NEUMANN.BERLIN

MEASUREMENT DESCRIPTIONS

Acoustical Controls
The acoustical controls are designed to compensate for commonly encountered issues seen in listening rooms:

Low Cut
Compensation for a lack of low-frequency damping

Bass
Acoustical loading from a wall

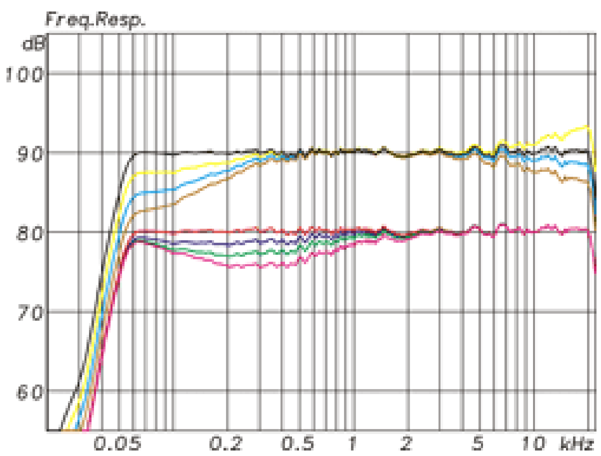
Low-mid
Acoustical loading from a desktop

LF Para EQ
Compensation for other deviations in the loudspeaker/subwoofer response

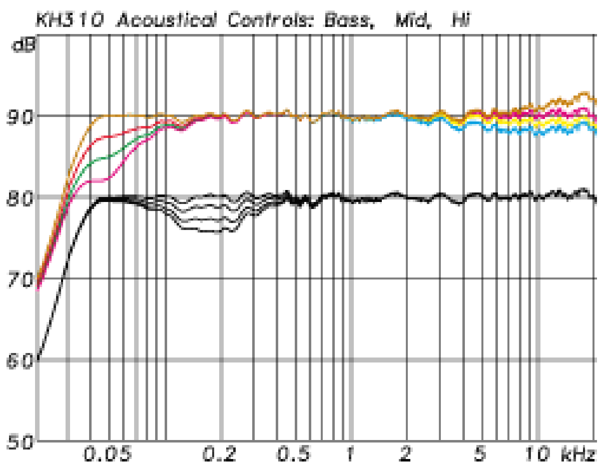
Mid
Midrange harshness caused by room acoustics

Treble
High-frequency harshness or over damping caused by room acoustics

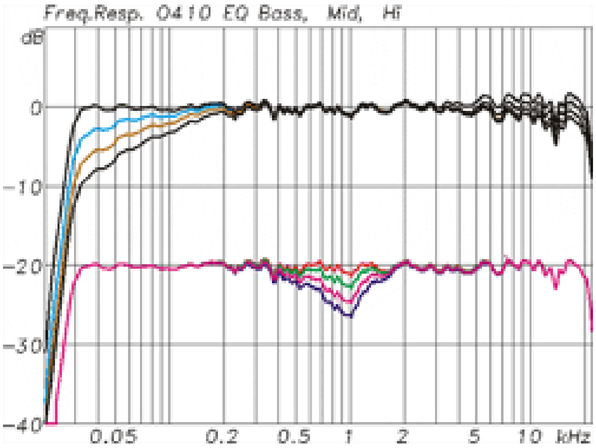
As the amount of the effect can vary, each control has four settings. Adjusting these controls can make a huge difference to the performance of the system, so read the operating manual for suggested settings depending on the loudspeaker's positioning in the room.



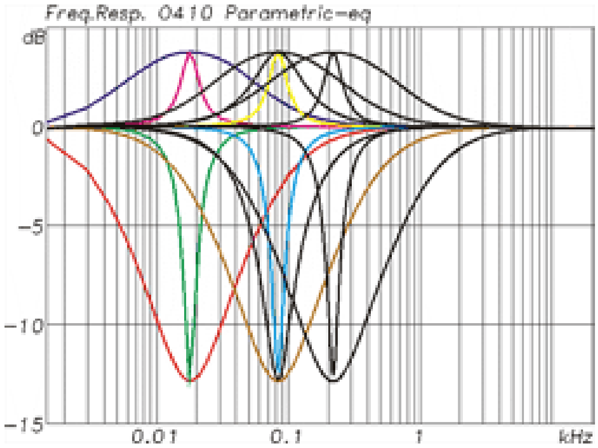
KH 120
Bass, Low-mid and Treble Controls



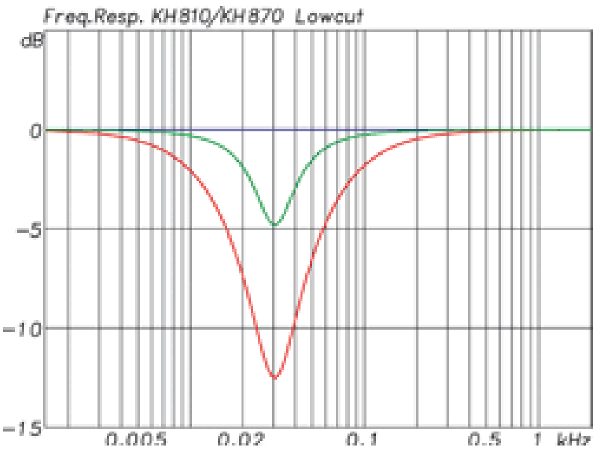
KH 310
Bass, Low-mid and Treble Controls



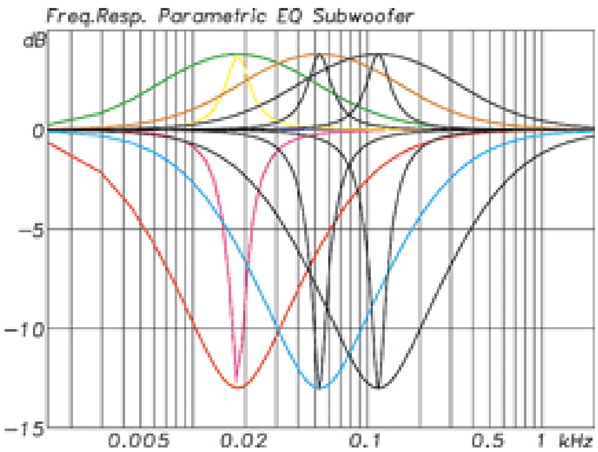
O 410
Bass, Mid and Treble Controls



O 410
Parametric Equalizer



KH 810
Low-cut Control



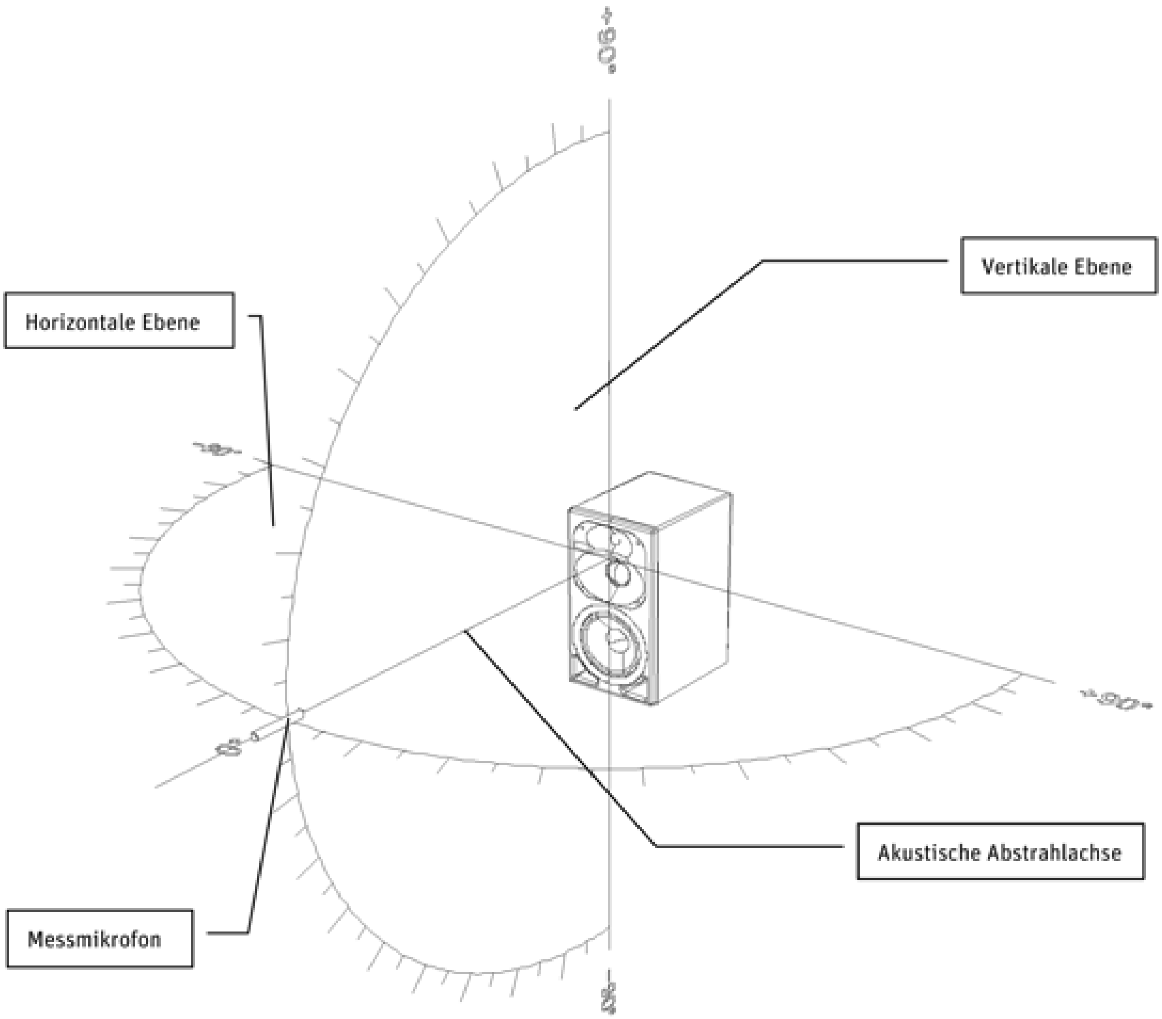
KH 810
Parametric Equalizer

Dispersion (Directivity)

Sound travels out from the loudspeaker in more than one direction. As the listener moves off axis, the level of higher frequencies reduces more than the level of lower frequencies. The reduction in level off-axis compared to on-axis depends on the size of the driver and the surface into which it is mounted (a wall, panel, or loudspeaker cabinet). The larger the combined diameter and surface, the smaller the radiation angle: lower frequency, longer wavelength, more omni-directional behavior; higher frequency, shorter wavelength, more directional behavior; midrange frequencies sit somewhere between these two extremes. It is not practical to control the dispersion of a bass driver using an acoustical horn as it would have to be very large to control the long wavelengths involved. This frequency-dependent dispersion leads to problems around the crossover frequencies. In a two-way loudspeaker system the woofer plays up to the crossover point, typically 2–3 kHz. The wavelength of this frequency is quite small in relation to the diameter of the bass driver, which is why the directivity of the bass driver is narrow in the kHz frequencies. Just above the crossover frequency the tweeter starts to radiate sound. The wavelengths of these frequencies, while short (10–20 cm), are quite large in relation to the diameter of the tweeter, and therefore the dispersion is wide. So the dispersion has changed from gradually narrowing with increased frequency in the bass driver region, to becoming wide again before narrowing with increased frequency in the treble driver region. This creates a non-smooth power response (the total sound energy radiated from the loudspeaker) which has consequences when the loudspeaker is placed into the listening room – see next paragraph. Even if the on-axis response is flat, the loudspeaker can have a tonality which is more apparent with decreased quality of acoustical treatment in the room. Neumann avoids this off-axis problem by using waveguides (acoustical horns) in front of the midrange driver and tweeters.

Moving onto the sound in the room, even if the listener is positioned on-axis, the total sound at the listening position is the sum of the direct on-axis sound plus the reflected off-axis sound. Firstly, the direct sound should be a flat as possible. Secondly, the off-axis sound is reflected by the surfaces in the room (equipment, furniture, and walls). This reflected sound will be colored in some way and attenuated in level due the reflecting surfaces’ acoustical properties, although it is possible with good acoustical design to minimize the adverse affects of this. The off-axis sound from the loudspeaker should not be colored otherwise the acoustics of the room will have to cope with this additional sound quality factor, which is not practical nor even possible in some cases. Finally, if there is a non-linear reverberation time in the room, the result at the listening position will also be colored.

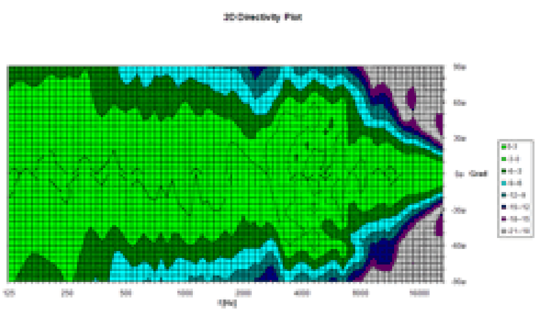
To measure the way in which sound radiates, many measurements are required around the loudspeaker. To avoid the need to measure every angle in all directions, typical just the horizontal and vertical planes are measured with a resolution of 5 deg. From this data, interpolation can be used to derive frequency responses of the angles that were not measured. Furthermore, symmetry can be used to further reduce the number of measurements required.



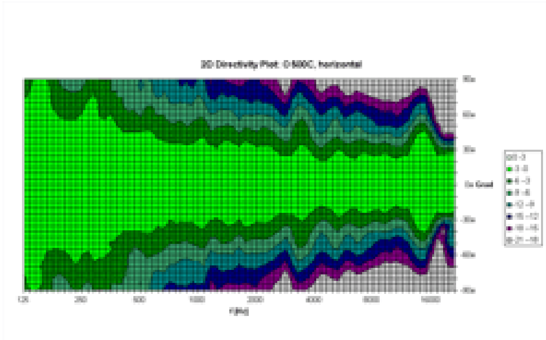
Anordnung zur Bestimmung des Abstrahlverhaltens

From the measured data many different pictures can be drawn: dispersion plot, directivity balloon, isobars, polar plot, directivity index/factor, power response, etc. Additional post processing is often applied to the data: frequency smoothing, level normalization, etc.

Dispersion plots display frequency the horizontal axis and angle on the vertical axis. The level of the sound at a particular frequency and angle is displayed using different colors. Below are examples of a l arge three-w ay lou dspeaker without any dispersion control (drivers mounted directly to the front panel) and a large three-way loudspeaker fitted with a waveguide. At lower frequencies the widening of the dispersion can be seen in both loudspeakers. At higher frequencies the lack of dispersion control is clearly visible in the 3-6 kHz region (left plot). The off-axis sound of this loudspeaker will be quite colored and so the perceived sound at the listening position will also be colored. And the higher the reverberation time, the higher the audible coloration will be.



A loudspeaker with no dispersion control

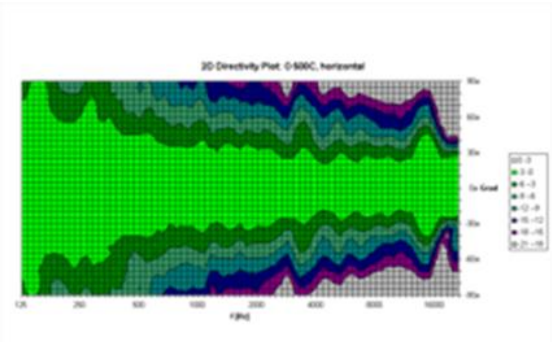


A loudspeaker with dispersion control (O 500 C)

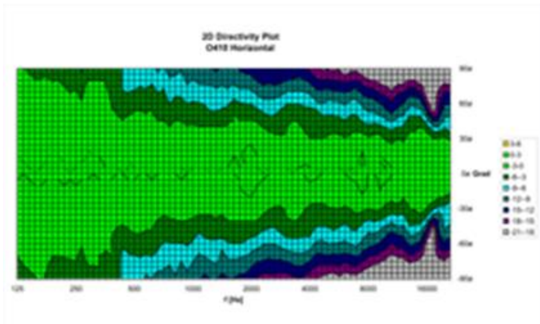
In the loudspeaker without dispersion control, the frequency region 1-7 kHz region can be seen to be very wide so in rooms with less than ideal acoustics, the loudspeaker will sound colored. Additionally, the high frequency collapse in dispersion creates a narrow sweet spot in that frequency range.

In the past waveguides were designing by making a physical prototype out of clay and then checking it using anechoic measurements. Incorrect results meant remaking the physical prototype and then taking a new set of measurements. These days, the waveguides in Neumann loudspeakers are iteratively designed in a computer using acoustical models until the target is achieved. Then only a verification of the final design needs to be performed with an anechoic measurement (Mathematically Modeled Dispersion™ waveguide, MMD™). This is analogous to way that automotive aerodynamics were, and are now, designed using computers, clay models, and wind tunnels.

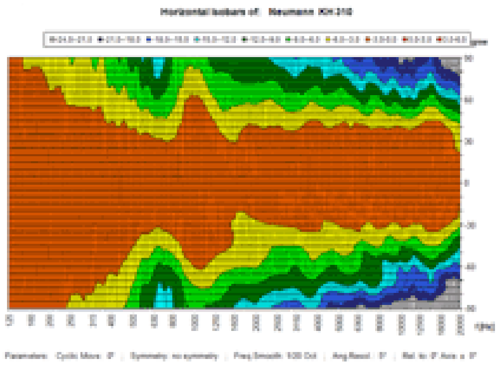
The aim of the waveguide is to provide a constant directivity throughout the crossover region(s) and above. Ideally all models should have the same directivity, or same character of the directivity, so that products can be mixed and matched to make suitably specified sound reproduction systems. Below are four examples from the Neumann range.



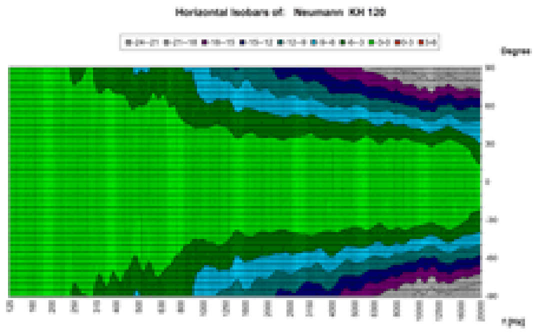
O 500



O 410



KH 310

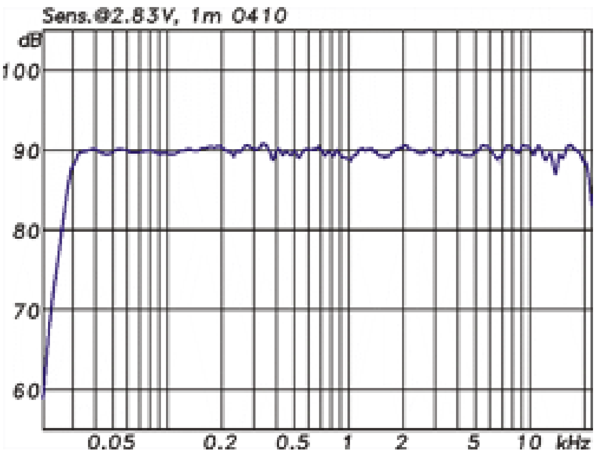


KH 120

Frequency Response

When a Fourier transformation is taken of a time series the result is a complex frequency response. In the case of loudspeakers, the time series to be transformed is the impulse response. The impulse response is the response of the loudspeaker to an impulse input (a very short and loud sound – like a handgun or balloon burst). The impulse response gives a complete description of a linear time-invariant system (in this case, a loudspeaker playing at moderate levels). In acoustics, a complex frequency response is not very useful in its raw form, so the magnitude and phase responses are calculated to give the "magnitude of the frequency response" and "phase of the frequency response". These are normally abbreviated simply to "frequency response" and "phase response". Finally, the magnitude is normally converted to decibels as our perception of sound pressure is logarithmic and as a consequence easier to interpret on a log scale.

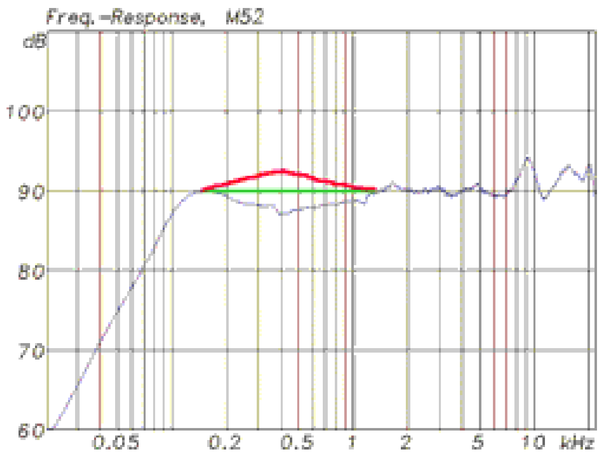
Ideally, an accurate loudspeaker should have a flat magnitude of the frequency response in anechoic conditions across the entire audible frequency range (20 – 20k Hz). Technically this is impossible to achieve but we can get close. There will be a low frequency roll-off that limits bass extension. There will also be some deviation away from a perfectly flat response - this deviation should be minimized. Below is an example of a very good loudspeaker:



Frequency response of a large three-way loudspeaker (O 410)

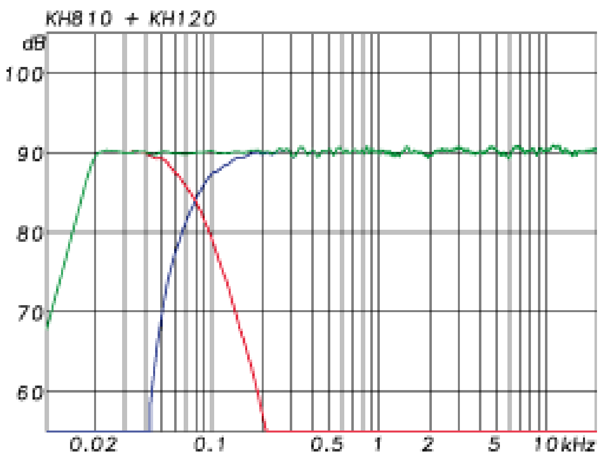
Less capable loudspeakers would have less bass extension and a greater deviation of the frequency response away from flat. Examples of the causes of deviations are poor low-frequency alignment, cabinet and port resonances, crossover alignment, edge diffraction, and inappropriate driver selection.

Sometimes the frequency response is made to be deliberately non-flat in anechoic conditions. This occurs when the typical loudspeaker positioning is known, and there is no possibility for the user to adjust the response on the loudspeaker. An example is the M 52 that is designed to be located next to a mixing console which would increase (red curve) the deliberately lower level in the 200 – 1000 Hz region. The result is a flat response in the listening room (green curve).

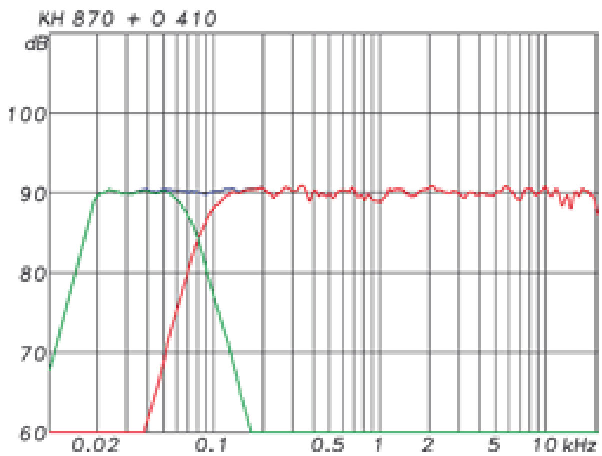


Frequency response compensation due to desktop loading

The low-frequency extension of a loudspeaker can be extended by using a subwoofer. This brings some additional advantages. Some of them are: increased SPL, reduced distortion from the main loudspeakers, and a decrease of the frequencies where group delay increases. There is one disadvantage of increased group delay around the crossover frequency.



Small two-way loudspeaker with a subwoofer



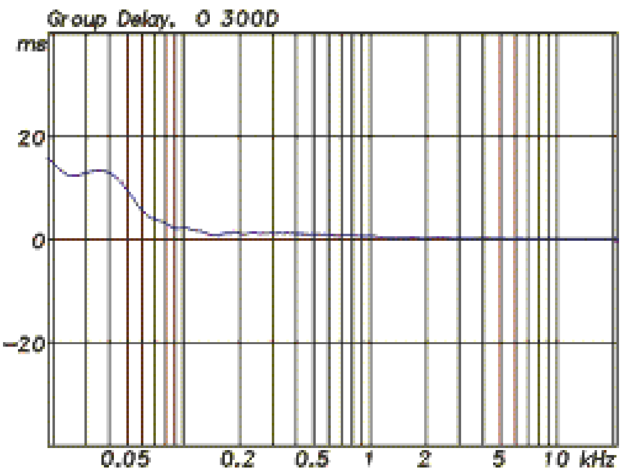
Large three-way loudspeaker with a subwoofer

Group delay

When a Fourier transformation is taken of a time series the result is a complex frequency response. In the case of loudspeakers, the time series to be transformed is the impulse response. The impulse response is the response of the loudspeaker to an impulse input (a very short and loud sound – like a handgun or balloon burst – also known as a “Dirac”). The impulse response gives a complete description of a linear time-invariant system (in this case, a loudspeaker playing at moderate levels). In acoustics, a complex frequency response is not very useful in its raw form, so the magnitude and phase responses are calculated to give the “magnitude of the frequency response” and “phase of the frequency response”. These are normally abbreviated simply to “frequency response” and “phase response”.

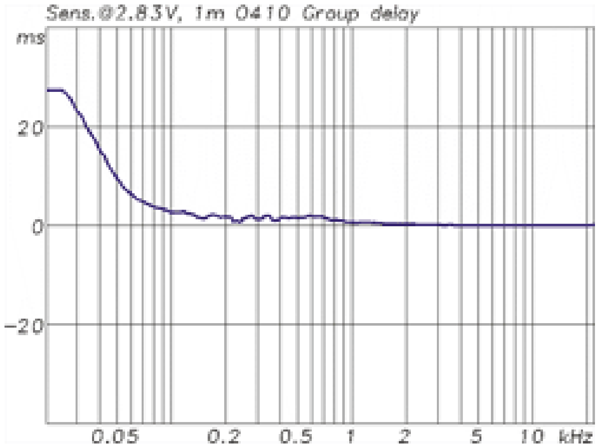
The phase response can be further processed to give the group delay (it is the negative slope of the phase response, $-d\phi(\omega)/d\omega$). Group delay is the time it takes for the electrical input to pass through the loudspeaker system and become an acoustical output. Ideally group delay should be zero at all frequencies, i.e. all of the signal takes the same amount of time to pass through the loudspeaker, and that this time is a minimum. In practice, group delay increases with reduced frequency due to electronic (infrasonic) protection filters and the natural roll-off seen in vented cabinets (equivalent to a 4th order filter) or sealed cabinets (equivalent to a 2nd order filter). The higher the order of the filter, the higher the group delay. The lower the corner frequency of the filter, the higher the group delay.

In this example the group delay increases towards low frequencies but psychoacoustic tests show that the values indicated are on, or just below, the threshold of audibility.



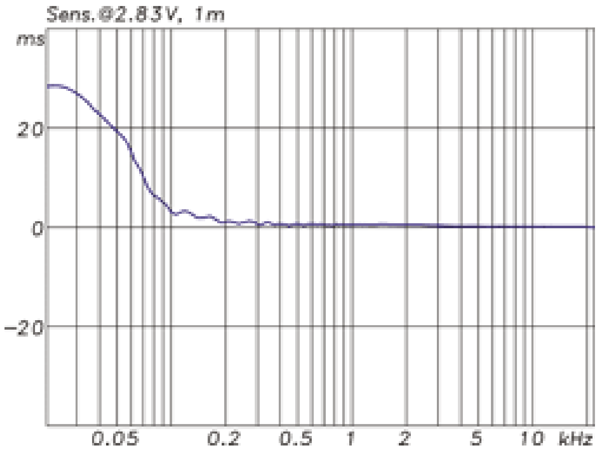
Group delay of a sealed compact three-way loudspeaker

In the example below, a larger loudspeaker appears to have a higher group delay. However over the same frequency range as the example above (>40 Hz) it can be seen to be the same. This is because the low-frequency cut-off of the loudspeaker is lower. This large loudspeaker happens to be vented but the result is a loudspeaker with the same time-domain response as a compact sealed enclosure within the frequency range of the smaller monitor.

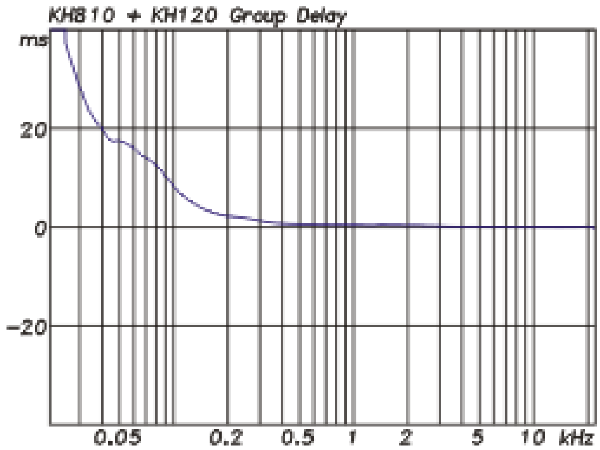


Group delay of a large three-way loudspeaker

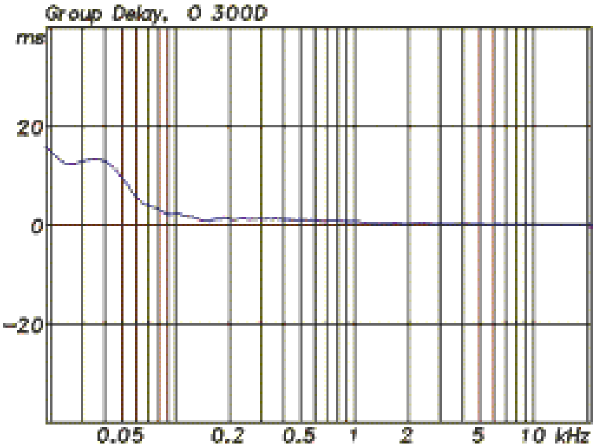
Adding a subwoofer extends the low-frequency extension of the loudspeaker and increases the maximum SPL of the system. There are also changes to the group delay in the form of a decrease of the frequencies where group delay is high due to the lower corner frequency (this subjectively beneficial), and increased group delay around the crossover frequency due to the crossover (this is likely to be audible in a good room).



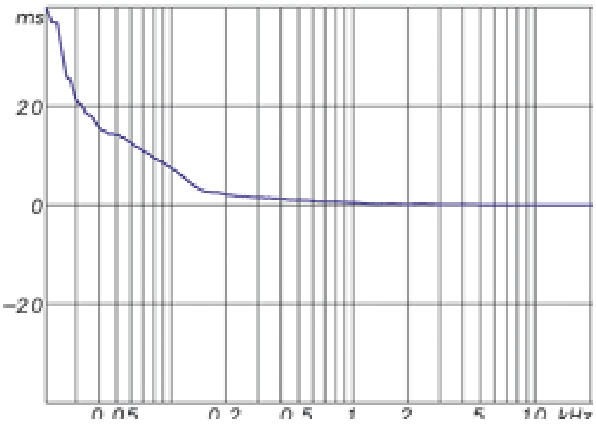
Small two-way loudspeaker KH 120



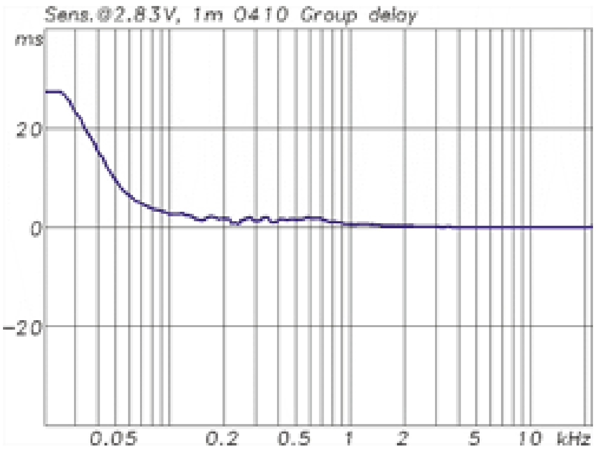
Small two-way loudspeaker KH 120
with a subwoofer KH 810



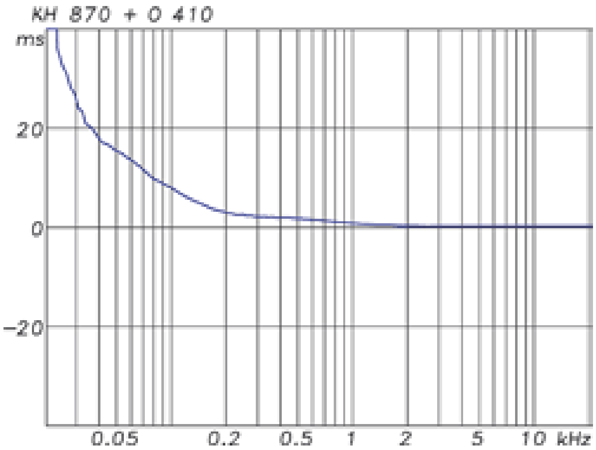
Compact sealed three-way loudspeaker O 300 D



Compact sealed three-way loudspeaker O 300 D with a subwoofer KH 810

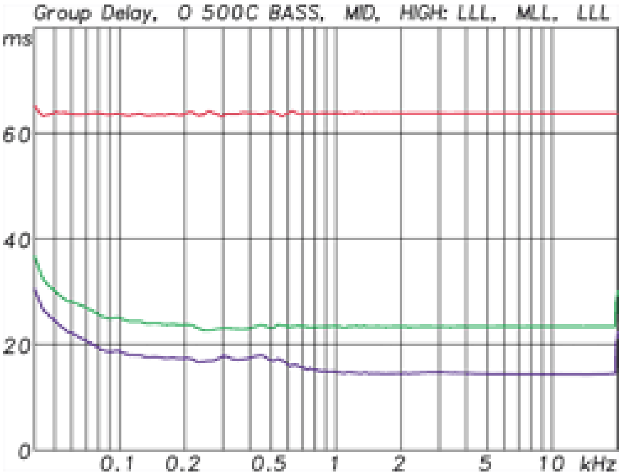


Large vented three-way loudspeaker O 410



Large vented three-way loudspeaker O 410 with a subwoofer KH 870

To eliminate frequency-dependent group delay (blue curve below), a frequency-dependent time delay can be added. The result is what is known as a linear-phase system – all frequencies pass through the system in the same amount of time (red curve). Whilst a linear-phase loudspeaker sounds incredible, there is a cost in the form of latency. Another possibility is to partially correct the group delay, in this case down to about 100 Hz (green curve). This has the advantage of correcting the just the upper part of the response while keeping latency down to an acceptable value.



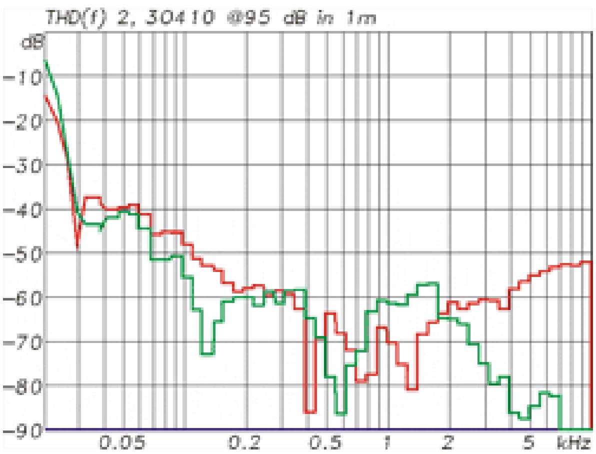
Group delay modes of a DSP loudspeaker: linear (LLL),
mixed (MLL), minimum (MMM)

Harmonic Distortion

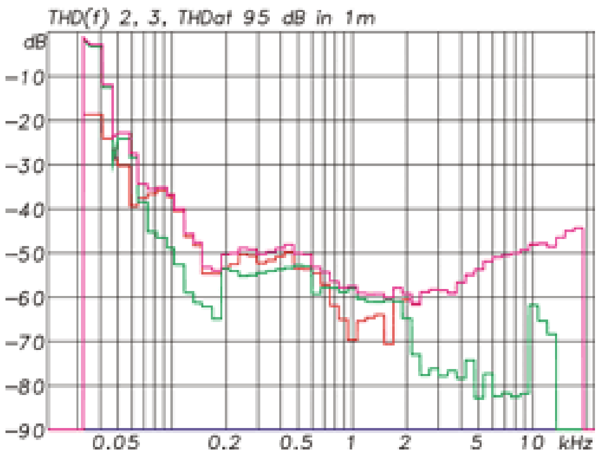
Loudspeakers suffer from linear and non-linear distortion. Linear distortion can be seen as a non-flat frequency response and/or a non-flat group delay. This is discussed in other sections. Non-linear distortion adds new frequencies to the acoustical output that were not present in the electrical input signal. To measure this, a frequency (the fundamental, say 100 Hz) is played into the loudspeaker at a particular sound pressure level and a frequency above that test frequency is measured. Double the frequency is the second harmonic (200 Hz), triple the frequency is the third harmonic (300 Hz), etc. Sweeping the fundamental frequency allows a graph of frequency-dependent harmonic distortion to be plotted. Total harmonic distortion (THD) is the relation between all sound coming out of the loudspeaker compared to all the additional sound in the output that was not present at the input: second + third + fourth + fifth + etc.

Harmonic distortion can be expressed in decibels or as a percentage. Second-order harmonic distortion is generally caused by asymmetries in the system. Third-order harmonic distortion is generally caused by “clipping” in the system, and this can come from the electronics or the acoustics, for example short voice coils. Odd-order harmonics generally sound a lot worse than even-order harmonics. Higher order harmonics will be lower than the second- and third-order harmonics, and should be at reasonably low levels in a well-designed system. Ideally the lower the harmonic distortion, the cleaner, or more transparent, the loudspeaker will sound. Less than -30 dB (3%) at low frequency and less than -40 dB (1%) at mid-high frequencies is normally considered to be good, lower values than these is of course better.

Harmonic distortion is non-linear with level, in that an increase of 10 dB in the test signal typically results in a far greater increase in the level of harmonic distortion. As a result, one should check the test conditions before comparing measurements of different loudspeakers. In general, larger loudspeakers suffer from less harmonic distortion than smaller loudspeakers when played at the same level. Additionally, three-way loudspeakers will suffer from less harmonic distortion than two-way loudspeakers as each driver has less work to do. Both these effects can be seen together in the two examples below which are tested at the same sound pressure level.



Großer 3-Wege-Lautsprecher O 410



Kleiner 2-Wege-Lautsprecher KH 120

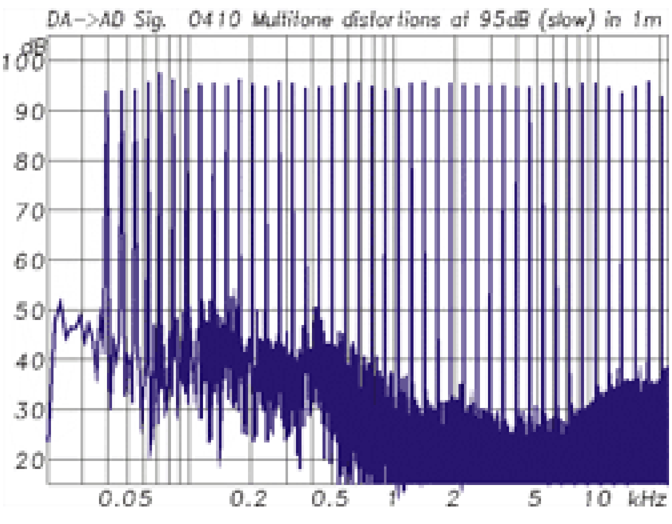
Unfortunately, the level of harmonic distortion does not correlate well with subjective sound quality, for example, an audio system with high levels of second order harmonic distortion can sound quite pleasant, whilst the same system with same level of third order harmonic distortion would sound rather poor. However, harmonic distortion plots are very useful for design engineers to use as a tool to trace problems in their designs.

Note also that even if second order harmonics created by a monitoring system might sound good (like a tube distortion, exciter, compression, etc.) the signal is changed by this distortion. The acoustic output therefore differs from the electrical input signal, which must be avoided as much as possible.

Intermodulation Distortion

Loudspeakers suffer from linear and non-linear distortion. Linear distortion can be seen as a non-flat frequency response and/or a non-flat group delay. This is discussed in other sections. Non-linear distortion in the loudspeaker system adds new frequencies to the acoustical output that were not present in the electrical input signal. One measure of this is harmonic distortion, where a single frequency is played into the loudspeaker and the level of each harmonic measured. Another measure of distortion is to play a more complex signal into the loudspeaker and measure the additional frequencies in the output. The problem with this is that there are no internationally agreed standards for the test signal so comparison of different measurements is impossible unless the test conditions are identical. Since an intermodulation distortion measurement test signal consists of many discrete tones played at the same time, this type of broadband measurement signal is representative of real world usage of the system unlike the pure sine tones used for THD measurements.

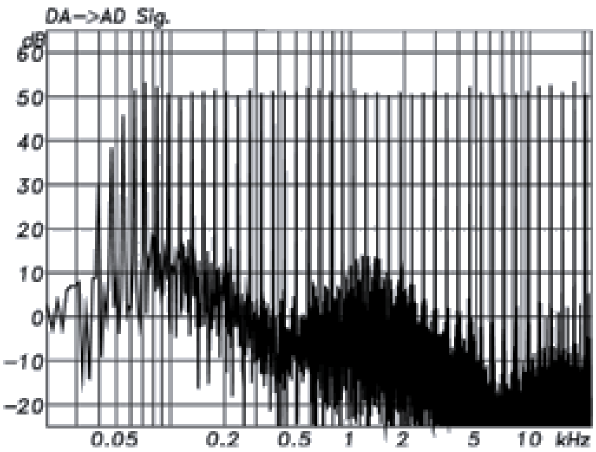
The test signal is commonly known as a multi-tone because it consists of a collection of tones. It sounds like someone leaning on a church organ with both forearms. In the example below there are 43 tones logarithmically distributed between 40 and 20k Hz. All the detail on the graph between these tones is the intermodulation and harmonic distortion, and this is what should be minimized. Like harmonic distortion, a lower intermodulation distortion results in a more transparent, cleaner sound quality. Unlike harmonic distortion, intermodulation distortion levels do correlate well with perceived sound quality, the lower the better.



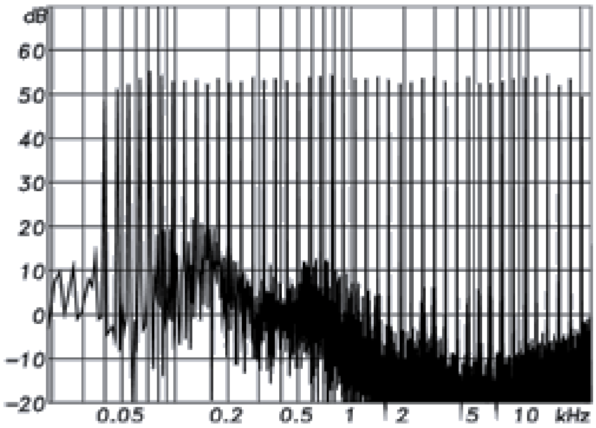
Intermodulation distortion of a large three-way loudspeaker

Intermodulation distortion is non-linear with level, in that an increase of 10 dB in the test signal typically results in a far greater increase in level in the intermodulation distortion. As a result one should check the test conditions before comparing measurements of different loudspeakers. In general, larger loudspeakers suffer from less intermodulation distortion than smaller loudspeakers when played at the same level. Additionally, three-way loudspeakers will suffer from less intermodulation distortion than two-way loudspeakers as each driver has less work to do.

In the graphs below, one can see the advantage of introducing a midrange driver. The midrange intermodulation distortion is reduced by 10–15 dB.

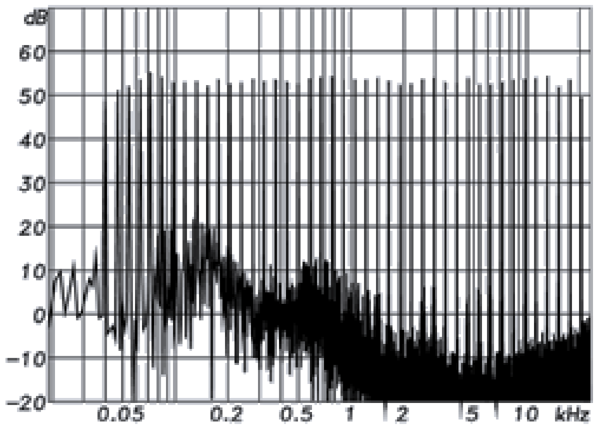


Intermodulation distortion of
a two-way loudspeaker

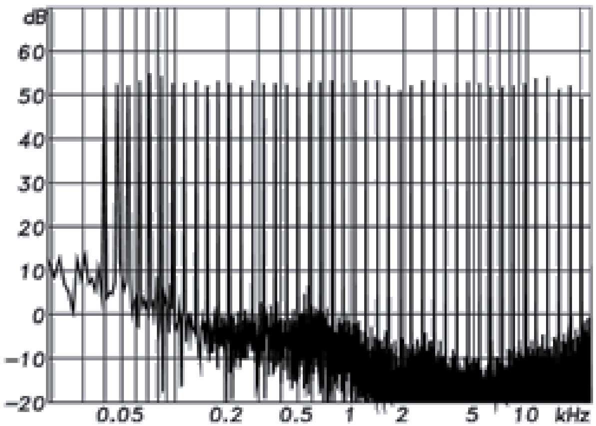


Intermodulation distortion of
a three-way loudspeaker

In the graphs below, one can see the advantage of adding a subwoofer to the three-way system. The low-frequency intermodulation distortion is reduced by 10-15 dB, and the midrange intermodulation distortion is reduced by about 10 dB. Additionally, Doppler distortion seen in the 2-6kHz region is removed when a subwoofer is added. This is because the bass driver no longer modulates those frequencies as its excursion has been reduced. This is audible as a cleaning up of the high-frequency region. The result is an audio transparency from a compact three-way / subwoofer system that is comparable to that of a larger three-way system.



Intermodulation distortion of
a three-way loudspeaker



Intermodulation distortion of
a three-way loudspeaker with a subwoofer

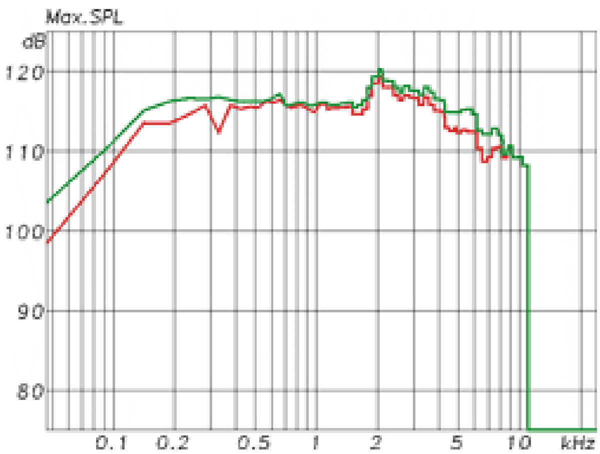
Maximum Sound Pressure Level (SPL)

Typically maximum sound pressure level (SPL) figures are quite simply specified as a number. This means nothing unless the measurement conditions are known: frequency range, test signal, measurement type, distance of the microphone to the loudspeaker, duration of the signal, location of the loudspeaker, etc. As an example, here is the specification for the O 410:

Maximum sound pressure level in half space at 3% THD at 1m averaged between 100 Hz and 6 kHz = 120.0 dB.

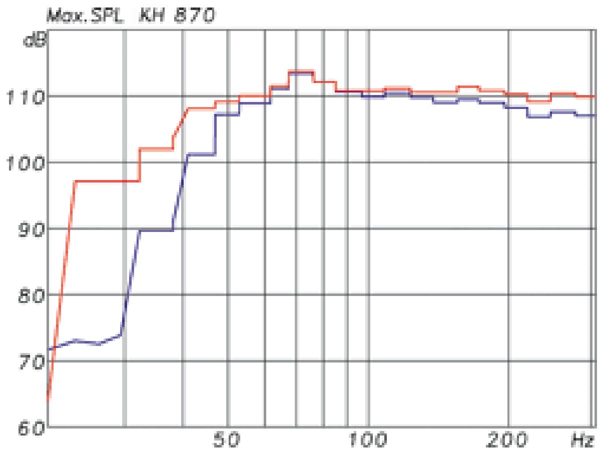
Changing any of the measurement conditions will give a different value. To measure the maximum SPL, a sine wave is played into the loudspeaker and it is increased until the distortion reaches some value, say 1% or 3%, and then the SPL is noted. Sweeping the sine wave's frequency allows a graph to be plotted or an average to be calculated. Alternatively a broadband signal can be used, for example, pink noise, IEC-weighted noise, or "program material" (whatever that means!).

As with most aspects of acoustics, maximum SPL is frequency-dependent. Lower frequencies are harder to reproduce at high levels as the displacement of the drivers must be high (four times the displacement required with each halving of frequency). This is seen as a drop in maximum SPL with reduced frequency.

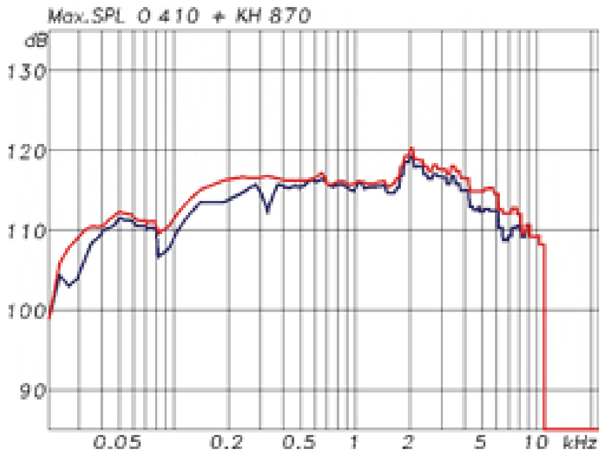


O 410 Harmonic Distortion at 95 dB SPL
(Red: 2nd harmonic, Green: 3rd harmonic)

The loss of headroom at low frequencies can be made up by adding a subwoofer to the system:



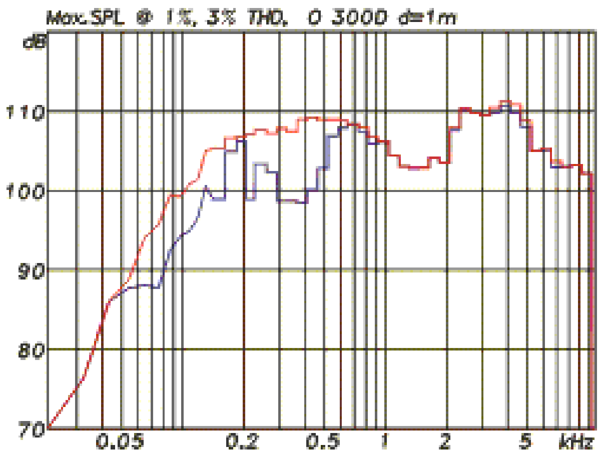
KH 870 Maximum SPL at 1 m
(Red: 3% THD, Blue: 1% THD)



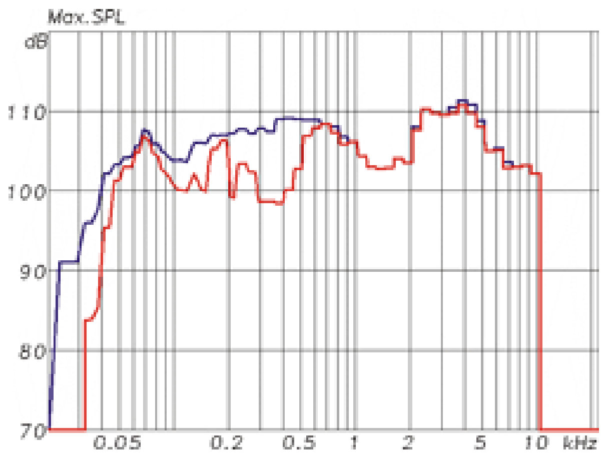
O 410 with KH 870 Maximum SPL at 1 m
(Red: 3% THD, Blue: 1% THD)

Note these measurements are for a subwoofer located in half-space. Subwoofers are typically located in quarter-space and so the subwoofer part of the curves (below about 100 Hz) would be 6dB higher.

Adding a subwoofer also brings a decrease in the low-frequency cut-off of the system, reduces harmonic and intermodulation distortions (even in the midrange), and changes to the group delay. Adding a subwoofer to a smaller loudspeaker gives a more dramatic improvement to the maximum SPL.



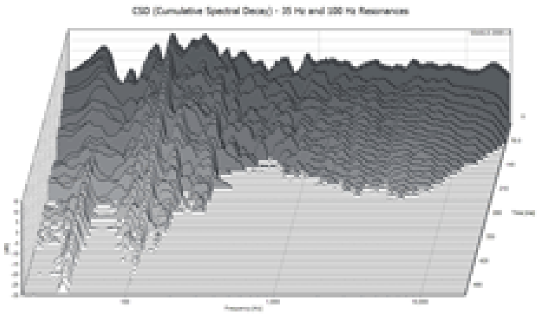
O 300 Maximum SPL at 1 m
(Red: 3% THD, Blue: 1% THD)



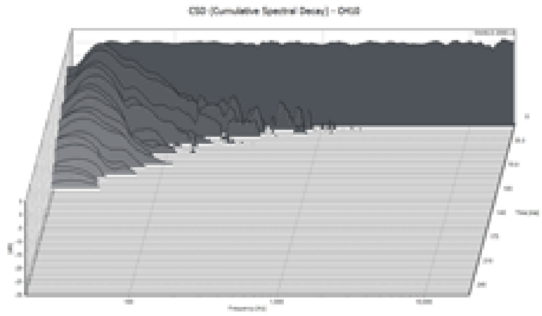
O 300 with KH 810 Maximum SPL at
1 m (Red: 3% THD, Blue: 1% THD)

Waterfall Plots

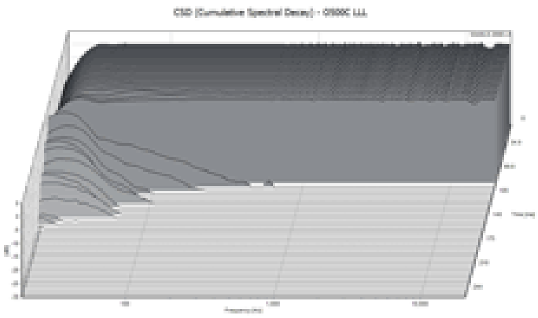
A waterfall plot is a set of frequency responses that are plotted onto a 3D axis. Each frequency response occurs a little bit later in time so that a picture is built up of how the loudspeaker behaves once the sound is turned off. Resonances (left plot) are easy to see in the form of decaying ridges extending towards the front of the plot. Resonances can occur in loudspeakers and in rooms. In both cases these resonances should be minimized. Loudspeaker resonances are reduced by careful design. Room resonances are reduced by adding acoustical treatments to the room.



Waterfall plot of a room with a strong resonance at 100 Hz



Waterfall plot of a loudspeaker with no resonances



Waterfall plot of a linear-phase DSP loudspeaker