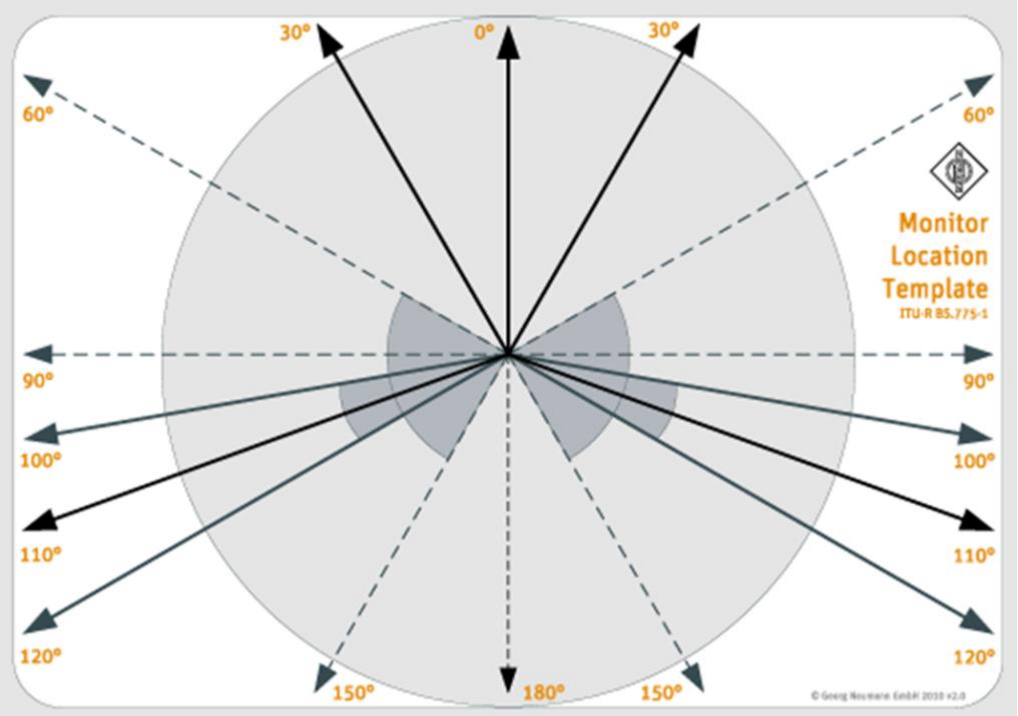
ACOUSTICAL / APPLICATION ANSWERS

How do I set up the loudspeakers in my room?

Installation of the loudspeakers is explained in the supplied very extensive operating manual, so please read it. If the operating manual is not available, it can be downloaded from the "Download" section of the product's webpage. Here is a brief check list of things to do:

- Locate the loudspeaker at the correct angle for the signal it is to reproduce – see Monitor Location Template below.
- Point the acoustical axis towards the listening position or centre of the listening area.
- If the loudspeakers are not positioned at the same distance, time delays are required for the closer ones.
- Wire the loudspeakers and subwoofer(s) in accordance with the user manual instructions.
- Calibrate the loudspeakers and subwoofers to achieve the desired response at the listening position.



Use this template to help position the loudspeakers' acoustical axis at the correct position:

- Print this page.
- Place it at the listening position or center of the listening area.
- Pull a cable from the centre of the circle towards the loudspeaker's acoustical axis, as defined in this document.
- Adjust the position of the loudspeaker so that the cable runs along the appropriate arrow.
- The loudspeakers' distance from the centre of the circle can be simultaneously adjusted.
- For 2.0 systems use ±30°.

- For 5.1 systems use ±30°, 0°, and ±110°
- For 6.1 systems use ±30°, 0°, ±110°, and 180°
- For 7.1 systems use ±30°, 0°, ±90°, and ±120° or ±150°.

What is the target response of a loudspeaker at the listening position?

Professional Monitoring

A studio monitoring system should, at the listening position, acoustically exactly reproduce the electrical input signal. This leads to the target definition that is a flat frequency response. This means that the magnitude of the frequency response should be flat and that the phase of the frequency response should be linear.

When looking at systems used to reproduce the program material, the responses are generally not flat. Usually this then leads to a long discussion about "translation". To short-circuit this discussion, take a sample of reproduction system responses (home hi-fi, car audio systems, small radios, personal audio devices, distributed audio systems, PA systems, etc.) and average them together. The result will be a flat magnitude response. So a single "reference" listening system, for example a car audio system, is not a safe reference.

The monitors are designed to have the flattest magnitude response and the most linear phase response possible. The response changes in an often predictable way when the loudspeaker is placed into a non-anechoic space, for example the bass boost due to the acoustical loading from a wall. The acoustical controls ("Bass" and subwoofer level) can then be used to get back to a flat response at listening position.

There some cases where a non-flat response is required or even desirable…

Movie Industry X Curves

X Curves in movie mix rooms to simulate the response typically seen in movie theatres (ANSI/SMPTE 202M-1991 and ANSI/SMPTE 222M-1994). This response rolls off at high frequencies due to the perforated screen positioned in front of the loudspeakers, the long listening distances, and the multiple off-axis listening positions. There are two types of X Curve:

- **X Curve for a small room** (<150 cubic meters or 5300 cubic feet) has a flat response up to 2 kHz and then a 1.5 dB/oct. roll-off above 2 kHz. The tolerance is +/–3 dB with some additional tolerance at the frequency extremes.
- **X Curve for a large room** (>150 cubic meters or 5300 cubic feet) is flat from 63 Hz to 2 kHz and then rolls-off at 3 dB/oct. above 2 kHz. Above 10 kHz there is a steeper 6dB/oct. roll-off. Additionally, below 63 Hz there is a 3 dB/oct. roll-on. The tolerance is +/–3 dB with some additional tolerance at the frequency extremes.

Note that the X Curve should not be used for mixing material to be placed onto DVDs for domestic consumption. A monitoring system with a flat response system should be used for reasons given above.

Listening for Pleasure

When listening for pleasure, some cosmetic changes to the sound are often required. Every detail in the high frequency is not necessarily desirable, for example the sound of shuffling papers in news broadcasts. Conversely more low-frequency energy can create more impact in movies. This is a subjective area, however as a general guideline, try to avoid more than 3 dB of change away from a flat response.

A good room designer can help you with this.

How often should I calibrate the loudspeakers?

There are two types of calibration:

- **Production Calibration** – this only needs to be performed if the electronics or one of the drivers is replaced during servicing. This process can only be performed by suitably equipped service centers.
- **In-situ Calibration** – this only needs to be performed if something changes in the listening room, for example, the loudspeakers are moved, the listening position is moved, or there is a change in the room acoustics.

How do I set the acoustical controls on the loudspeaker?

There are three ways to find the best combination of acoustical control settings on a loudspeaker or subwoofer:

- **Listening** – Listen to program material and adjust the acoustical controls until the loudspeaker reproduces the material correctly. This method is unreliable, time consuming, and affected by the listener’s preconceived ideas.
- **Advice** – The operating manual gives recommended acoustical control settings for the most commonly encountered acoustical conditions. They are a good starting point but not always the best setting for a particular installation, so some experimentation is still required.
- **Measurement** – This is the best method for setting the acoustical controls, unfortunately not everyone has a calibrated acoustical measurement system and/or the knowledge to use one. Contact your local distributor to see whether they can help you with calibrating your loudspeaker system.

A good room designer can help you with this.

I can hear a strange pressure difference between my ears?

The left and right loudspeakers are possibly wired out of phase. If this is the case there is probably a lack of bass too. Check the cabling polarity and the source for any phase inversions. Also use different program material to ensure that the sound is not some special effect used in the production stages. Additionally ensure that the distance to both loudspeakers is the same and that any distance difference is compensated with an appropriate time delay. The formula for this is:

$$t_{delay} = (d_{loudspeaker1} - d_{loudspeaker2}) / c$$

Where:

- t_{delay} is the time delay in seconds required to compensate for the shorter listening distance of that loudspeaker
- $d_{loudspeaker1}$ is the distance of the loudspeaker furthest from the listening position
- $d_{loudspeaker2}$ is the distance of the loudspeaker nearest to the listening position that needs to be delayed
- ***c*** is the speed of sound in air at 20°C = 344 m/s, or use 1120 ft/s if the measured distances are in feet, or 13440 in/s if the measured distances are in inches.

Example:
The left channel loudspeaker is located 2.3 m from the listening position and the right at loudspeaker 2.1 m. The required time delay for the right loudspeaker is:
$t_{delay} = (2.3 - 2.1) / 344 = 5.81 \times 10^{-4}$ s = 0.581 ms

Time delays are often not available on the monitor outputs of mixing consoles or have a coarse resolution in the case of home theatre decoders.

To overcome these issues we have started to build in time delays to the D versions of products.

How do I improve the imaging in my listening room?
When a loudspeaker is placed some distance from a reflective surface a cancellation is possible. The longer the distance, the lower the frequency of the cancellation. The more solid the surface is and the closer the loudspeaker is to the surface, the stronger the reflection and the deeper the cancellation. If the reflections have different distances the response at the listening position will not be the same from both loudspeakers as the cancellations will occur at different frequencies. At each frequency the image will move towards the stronger sound, i.e. away from the weaker cancellation. The result is a blurring or spreading of the image. Another source of differences in the response at the listening position is acoustical loading which may be different for different loudspeakers in a pair. Again, the image will move towards the higher level (uncorrected loading) frequency.

To achieve good imaging one should start with well-matched loudspeakers. Next, follow this frequency-related guide for good imaging:

- Poor bass imaging comes from an unsymmetrical room, so ensure that the room and loudspeakers' positioning in the room is symmetrical.
- Poor mid-range imaging comes from objects in the room, so all furniture, equipment, and racking should be symmetrically placed in the room.
- Poor treble imaging comes poor angling of the loudspeakers, so point all loudspeakers at the listening position.
- Poor imaging can also come from loudspeakers positioned at different heights.

Finally, the acoustical controls should be set to achieve the same response from both loudspeakers when measured at the listening position. In a symmetrical room the acoustical controls should be set to the same values for a left/right pair so that the phase response is also matched.

A good room designer can help you with this.

I have too much bass from my loudspeakers / My mixes are bass light

When a loudspeaker is placed near a large solid surface the sound pressure increases. This increase is called acoustical loading and has a first-order characteristic. The size of the object defines what frequency range is affected. For a wall, the bass response below a certain frequency is increased (shelf shape). The corner frequency of this loading depends upon the size of the bass driver and the cabinet: a smaller cabinet results in a higher corner frequency. For a mixing console, the loading will occur over a narrow frequency band, typically in the range 100-300 Hz (PEQ shape) depending on the size of the mixing console. The solidity of the wall or the console and the distance of the loudspeaker from them define how much boost is seen. The acoustical controls for wall and console loading are “bass” and “low-mid” respectively. Uncorrected acoustical loading results in a colored sound quality, auditory masking, and mixes with the bass mixed out.

The same is true of subwoofers, except that the entire passband level is increased. They do not have a shape to their loading due to their limited bandwidth. The solution here is simply to reduce the level of the subwoofer using the level control(s). Typically 4–6 dB is required when a subwoofer is placed next to the front wall. Additional attenuation will be required, together with some use of the “low-cut” control, when a subwoofer is placed in a corner.

Another source of excessive bass is a strong resonance(s). Typically they come from parallel walls, parallel floor-ceiling, a lack of acoustical damping in the room, undamped cavities, and/or undamped air conditioning ducts. These should be traced and treated acoustically. A good room designer can help you with this.

I have too little bass from my loudspeakers / My mixes are bass heavy

When a loudspeaker is placed some distance from a reflective surface a cancellation is possible. The longer the distance, the lower the frequency of the cancellation. The more solid the surface is and the closer the loudspeaker is to the surface, the stronger the reflection and the deeper the cancellation. A lack of bass in the monitoring system will result in an increase in bass in the mix to compensate. Cancellations cannot be equalized out so the source of the reflection should be traced and the surface acoustically treated (absorption or diffusion added to reduce the level of the reflected sound at the listening position), angled (to redirect the reflection away from the listening position), or removed completely. Low-frequency cancellations (below 150 Hz) are caused by large reflecting surfaces some distance from the loudspeaker, such as walls. Low-mid (150-500 Hz) cancellations are caused by the second and higher “teeth” of the comb filter resulting from lower frequency cancellations. So tackling the low-frequency first-order “tooth” first will improve the low-mid frequencies too. A good room designer can help you with this.

What measuring techniques and equipment do you recommend to ensure that my monitors are calibrated correctly for my room?

Product Specialists in the distribution network use the exponential swept-sine wave technique to measure loudspeakers in the design process, production, and in-situ calibrations. It is fast, reliable, repeatable and accurate, and can be found in a number of software packages, for example Monkey Forest (for engineers and researchers) and WinMLS (somewhat more user friendly). A good alternative is the MLS technique. Other methods maybe slow, unreliable, and/or inaccurate.

Once measurement software has been selected it should be correctly set up and used with calibrated equipment. These days even quite cheap soundcards have a reasonable response, especially when the response is corrected using a loopback measurement (we use the Edirol UA-25). Measurement microphones should also be used in the correct way and calibrated. Unless room noise floor measurements are being taken, the microphone does not need to have a low self-generated noise (we use a custom designed Sennheiser MKE 2 that unfortunately is not commercially available). Earthworks manufacture a good commercially available omni-directional measurement microphone. B&K are another, but much more expensive, manufacturer.

How should my room be acoustically treated?

In general the following guidelines should be observed:

- Avoid parallel walls which lead to strong resonances.
- Aim for a low (0.2 to 0.4 seconds) and flat reverberation time.
- Minimize strong reflections that get back to listening position. Those that do should have 20 dB less level than the direct sound level.
- For good imaging, ensure left-right symmetry of the room, equipment, and loudspeaker positioning.
- Diffusers are generally used at the back of the room. Absorption will be more widely distributed around the room in a multichannel installation, rather than at the back in a 2-channel stereo room.
- Flush-mounted loudspeakers should sit in a well-constructed hard wall.
- Position the loudspeakers at the correct angles (see international recommendations) and point towards the listening position (use the acoustical axis).
- Calibrate the loudspeakers using a reliable acoustical measurement system or use the recommendations in the operating manual.

Further advice can be found in text books, magazines, and the internet, although one has to verify that internet advice is accurate and relevant. There is no substitute for a good room designer.

Can I use multiple loudspeakers to increase the sound level in the room?

The use of multiple loudspeakers comes from sound reinforcement applications. The products in use are specifically designed to be arrayed as they have a narrow dispersion, such as 80x50 or 50x40, and trapezoidal-shaped cabinets for fan-shaped clustering. This is done so that the interaction of the sound energy between the cabinets, and hence comb filtering, is minimized. The result is a loudspeaker that can be arrayed to give the desired coverage angle and maximum SPL.

Conversely, loudspeakers designed for studio applications have a wider horizontal dispersion, typically 90°, and straight-sided cabinets. Arraying these cabinets will result in considerable comb filtering in the midrange and high frequencies. This will be clearly audible as the listening position is moved around in the front of the loudspeaker array. So if the maximum sound level from a loudspeaker is insufficient, a larger loudspeaker should be used not a pair of loudspeakers. An exception to this is subwoofers which can be arrayed due to the long wavelength of low frequencies.

Do I need to leave space behind the loudspeaker for cooling?

Although the heat management in the loudspeakers is very efficient, some space 5 cm (2”) is required for cool air to pass over the heatsinks. Failure to adequately ventilate the electronics will result in premature activation of the amplifiers’ thermal limiters, thereby limiting the maximum output level available to the listener.

How do I know in which direction to point the loudspeakers?

Please refer to the Acoustical Axis Definitions document.

Subwoofers are omni-directional (same output level in all directions) in their passband and so they can be pointed in any direction.

Can I setup the loudspeakers at angles different to 60° for two-channel stereo?

Listening tests conducted by Alan Blumlein back in the 1930s showed that ±30 degrees is a good compromise between stereo width and phantom center image. Wider placements widen the stereo width but reduce the solidity of the phantom center image. Narrower placements do the opposite. All two-channel listening systems and the left-right pair in multichannel systems should be set up with the loudspeakers’ acoustical axis positioned at ±30 degrees. In all cases the loudspeakers’ acoustical axis should point towards the listening position.

Can the height of the loudspeakers be different?

In two-channel listening systems and for the left-right pair in multichannel systems, the loudspeakers should be positioned at the same height to ensure that the phantom center image is as good as it can be. In multichannel systems the center loudspeaker can be positioned slightly higher (up to 7°) than the left-right pair without affecting vertical imaging when panning across the front. The rear loudspeakers in multichannel systems can be positioned as much as 15° higher than the front loudspeakers as humans have poor localization behind the head. Side loudspeakers in multichannel systems have a similar limit.

Where is the best place to position a subwoofer in a listening room?

To find the best location of subwoofer in a listening room it should ideally be measured using a calibrated acoustical measurement system. Failing this, here are some recommendations:

Single subwoofer systems…

- Slightly off-center (60 cm, 2ft) from the center of the front wall.
- In the corner – not recommended as localization is possible for extreme off-axis positions.

Multiple subwoofer systems…

- Distributed evenly along the front wall to create a Plane Wave Bass Array™ – see operating manual for details.
- One in each corner.

In all cases the subwoofer should be placed next to the wall to avoid cancellations in its passband, then the acoustical loading should be corrected using the input and output level controls. Subwoofers placed in corners may suffer from high levels of very low frequency energy, which should be corrected using the "Low-Cut" control. Distance differences compared to the main loudspeakers should be corrected using the "Phase" controls. Flush mounting is also possible to save space in the listening room.

How do I connect a subwoofer into my system?

This depends on the equipment in the system:

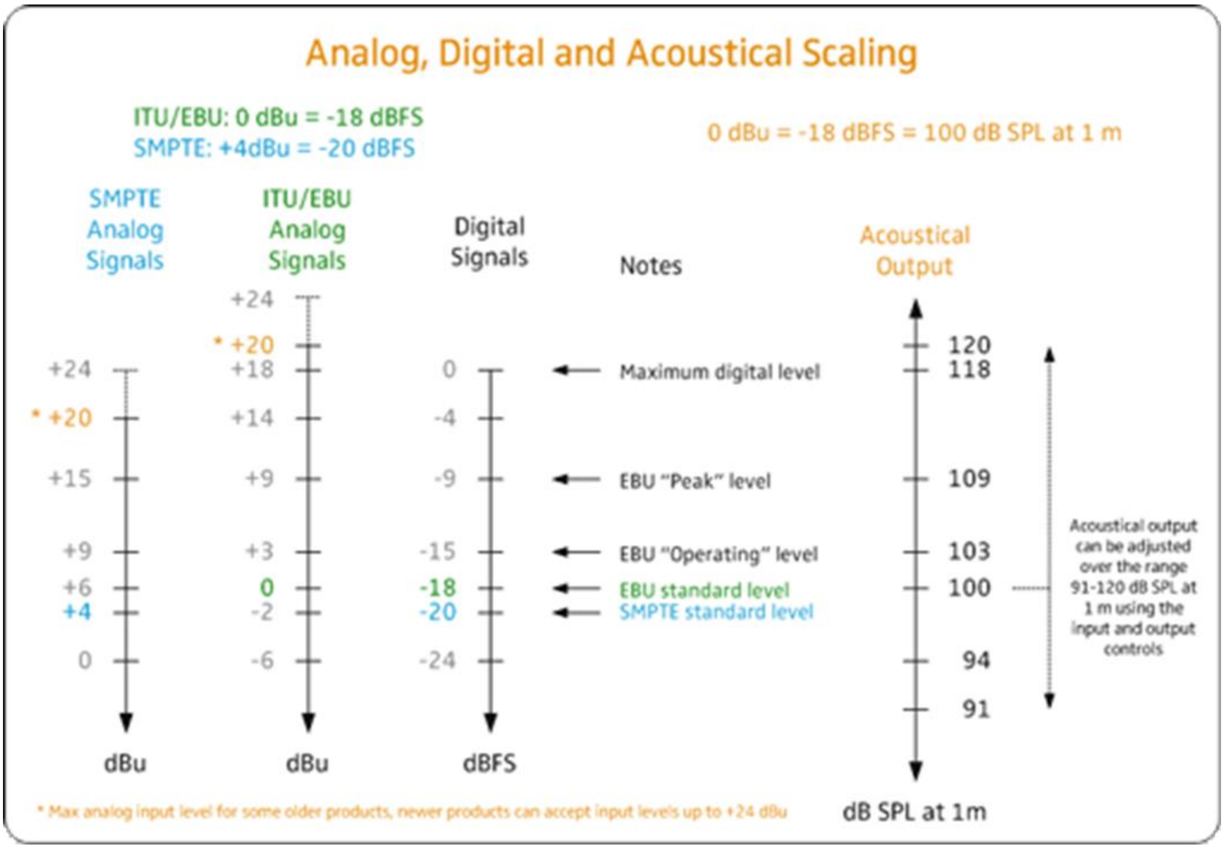
- The outputs from the source equipment should first be connected to the inputs of the Pro M 1012 and then the loudspeakers and subwoofer(s) connected to the outputs of the Pro M 1012.
- The outputs from the source equipment should first be connected to the inputs of the subwoofer's bass management system. Then the loudspeakers are connected to the outputs of the subwoofer's bass management system.
- In some cases the source signal can pass straight to the loudspeaker, for example rear channels that will not be bass managed.

How do I align the levels in a multichannel system?

To help you align the levels in a multichannel system, we have made a set of test signals and instructions.

How are analog and digital signal levels aligned to loudspeaker acoustical output?

In Europe, 0 dBu is aligned to –18 dBFS (ITU/EBU standard). In the US and Japan, +4 dBu is aligned to -20 dBFS (SMPTE standard). So the European alignment has a 9 dB lower analog signal level for the same digital scale level. This results in a higher analog headroom but at a cost of a higher noise floor. These alignments are not a function of the loudspeaker but of the source - see picture below.



Next the signal is aligned to an acoustical output of the loudspeaker that results in the required level at the listening position. Here is the complete process:

- A 0 dBu pink noise alignment signal is input to the mixing console
- The gain is then trimmed somewhere in the desk so that the metering displays –18 dBFS or –20 dBFS
- Then each loudspeaker’s input sensitivity is trimmed so that the reproduced acoustical level at the listening position with this signal is:
  - 85 dB SPL (used for the movie industry) or,
  - 79-83 dB SPL (used for the broadcast industry due to the lower replay levels at home).
    This measurement is conducted using a sound level meter set to “C-weighted” and “Slow”.

All the loudspeakers have an input attenuator (after the D-A converter in digital input versions) that allows the acoustical output to be adjusted so that it can achieve these reference levels in typical conditions with these input signals.

How do I time-align my loudspeakers?

The loudspeaker drivers are already time-aligned relative to each other as part of the design process. Once the loudspeakers are installed into the listening room, the sound from each loudspeaker cabinet should arrive at the listening position at the same time. The simplest and cheapest way to achieve this is to place the loudspeakers on a circle. If this is not possible, the loudspeakers that are closer to the listening position should be delayed to compensate for the earlier arrival time of the sound. The formula for this is:

$$t_{delay} = (d_{max} - d_{loudspeaker}) / c$$

Where:

- t_{delay} is the time delay in seconds required to compensate for the shorter listening distance of that loudspeaker
- d_{max} is the distance of the loudspeaker furthest from the listening position
- $d_{loudspeaker}$ is the distance of the loudspeaker to the listening position that needs to be delayed
- c is the speed of sound in air at 20°C = 344 m/s, or use 1120 ft/s if the measured distances are in feet, or 13440 in/s if the measured distances are in inches.

Example:

The rear channel loudspeakers are located 2.3 m from the listening position. The left, centre, and right front loudspeakers are all located 2.1 m from the listening position. The required time delay for the three front loudspeakers is:

$t_{delay} = (2.3 - 2.1) / 344 = 5.81 \times 10^{-4}$ s = 0.581 ms

To keep the adverse effects (comb filtering) of a time delay difference due to loudspeaker distance differences above 15 kHz, and thus inaudible, the delay difference must be less than 33 µs. This corresponds to a loudspeaker distance difference of 1.2 cm (0.5 inches).

Time delays are often not available on the monitor outputs of mixing consoles or have a coarse resolution in the case of home theatre decoders.

Time-alignment in subwoofers is achieved using the “Phase” control. The use of this is explained in the subwoofer’s operating manual.

To overcome these issues we have started to build in time delays to the D versions of products, such as **KH 120 D** and **KH 310 D**.

What is the difference between nearfield and midfield/main monitors?

The main differences between nearfield and midfield/main monitors are in the maximum SPL and the directivity. There are some other factors to consider but here we describe in detail only these two main factors.

Maximum SPL

A shorter distance from the monitor to the listening position requires less maximum SPL from the loudspeaker if one wants to achieve the same SPL at the listening position. This is because SPL drops at a rate of 6 dB per doubling of distance in free field conditions, with about 3–4 dB under real listening conditions in a studio. Since midfield and main monitors are often used to present music to a larger group of people (presentation of mixed material to the client to make a good impression) a higher SPL might be required at the listening position thereby creating even higher demands from loudspeaker.

An example:

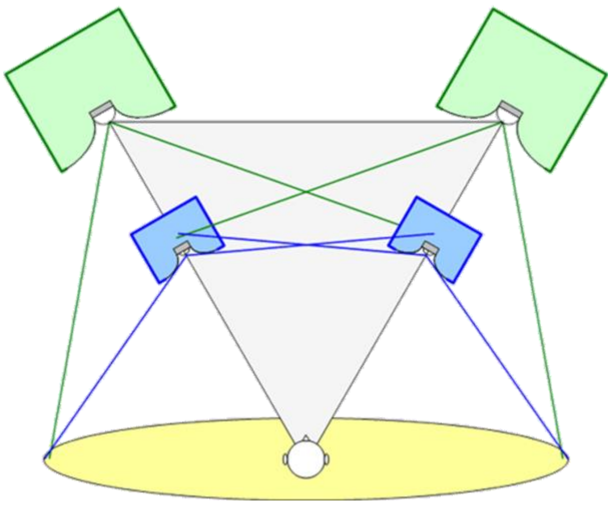
A nearfield monitor has a max SPL = 111.1 dB SPL at 1 m. At a listening distance of 1.4 m (4') the max SPL would be 108.2 dB.

A midfield monitor has a max SPL = 120.0 dB SPL at 1 m. At a listening distance of 2.2 m (6'6") the max SPL would be 113.2 dB.

This gives a headroom of 5 dB for "impressing the clients".

Directivity

Near field monitors have a shorter distance between the monitor and the listening position compared to mid field and main monitors. Even so, one still needs to have the same freedom of movement along the mixing console whilst maintaining a consistent sound quality. Therefore larger monitors, which are typically used longer distances, need to have a narrower dispersion.



The longer distance between a larger monitor and the listening position brings in more of the character of the room, i.e. modes and reflections. To control the impact of these influences on the perceived sound quality, the dispersion on larger monitors should be narrower.

Conversely, nearfield monitors are often used in cramped conditions so they need to be small. The smaller a woofer is the wider its directivity is at a particular frequency. If a tweeter and a woofer are combined to achieve a smooth and constant directivity, the tweeter also has to have a wide dispersion at the crossover frequency.

Conclusion:
Nearfield monitors:
- wide dispersion due to short listening distances
- smaller woofer gives a wider dispersion
- compact cabinet gives no localization of the different drivers even at short listening conditions
- lower max SPL

Midfield/Main monitors:
- narrow dispersion reduces unwanted negative influences from the room
- large woofer diameters lead to smaller directivity
- higher max SPL
- acoustical controls are adapted to their positioning conditions