlated subwoofers.**

## 5.5 22.2 Japanese Broadcasting Loudspeaker Set-up

In the scope of the next generation broadcasting system, the Japanese Broadcasting Corporation (NHK) has developped this 'ultimate' 22.2 multichannel audio system for ultrahigh-definition video with 4000 scanning lines[5]. The 22.2 reproduction system has three vertical layers of loudspeakers with 2 LFE's, namely 3 loudspeakers in the bottom layer, 10 loudspeakers in the middle layer and 9 loudspeakers in the upper layer – see Figure 14. The source materials used to develop and evaluate this system were various multitrack recordings and different mix-downs of these tracks.

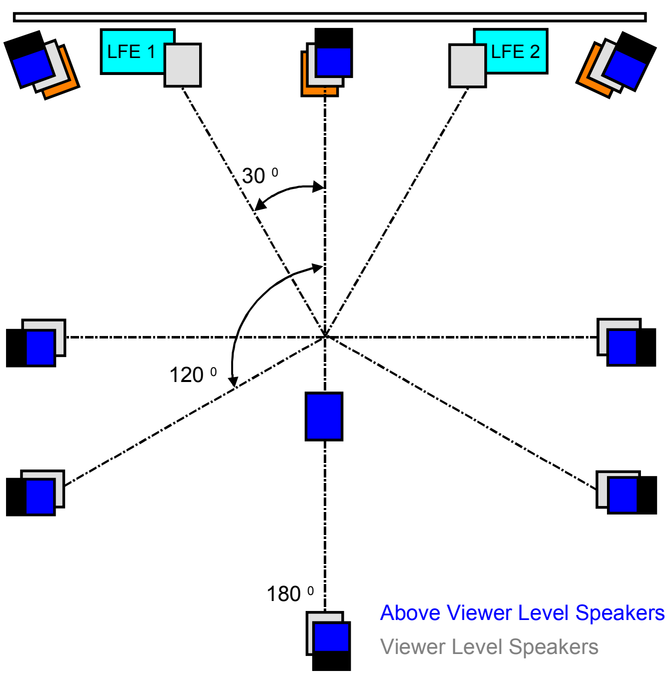


**Figure 14 – 22.2 Japanese Broadcasting (NHK) Loudspeaker Set-up for ultra-high definition pictures**

[5] *"Advanced Multichannel Audio Systems with Superior Impression of Presence and Reality",* Kimio Hamasaki, Koichiro Hiyama, Toshiyuki Nishiguchi, and Kazuho Ono, NHK Science & Technical Research Laboratories, Tokyo, Japan. Paper 6053, presented at the 116th AES Convention, May 2004, Berlin, Germany.

# D THE LFE CHANNEL

## 1 What is the *LFE* (.1) Channel?

Many people call the LFE channel the "subwoofer channel." In fact, the LFE channel is the space on the medium for the .1 encoded and band limited audio channel. The "subwoofer channel" is not a "channel" as such, as the subwoofer, together with the bass management system, replays a specific low frequency bandwidth. The subwoofer feed may consist of the LFE channel AND / OR the low frequencies of the main channels depending on whether bass management is used and how the system is set-up.

In a 5.1 audio production, the five main channels (*Left*, *Centre*, *Right*, *Surround Left* and *Surround Right*) are all full bandwidth, i.e. 20 Hz to 20 kHz. Before encoding, the LFE channel is NOT band limited, i.e. it is just another full bandwidth channel. Figure 15 illustrates the audio bandwidth of each channel in a 6.1 production, where the 6th rear surround channel has also full bandwidth.

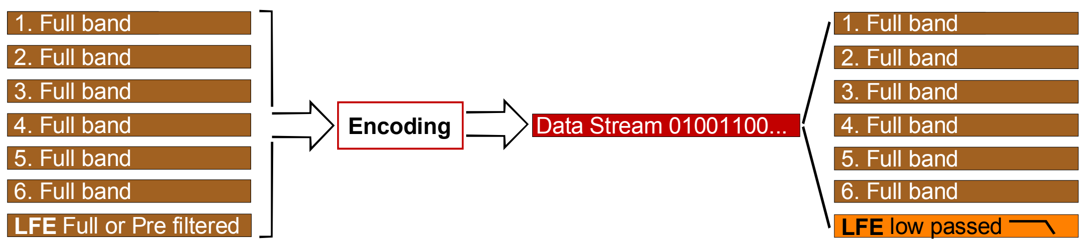


**Figure 15 - Audio Bandwidth of Encoded 6.1 Channel Material**

Once encoded, the *LFE* channel has a limited bandwidth (hence the label *'.1'*.) from 20 Hz up to various upper cut-off frequencies depending on the encoding formats – see Table 4.

| Encoding Formats | LFE Upper Cut-off Specifications |
|---|---|
| **Dolby Digital** (AC-3)<br>**Dolby Surround EX** (6.1) | 120 Hz<br>120 Hz |
| **DTS**<br>**DTS-ES** (6.1) | 120 Hz<br>120 Hz |
| **DVD-Audio** (MLP) | Full bandwidth channel |
| **Super Audio CD** (DSD) | Full bandwidth channel |
| **AAC** (MPEG-2 / MPEG-4) | Variable (up to 1 kHz) |
| **MPEG-2 BC** | Most likely variable… |
| **SDDS** | Variable up to 330 Hz |

**Table 4 – LFE channel upper bandwidth specifications.**

The *.1* channel was originally designed for use in theatres where the main channels could not reproduce the lower frequencies and additional headroom was required at these low frequencies to reproduce high SPL's. It is often given different names depending on its use:

- **L**ow **F**requency **Enhancement**
  This is a commonly used name as the *LFE* channel is often used to ‘enhance’ the bass in some way.

- **L**ow Frequency **Effects**
  This name is often used as the *LFE* channel is commonly used for explosions and other special effects requiring high levels of low bass energy.

- **L**ow Frequency **Extension**
  This is not an accurate name as the *LFE* channel’s frequency response extends down to the same frequency as the main channels, i.e. 20 Hz.

The use of the LFE channel is not consistent throughout the audio industry, as the needs of Movie Theatres (Cinemas), Home Theatres and Digital TVs are all different. Also, sound engineers new to multichannel surround sound mixing are experimenting with new ideas and techniques all the time. Various multichannel encoding/decoding processes exist, but one consistent feature for all formats is that all main channels remain full bandwidth and the LFE channel has variable upper cut-offs as already detailed.

# 2 How to monitor the LFE Channel

## 2.1 Headroom in the LFE channel

If the LFE channel is recorded (to tape, hard disk, etc) at the same nominal level as all the main channels then a loud explosion effect could easily overload the recording machine. If this were the case, this would defeat the object of having a special channel for loud sound effects as the main channels would have to be recorded at a lower level to accommodate the ‘big bangs’ on the LFE channel. To overcome this problem some encoding systems - Dolby Digital & DTS - require the LFE channel to be monitored at +10 dB. In doing so, the audio level recorded ‘to tape’ will have an additional 10 dB of available headroom as the engineer will naturally reduces the LFE channel level by 10 dB on the mixing desk output to maintain the sound level balance between the channels. Note that this increase in headroom is at the expense of 10 dB of signal-to-noise ratio, but as the LFE channel is eventually band limited in the encoder, it is a price worth paying.

## 2.2 How is this achieved in practice?

In Dolby Digital and DTS format, between the LFE channel monitor section fader and the acoustic output of the monitoring system, there must be a +10 dB level change - see Figure 17. This can happen in various places in the audio chain (see Chapter D 3), but it must be implemented during the production stage. Note that there are **NO** level changes between the console outputs and the inputs to the tape or hard disk recorder. The net effect is that there is 10 dB additional headroom on the LFE channel recorded to tape compared to the main channels.

All the tracks are then encoded into a single bitstream using various encoding schemes (namely Dolby Digital or DTS) and replayed accordingly - see Figure 16.

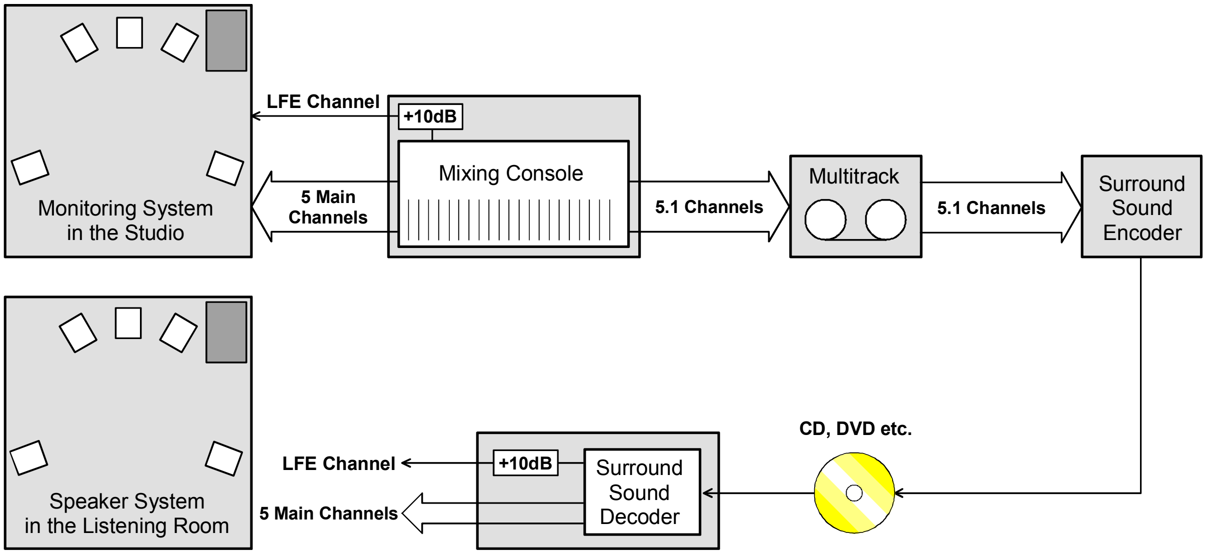


**Figure 16 - Complete Multichannel Audio Production Chain (Dolby Digital & DTS)**

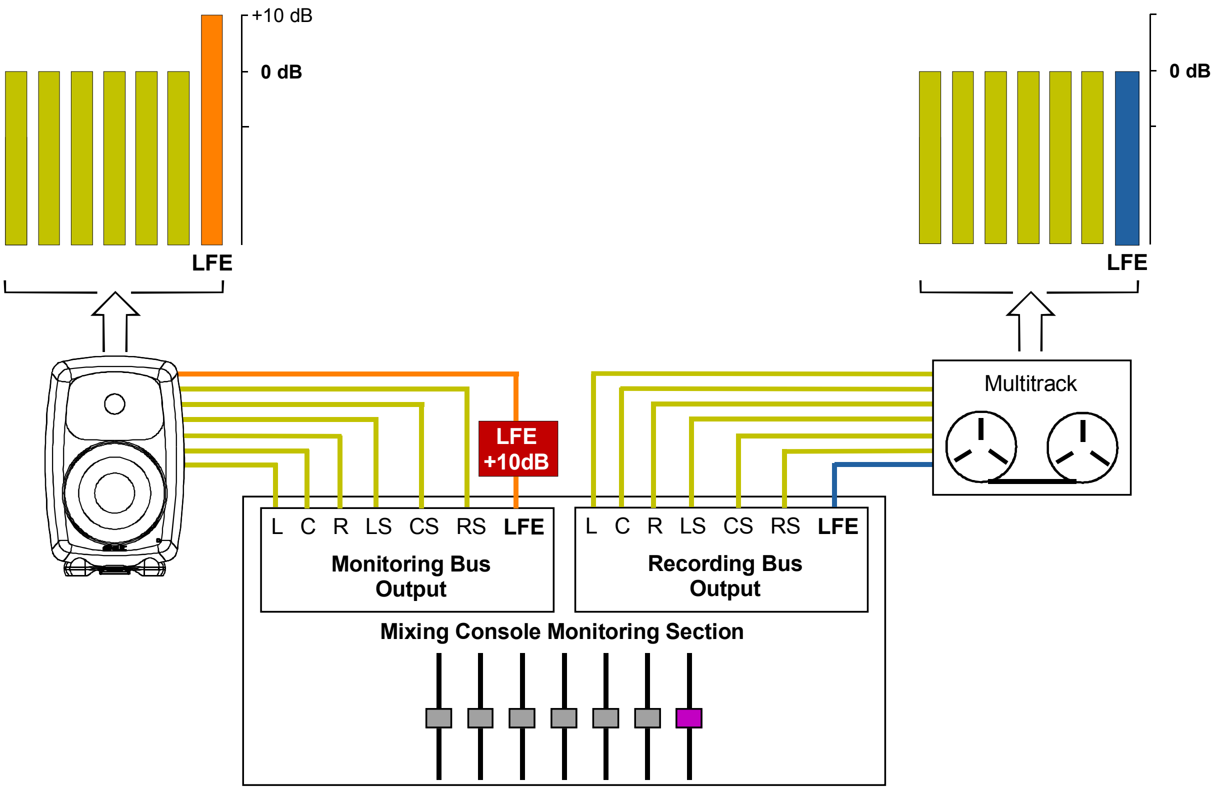


**Figure 17 – Mixing console 6.1 monitoring section with LFE channel initial gain structure for both the monitoring and the recording output busses. The LFE channel output level is then reduced to achieve additional 10 dB of recorded headroom for the LFE channel.**

# 3 Where is the LFE +10 dB used and inserted?

This method of increasing the headroom of the recorded LFE channel is used by all Dolby Digital (AC-3) and DTS encoding systems. All Dolby Digital and DTS consumer decoders automatically add 10 dB to the decoded LFE channel to restore the level balance between the main channels and the LFE channel as was originally heard in the production studio. In both formats, consumer decoders, DVD players with built-in decoders and theatrical (cinema) decoders use this scheme for the LFE channel.

The 10 dB gain on the LFE channel has to be implemented during the production stage to achieve correct level before encoding the data. Only **one** +10 dB gain is needed but two choices exist for the placement of the gain stage in the monitoring path:

1. If a large format console is used, the output matrix of the monitoring section will usually provide an internal +10 dB gain assignable on the LFE channel buss. In this case, no additional +10 dB gain, as provided by the Genelec bass management system, should be used.

2. If the monitoring section of the console cannot provide a +10 dB gain on the LFE channel buss, then the 'LFE +10 dB' DIP-switch provided on the Genelec bass management (7060A, 7070A, 7071A and 7073A) should be used.

As many different pieces of equipment are connected and used in a control room, the production engineer has to be aware of the detailed audio path of the monitored signal. The +10 dB gain on the LFE channel may, or may not, be necessary depending on the signal source. Here are some cases when the 10 dB LFE boost of the Genelec bass management system should **not** be used:

- If the +10 dB gain on the LFE channel is already implemented by another device (mixing console monitoring output or an external device).

- When producing an audio format that does not require the use of the +10 dB gain on the LFE channel at the decoding stage, such as DVD-Audio (MLP), SACD (DSD), etc. Please note when Dolby Digital or DTS recordings are "re-mastered" on DVD-A tracks the +10 dB gain will have to be utilized.

- When Dolby Digital Encoder/Decoder devices are connected in the monitoring path. In this situation, the 10 dB gain on the LFE channel will already be provided by the decoder.

- If a DVD player with built-in decoder is connected and monitored through the console output buses, then the +10 dB gain on the LFE channel is already provided by the decoding stage.

# 4 Producing Audio for the LFE Channel

The inclusion of a discrete LFE channel in multichannel audio is derived from the movie industry sector. Originally, and as stated in various standards, the LFE channel should carry signals, which are not included in the main channels. In the film industry, engineers use the LFE channel to produce special low frequency effects. However, the practical use of the LFE channel is open to many possibilities.

### 4.1 LFE Pre-Filtering at the Productions Stage

As seen above the LFE channel will be band limited after encoding, therefore many engineers suggest that it is highly important to filter the LFE channel at the production stage.

In the Dolby Laboratories publication, *'5.1 Channel Production Guidelines'* it clearly states about LFE channel:

> "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."

Dolby Laboratories also mentions that:

> "Normally, in low end decoders, one cannot change the low-pass frequency say to 100 Hz or 120 Hz."

That is mainly why this pre-filtering should occur in the production stage for all consumer multichannel productions.

### 4.2 Audio Content of the LFE Channel

Even if various recommendations exist concerning what should be put in the LFE channel, technically anything is possible. This has even more truth with formats like MLP (DVD-A) and DSD (SACD) that have an LFE channel that has full bandwidth after encoding (!). The production engineer has mainly two choices:

- **Coherent Signals**
  The main channels and the LFE channel receive the same low frequency content. The LFE is used just as an additional track to put some more low frequency energy. That is in some ways dangerous because there might be non-coherent radiation of the same programme content in the room by both the subwoofer and the main loudspeakers. This results in unpredictable acoustical summing of audio, which leads to inconsistent reproduction highly dependent on the listening environment.

- **Non-Coherent Signals**
  The LFE channel has different audio content than the low frequencies mixed in the main channels. This is the general guideline and practice followed by the movie industry that is far safer in terms of the acoustic summing found in different listening rooms.

### 4.3 Useful International Recommendations

Various bodies have given recommendations on how to use the LFE channel and what to put in that channel. This list is not exhaustive, but gives some ideas and suggestions and various opinions that exist amongst audio professionals:

- **ITU-R BS 775.1 Recommendation:**
  "The purpose of the LFE channel is to enable listeners, who choose to, to extend the low frequency content of the reproduced programme. (…) The subwoofer channel should not be used for the entire low frequency content of the multichannel presentation (…). The

subwoofer channel is an option at the receiver and thus should only carry the additional enhancement information. (…) The LFE channel should be capable of handling signals in the range 20 Hz – 120 Hz".

The use of the LFE channel for music only productions is strongly discouraged by many experienced recording engineers because:

- All main channels contain full bandwidth that can allocate any low frequency information.
- No LFE channel leads to less potential replay error at the consumer end.

In 'story telling' TV productions, it is highly recommended not to use the LFE channel and its content to avoid low frequency masking on the essential dialogue parts.

- **Dolby Laboratories Recommendations:**
  *Dolby Digital*:
  "Dolby generally recommends limiting the LFE signal to 80 Hz in the console to ensure best uniformity no matter how the program is delivered."

  *Dolby Film/Movie Theater:*
  The Subwoofer has to have a bandwidth from 20 Hz to 120 Hz without any crossover network to simulate the movie theater replay system.

The ITU recommendation is thus very important for the broadcast industry that has to produce multichannel material that is 5.1 ***and*** 5.0 compatible.

## 4.4 Implications for Multichannel Monitoring

As there are six (or more) channels to be replayed, it may seem logical to connect each output of the multichannel source directly to the appropriately positioned loudspeakers and to the subwoofer(s). However, without taking care that the entire bandwidth of each channel is properly monitored (with bass management or redirecting the LFE channel correctly), losses of important parts of the audio bandwidth or even serious cancellation effects can be induced, both electrically and acoustically.

Firstly, unless they are full bandwidth, each loudspeaker replaying the five or six main channels will have a -3 dB LF cut-off that is higher than 20 Hz. Therefore, the lowest frequencies of the main channels will not be reproduced and monitored. This is a serious compromise as all five or six main channels are full bandwidth even after encoding and decoding.

Secondly, Genelec subwoofers have their crossover point fixed at 85 Hz. After many investigations and subjective listening tests a fixed 85 Hz frequency has been chosen for best acoustical results. If one connects the encoded LFE channel to the subwoofer and the main loudspeakers are not connected to the outputs of the subwoofer, only audio material up to 85 Hz will be monitored via the subwoofer. No information above that frequency will be heard – even though the LFE upper cut off can be much higher!

That is why the Genelec 6.1 Bass Management system that is built-in to the LSE™ series subwoofers should be used, as this enables the rest of the bandwidth to be replayed (*see E Bass Management System chapter*).

# E BASS MANAGEMENT SYSTEM

## 1 What is Bass Management?

In stereo reproduction, signals from 20 Hz to 20 kHz need to be replayed. Multi-way loudspeaker systems will reproduce that bandwidth evenly. With multichannel audio, professional and consumer audio systems must be able to reproduce all frequencies from each channel. Main loudspeakers, subwoofers and crossovers should work together to provide a flat response for each channel.

In small studios with basic acoustic treatment and geometry, the frequency response of a loudspeaker system below 100 Hz is dominated by the modal response of the room. Strong low frequency standing wave patterns can be observed in small rooms that have parallel walls. Therefore, it is very difficult to achieve consistent low frequency response from multiple full-range loudspeakers in such limited space.

One solution to this basic acoustical problem is to employ a system called 'Bass Management'. Using active electronic filters and crossovers, one can extract the low frequency information from the main channels and route that information to a single subwoofer feed. The low frequencies are now originating from one single source that can be placed in an optimum position in the room. Furthermore, the LFE channel can also be monitored via this subwoofer and added to the low frequencies of the other main channels. Therefore, the Bass Management's basic and main goal is to ensure that the entire audio bandwidth of all channels can be accurately monitored.

**Dolby Laboratories**
*"Bass Management allows the user to redirect low-frequency information from any of the main loudspeakers to the subwoofer."*

**Tomlinson Holman, TMH Laboratories**
*"Bass Management in monitoring can be used to reproduce both the very low frequency content of the main channels, as well as the LFE channel, over one or more subwoofers."*

## 2 Bass Management and Small Loudspeakers

If the multichannel set-up uses small or mid sized loudspeakers, then a subwoofer should be included to extend the frequency response of all main channels and correctly monitor the LFE channel. The LF cut-off of the subwoofer in the room should be close to 20 Hz. As mentioned earlier, having one source of omni-directional low frequency information in a small room is usually quite beneficial as this simplifies the LF radiation pattern and enables the user to find the optimum position for the subwoofer. The second benefit, as discussed earlier, is to avoid the LF cancellation due to reflections off the walls behind the loudspeakers. Concerning the subwoofer itself, the low-pass filter of the subwoofer crossover should be very steep so that midrange information is sufficiently attenuated.

Now, for the subwoofer bandwidth, a fixed crossover point has been carefully chosen at 85 Hz. This means that for the Bass Management function of the six main channels, the crossover point between the subwoofer and the main channels is set at 85 Hz.

If we look at the bandwidth of the LFE channel after encoding, (see Chapter D1) we observe that this bandwidth can vary quite significantly. For that reason, the Bass Management for the LFE channel bandwidth provides three different setting possibilities for reproduction.

Note that the setting of the LFE subwoofer bandwidth will not at all affect the encoding of the LFE channel in the various formats, but does provide different replay bandwidths only.

# 3 Genelec 6.1 Bass Management System

To provide an integrated solution the Genelec 6.1 Bass Management System is built-in the 7060A, 7070A, 7071A and 7073A subwoofers. The active electronics of the subwoofer part is combined with the bass management thereby offering a very versatile system. The 6.1 Bass Management system features:

- Six main channels of inputs & outputs are provided for low frequency redirection. Each channel is identical and has a fixed crossover frequency set at 85 Hz.

- Bass management Bypass Function is provided for the six main channels. The LFE channel is not affected by the bypass function, regardless of the LFE bandwidth and redirection settings. The bypass function can be activated via an optional remote control box (RJ11) or via the ¼" jack bypass connection.

- Discrete LFE input with user-selectable reproduction bandwidth:
  - Upper cut-off at 85 Hz or at 120 Hz
  - Possibility to redirect (on/off) the LFE channel content above 85 Hz to the Front Centre loudspeaker when the bandwidth is set to 85 Hz.

- Selectable +10 dB LFE gain for the LFE channel monitoring. This function can be either selected in the bass management located on the subwoofer's side panel, or remotely via an optional remote control box (RJ11 connection).

- 'Slave' subwoofer function, selected with the 'Sum In' DIP-switch. This selection must be done to the 'slave' subwoofer to guarantee proper operation and all other switches of that particular switch group should be set to the 'OFF' position. The 'slave' subwoofer receives only one input (LFE IN/SUM IN) and follows *automatically* the parameters selected in the 'master' subwoofer.

- The discrete 'Sum Out' output sends the total subwoofer signal from the 'master' unit to another subwoofer(s). This gives the possibility to daisy chains multiple subwoofers to achieve higher SPL at the listening position relative to the room size.

- The three-colour power LED mounted on the side panel of the suwboofer indicates signal 'On' (green), 'Clipping' (orange) and 'Driver Protection' (red). This status LED can also be mounted remotely in any control surfaces, using one of the optional (RJ11) remote control boxes.

- Versatile DIP-switches controls to adjust the phase at the 85 Hz crossover frequency between the subwoofer and the 'satellite' loudspeakers. A built-in 85 Hz test tone generator provides a tool for easy on-site adjustment.

- Tone controls: Bass roll-off adjustment of –2 dB, –4 dB and –6 dB to retrieve a flat suwboofer frequency response in any room environments.

- Variable input sensitivity from +12 dBu to -6 dBu (except for 7073A: variable from +6 dBu to -12 dBu)

The typical connection of the Genelec 6.1 Bass Management is shown in Figure 18. All channels including the LFE channel are connected through the Bass Management. The main six channels are then connected from the output of the bass management to each of the main loudspeakers. A single output from the 'master' subwoofer (Sum Out) is fed into the 'slave' subwoofer unit (LFE In/Sum In).

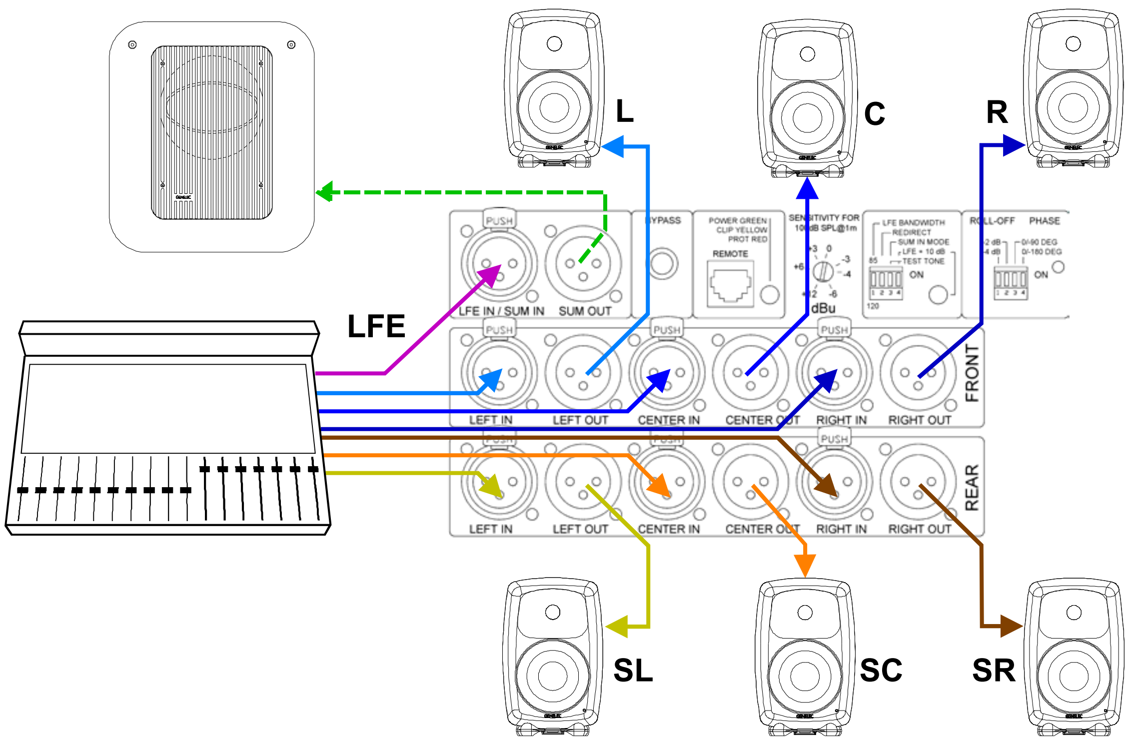


**Figure 18 - Connection to the Genelec 6.1 Bass Management.**

## 3.1 Full Range LFE Input

The first possibility for the LFE input bandwidth is the reproduction of the LFE channel by the subwoofer up to 85 Hz and then the redirection of the energy above 85 Hz to the front centre loudspeaker. This means that all information present on the LFE channel will be reproduced through the monitoring system, whatever the upper cut-off frequency of the LFE channel – see Figure 19.

Any noise, distortion artefacts and other unwanted sounds on the LFE channel can be monitored in addition to the LFE signal itself. In addition, any type of LFE content (coherent or non-coherent audio material) will be properly monitored without unpredictable summing of the low frequencies between the different channels. This should be the default setting for the bass management as it gives full range monitoring of the LFE channel.

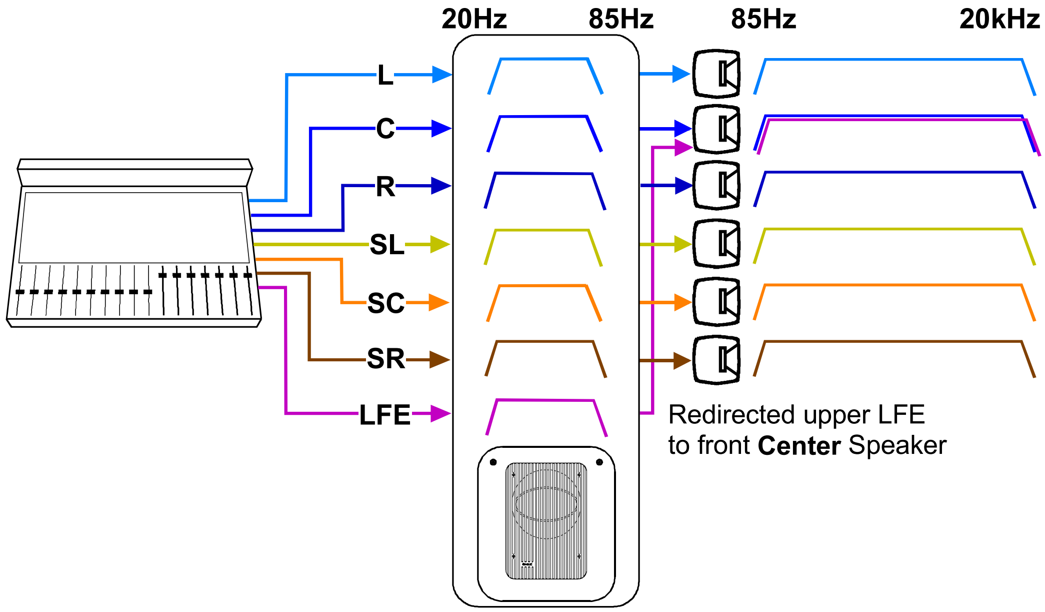


**Figure 19 - Multichannel Monitoring with Genelec 6.1 Bass Management. Redirection of the LFE channel above 85 Hz to the front centre loudspeaker.**

## 3.2 LFE Input Band-limited to 85 Hz

The second possibility for the LFE input bandwidth is the reproduction of the LFE channel by the subwoofer up to 85 Hz, *without* any redirection above 85 Hz to the front centre loudspeaker. This setting is comparable to pre-filtering the LFE channel in the mixing console using an 80 Hz LP filter. No information above 85 Hz present on the LFE channel will be heard through the monitoring system – see Figure 20.

This setting is not recommend for everyday use as the LFE channel has a bandwidth up to 120 Hz in Dolby Digital and DTS formats and wider bandwidths in some of the other formats. A good use for this setting is to simulate the effect of some (usually cheaper) consumer decoders that do not replay the LFE channel information that is above 80 Hz when the bass management is used. It is important for the mixing engineer to be aware of such a limitation so that the multichannel mix translates well in the home environment.

This configuration provides a replica of what will happen in many home surround system situations and yields to consistent reproduction of the low frequencies below 85 Hz. Also, any type of LFE content (coherent or non-coherent audio material) will be properly monitored.

As mentioned earlier, Dolby Laboratories and other companies recommend that, in DVD mixes for consumer home applications, the LFE channel is monitored and recorded with an 80 Hz low pass filter inserted in the console output buss. This second LFE channel bass management setting is a quick and easy way to simulate the monitoring part of that recommendation.

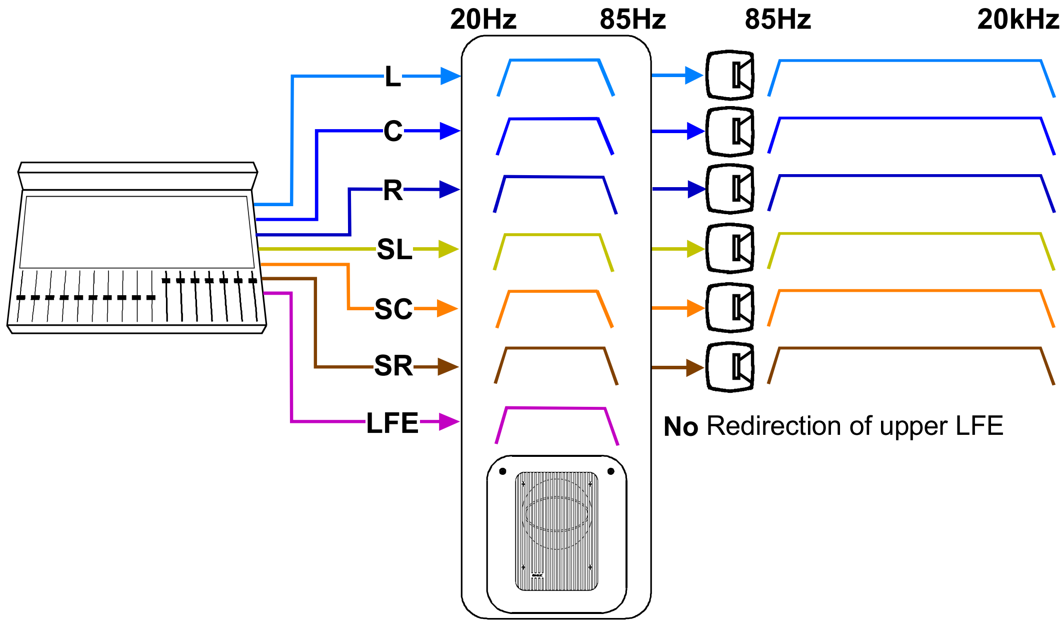


**Figure 20 - Multichannel Monitoring with Genelec 6.1 Bass Management. LFE Input band-limited to 85 Hz.**

## 3.3 LFE Input Band-limited to 120 Hz

The third possibility for the LFE input bandwidth is the reproduction of the LFE channel by the subwoofer up to 120 Hz *without* any redirection above that frequency. This means that no information above 120 Hz on the LFE channel will be heard through the monitoring system – see Figure 21.

This setting has been provided so that production facilities can emulate the replay systems that exist in movie theatres and cinemas. For these applications, there are strict rules on how the replay system should perform. Dedicated subwoofers with a bandwidth from 20 Hz – 120 Hz reproduce the low frequency content for the 35 mm movie soundtracks. However, it must be noted that when mixing music and sound effects for film release, engineers always use non-coherent low frequency information between LFE channel and main channels. In other words, the low frequency LFE channel content is different from any other main channel low frequency audio content so that, potentially, unpredictable acoustic summing in the room is avoided.

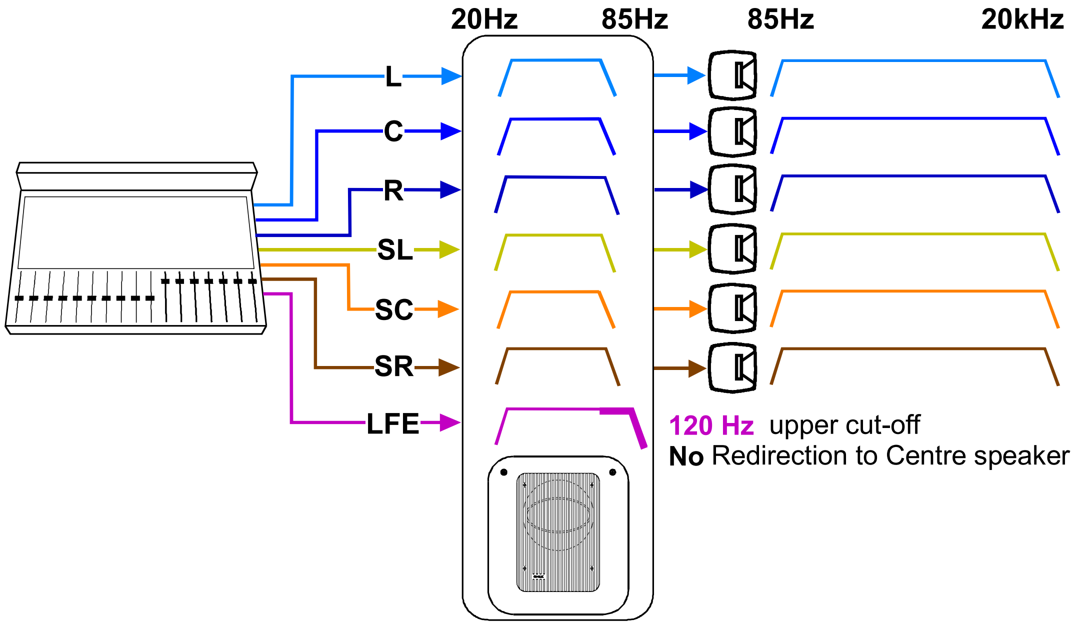


**Figure 21 - Multichannel Monitoring with Genelec 6.1 Bass Management. LFE Input band-limited to 120 Hz.**

## 3.4 Bass Management Bypass Function

The LFE channel is unaffected by the bypass function as it affects the main channels only. This means that the LF extension of all main channels, provided by the subwoofer, is bypassed and the full bandwidth is monitored via the main loudspeakers only. The low frequency reproduction of the main channels will extend as low as the main loudspeaker low frequency extension (i.e. for the 8040A: -3 dB LF @ 45 Hz). At the same time, the LFE input remains untouched and the settings concerning 'routing to the centre channel' and 'bandwidth' (85 Hz or 120 Hz) will be unchanged – see Figure 22.

The bypass function can be activated in two different ways in the 7060A, 7070A, 7071A and 7073A subwoofers:

1. Via the mono ¼" jack connector on the subwoofer side panel using a standard mono shorting switch.

2. Via the side panel RJ11 remote connector. The optional remote control box provides switching and status LED for the bypass function.

Note that the ¼" jack bypass switch should be set to 'OFF' for the RJ11 bypass remote control to work and display the status of the bypass function.

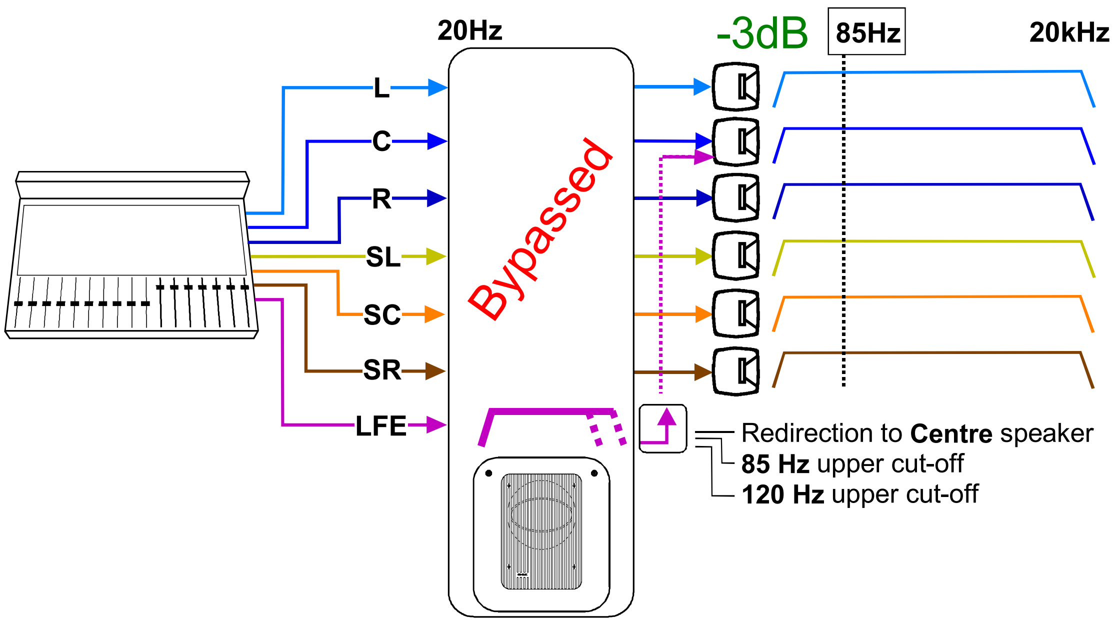


**Figure 22 - Multichannel Monitoring with Genelec 6.1 Bass Management. All main channels have the subwoofer bypassed, and the LFE channel remains unchanged.**

## 3.5 Optional Remote Control Facilities (RJ11)

When using the bass management RJ11 connection, two remote control boxes can be connected to the subwoofer RJ11 socket. Both remote boxes have dual RJ11 sockets for daisy chaining the control signal from one box to the next.

1. **Remote Power/Overload LED**
   This remote box has a three-colour LED (green, yellow & red) that copies the status of the three-colour LED mounted on the side panel of the subwoofer. The remote box has two RJ11 sockets for daisy chaining the control signal to the second remote control box. Schematic of the wiring and connections is detailed in the operating manual of the subwoofers so that the remote LED can be mounted in any equipment and environment.

2. **Bypass and LFE +10 dB Control**
   The second remote control box has two switches and two status LED's (yellow colour). The ¼" Jack bypass remote has to be 'OFF' for the remote control via RJ11 switch to work and display its status. The subwoofer side panel LFE +10 dB switch also has to be set to the 'OFF' position (no light) for the RJ11 remote control LFE +10 dB switch to work and have correct display status.

## 3.6 Input Sensitivity

The subwoofer input sensitivity is variable from +12 dBu to -6 dBu (except for the 7073A, variable from +6 dBu to -12 dBu). Typical when a subwoofer is placed close to a wall, it experiences acoustical loading which can be compensated by reduding the input sensitivity by 6 dB. So, the purpose of this sensitivity control is to allow the user to adjust the subwoofer acoustical level to be aligned with the main loudspeaker's level.

### 3.7 Bass Roll-off Adjustments

Bass Roll-off adjustments (-2 dB, -4 dB and -6 dB) can be useful to retrieve a flat subwoofer frequency response. When subwoofers are placed on the floor and against one or two walls, the effect is not only an in-band gain in the frequency response, but often a change in the frequency response shape. The adjustment of the Bass Roll-off will enable the user to achieve a flat subwoofer response.

### 3.8 Phase Adjustments

When the connection to the Genelec Bass Management is properly done, it is possible and necessary to adjust the Phase at the crossover point once the multichannel system is set-up in the room. Genelec Bass Management allows for 0, 90, 180 or 270 degrees @ 85 Hz. This guarantees that the main loudspeakers and the subwoofer are in phase at the 85 Hz crossover point. An 85 Hz tone generator is provided in the Genelec bass management so that it is easy do the necessary on-site phase adjustments.

This Phase adjustment has to be done for each subwoofer placed in the room, as they possibly will have different phase adjustments depending on their different physical location in the control room relative to the 'satellite' loudspeakers. This is especially important in the case where one uses a master/slave subwoofer configuration.

## 4 Monitoring the LFE with Full Range Loudspeakers

Some music recording engineers would rather not use any subwoofers at all as they would rather use full bandwidth loudspeakers without subwoofers. The point is, for example, that in a real orchestra performance all the low frequency information does not come from one physical point in space and that the harmonic content and various reflections in the hall guide our spatial cues towards different places for the instruments in the orchestra.

Other engineers in the audio post-production and movie industry sectors want to use subwoofers to reproduce the LFE channel only. In doing so, they emulate the replay conditions in movie theatres. In that situation, the audio content of the LFE channel has to be carefully considered as cancellations may occur between the audio replayed by the main channels and by the subwoofer.

Whatever the opinions are, it is also possible to monitor accurately all main channels without a subwoofer when using multiple large full range loudspeakers. The important point is to be able to monitor each channel in its full bandwidth. Therefore, in this case, the five or six full range loudspeakers will reproduce the five or six main channels and the LFE channel should be monitored via one, or more, of the full range loudspeakers. Re-routing is required and can be done inside the mixing desk's multichannel monitoring section or in a separate multichannel monitor controller – see Figure 23. The most widely used routing for redirecting the LFE channel is to send it to the centre channel. There are also other set-ups where one redirects the LFE channel to front left and front right loudspeakers (note that decoders do this if there is no subwoofer) or even to all three front loudspeakers (this is acoustically the best solution for SPL reasons).

Note that when routing the same LFE channel to more than one loudspeaker, the total SPL of that channel will increase. For example, routing the LFE to the front left and right loudspeakers will add 6 dB SPL to the overall LFE channel level, therefore, the output level of the LFE channel should be reduced accordingly in the console monitoring matrix output.

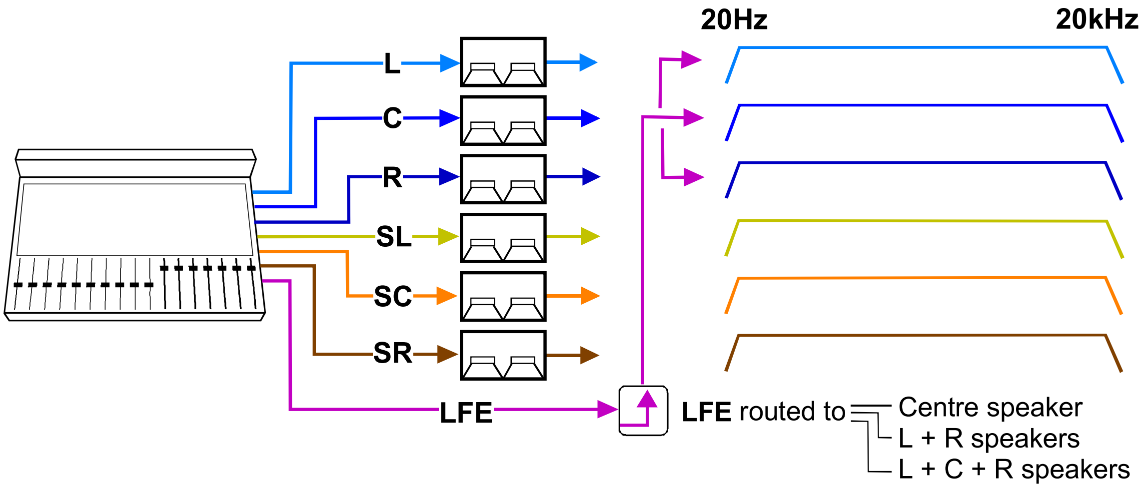


**Figure 23 - Re-routing the LFE channel to Full Range Loudspeakers**

# 5 Level Alignment of a Multichannel Surround System

## 5.1 How to Align the Levels of a Multichannel Systems

The main goal in the alignment of a multichannel system is to achieve that the subwoofer sound output level (within the band limited subwoofer frequencies) is the same as the sound output level compared to the main loudspeaker system (at other frequencies). The LFE output of the mixing desk or decoder should be connected to the LFE input on the subwoofer.

## 5.2 What Reference Level?

To ensure and achieve repeatable results in the finished production, the SMPTE (Society of Motion Pictures and Television) organisation has developed standard monitoring levels for cinema postproduction work. For film mixing, the SMPTE reference is 85 dB, with full bandwidth pink noise, read with a sound pressure level meter (SPL) set on C weighted/Slow scale.

For release of film material on television, various standards state that the operating mixing level should be somewhat lower so that low-level dialogues, which are easily heard in a quiet and acoustically well treated control room, are mixed slightly higher. This is to ensure that in the home environment with higher background noise levels the dialogue is clearly heard. However, for music mixes, there are no standardised levels – as for stereo – because the level that the engineer chooses is arbitrary and based on personal taste, as is the level chosen by the end user.

Thus, one absolute reference level does not really apply for all multichannel surround sound applications, so the following alignment procedures do not mention specific values but rather the relative levels between the various loudspeakers at the listening position.

## 5.3 Calibration of the Level and Frequency Response with an MLS Signal

First, there is no point in setting the channel levels until the loudspeakers have been calibrated for their individual frequency responses. Furthermore, there is no point in calibrating the frequency responses using the acoustic tone controls if there are fundamental acoustical problems in the room. Level setting is the last setting to be made once all the other issues have been resolved.

The acoustical performance of the main channels and the subwoofer should have a flat frequency response for accurate monitoring. First, make sure that the switch labelled 'LFE +10 dB' is set to 'off' position. Then proceed as follows:

- Calibrate the five or six main channel frequency responses using an MLS or similar measuring system with the subwoofer bypassed or disconnected.

- Then connect the Genelec subwoofer as described in the supplied operating manual and adjust the subwoofer level, bass roll-off and phase controls (relative to the centre channel) so that the measured frequency response of the subwoofer **and** near field monitor is extended smoothly down to 29 Hz (7060A) or 19 Hz (7070A, 7071A and 7073A).

Note that there should be no 10 dB level changes at low frequency (in the subwoofer bandwidth) compared to the mid and high frequencies as all the headroom level changes of the LFE channel are done electrically in the mixing desk, the bass management system or the decoder.

## 5.4 Alternative Level Calibration Methods

If MLS type equipment is not available for aligning the system then follow the guidelines that can be found in the operating manual for adjusting the frequency response. Remember though that there is still the need to align the system for level. Below are listed two alternative methods but one should note here that the accuracy of these methods depends greatly on the quality and the LF response of the SPL meter. First, make sure that the switch labelled 'LFE +10 dB' is set to 'off' position in the subwoofer bass management.

- **Level Calibration using a 1/3 Octave Real Time Analyser, *Broadband* Pink Noise and an SPL Meter**
  Connect the Genelec multichannel system and play broadband pink noise signal (20 Hz – 20 kHz) through the subwoofer and one of the main channels, for example, the centre channel. Set the level of each band on the RTA Analyser to read the same value, both in the subwoofer bandwidth **and** in the main loudspeaker bandwidth. The specific absolute reference level depends on your application area as mentioned earlier and can be checked using the sound pressure level meter. Place pink noise through the whole system and adjust the level of each channel on the mixing desk to give the same acoustical level in the room. The level depends on the level of the signal but if it is set to -20 $dBFS_{rms}$ (-18 $dBFS_{rms}$ in Europe) the SPL meter should read 85 dB for cinema/theatre work, 80-85 dB for television and 85-95 dB for music.

- **Level Calibration using *Filtered* Pink Noise and an SPL Meter**
  You need to have filtered pink noise to calibrate the levels of the subwoofer and the main channels. You can use a copy of the TMH Laboratories 'Multichannel Studio Test Tape' that includes the various test signals required. The essential point is that you need two types of test signal:

  - Filtered pink noise from 500 Hz to 2 kHz for the mid band frequencies of the main loudspeakers. This range is well away from the subwoofer's bandwidth and suffers less from room cancellation effects.

  - Filtered pink noise from 20 Hz to 80 Hz, to calibrate the subwoofer level.

Note: If the recorded level of both the filtered pink noises is –20 $dBFS_{rms}$ (-18 $dBFS_{rms}$ for Europe) then an absolute level calibration can be made according to your application area, for example, 83 dB SPL for theatrical work, 78-83 dB SPL for television and, typically, 83-93 dB for music. This level is 2 dB lower than for broadband pink noise as there will be less energy in the room due to the limited bandwidth signal.

- Connect the Genelec multichannel system then play the 500 Hz to 2 kHz filtered pink noise to adjust each of the main channels individually. Set the SPL Meter to C-weighting & Slow scale, and note the reading (say it is 83 dB SPL). All five channels have to be adjusted to this same level.

- Next, play the 20 to 80 Hz filtered pink noise through the subwoofer. The correct adjustment should give a reading 3 dB lower than the one for the main channel loudspeakers (in our example, 80 dB SPL). The reason for the difference in level reading is that most SPL meters have a built in HP filter. If there is no HP filter in the SPL meter then the reading should be the same as for the main loudspeakers as the bandwidth, and hence energy in the room, of the two signals is the same.

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