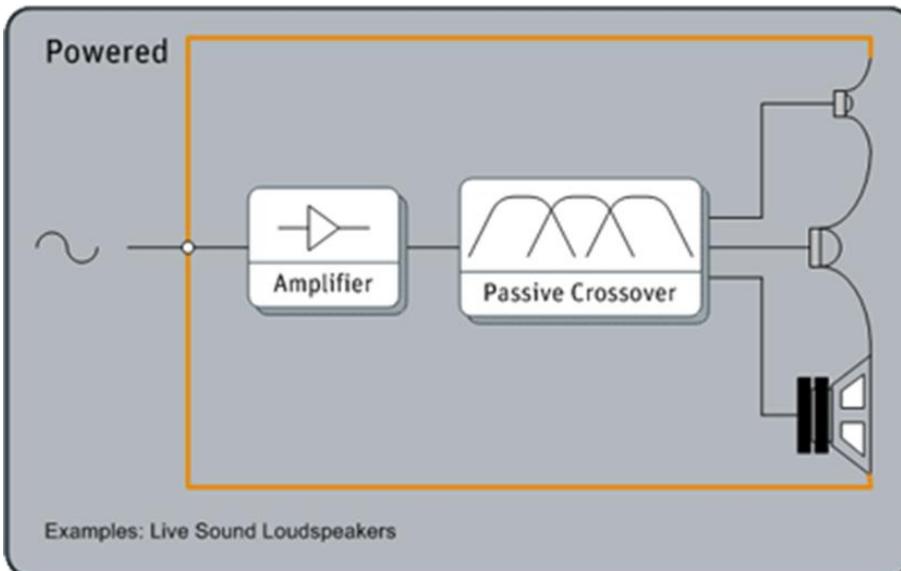
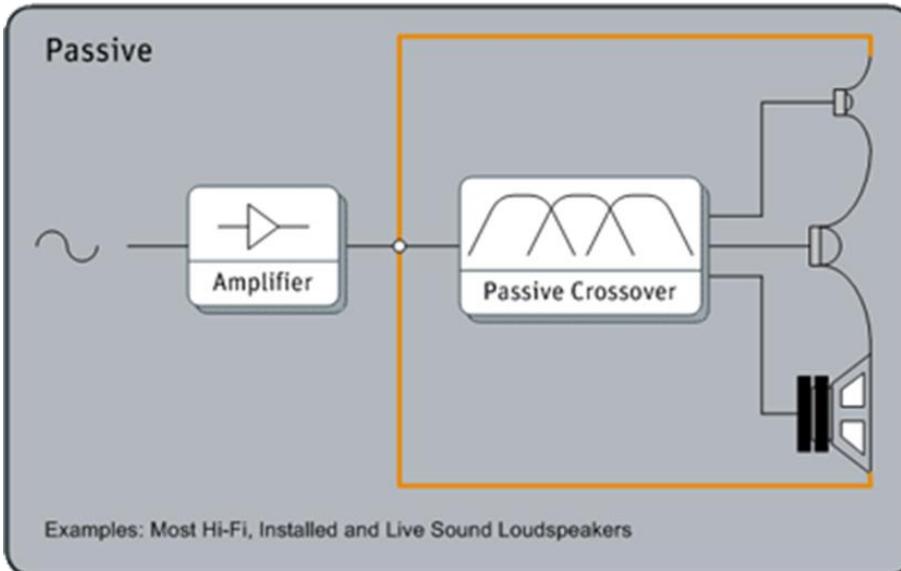


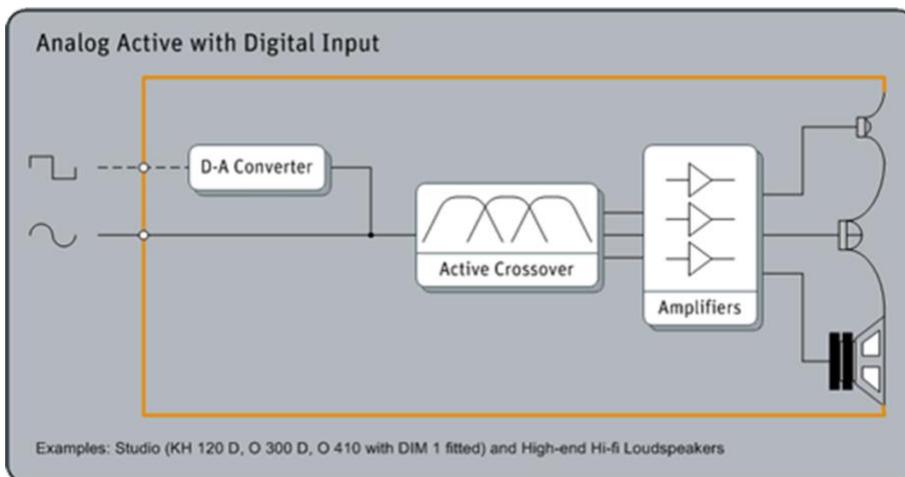
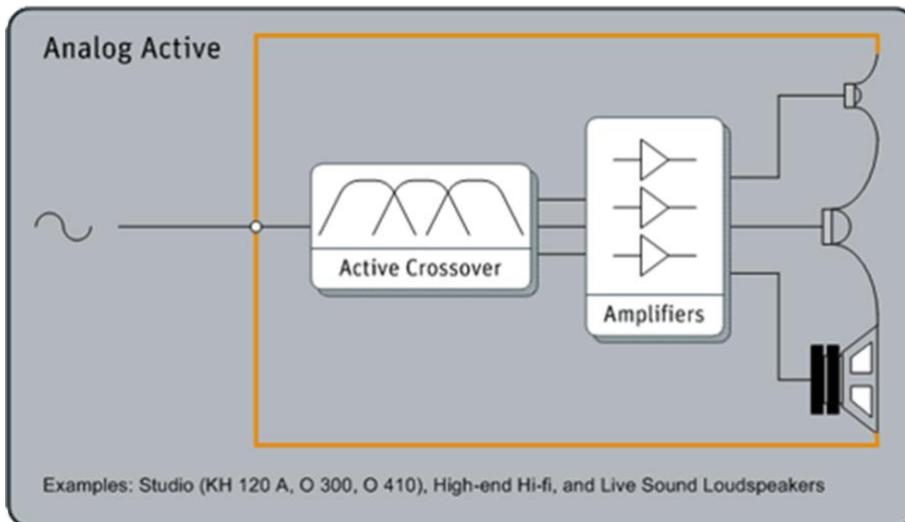
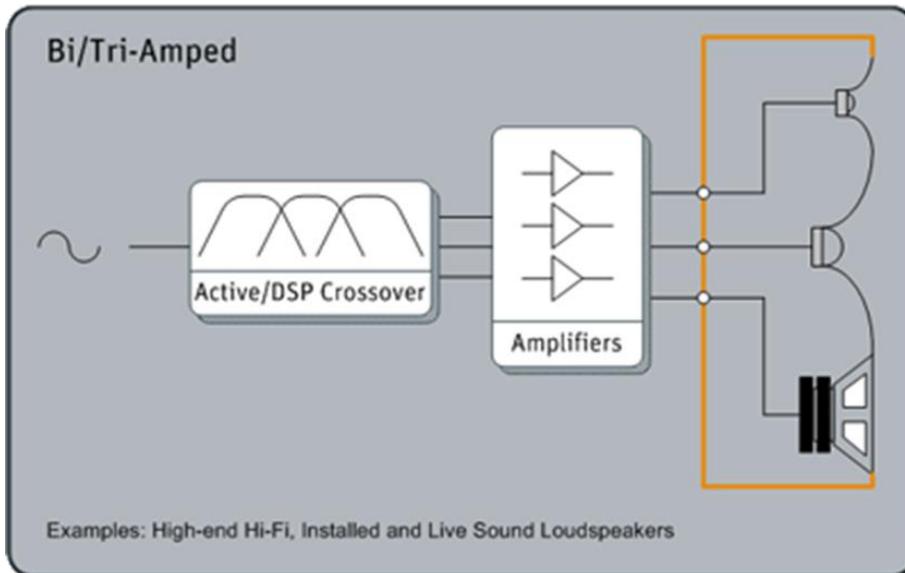


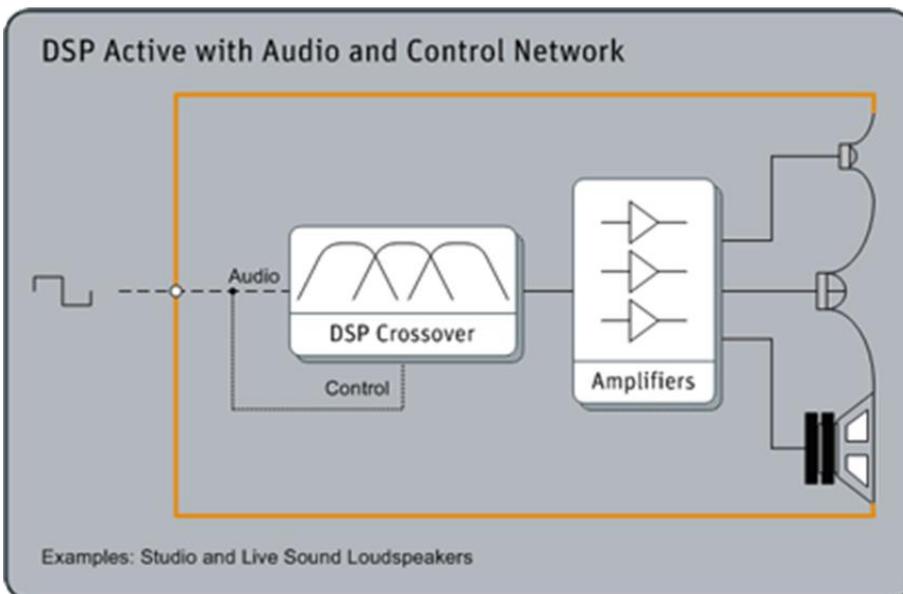
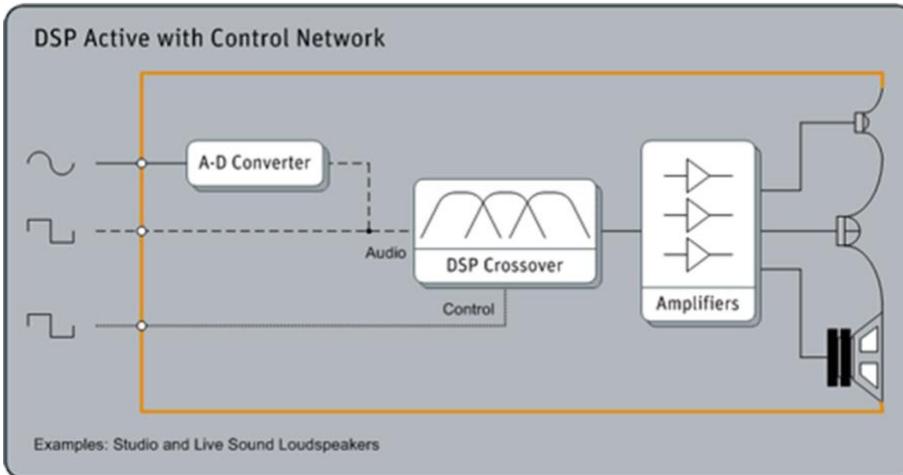
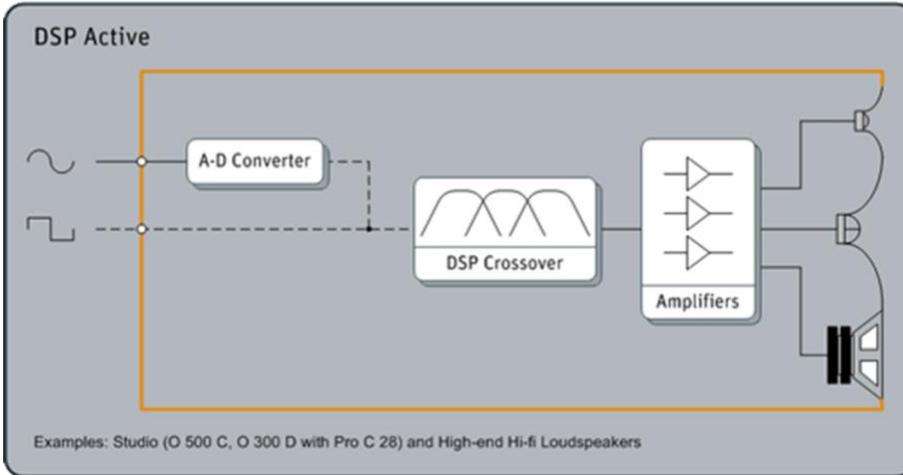
## GENERAL PRODUCT ANSWERS

**What types of loudspeaker are there (active, passive, DSP, etc.) and what technologies are used in them?**

The following basic types of loudspeaker design are commonly manufactured (Neumann does not make all of these types):









The following technologies are commonly seen in the different types of loudspeaker designs on the market (Neumann does not use all of these technologies. Also included are some less common techniques):

## **Inputs**

- Analog: electronic-balanced, transformer-balanced, loudspeaker level: low impedance, high impedance (70 V, 100 V)
- Digital: S/P-DIF, AES3, Firewire, USB, audio network

## **Converters**

- D-A: typically Oversampling or Interpolating used
- A-D: typically Sigma-Delta used
- SRC added if fixed-rate processing follows

## **Crossover/Processing**

- Passive: 1st, 2nd, 3rd order
- Active: 2nd, 3rd, 4th order
- DSP: IIR with 4th, 6th, 8th order, FIR with any order up to 16, linear phase, non-linear compensation

## **Amplifiers**

- Class A, B, AB, C, D, H

## **Power Supplies**

- Fixed linear (transformer), switchable linear (transformer), universal (switched-mode)

## **Protection**

- Passive: multifuse, soffite, relay + network, fuse, bimetal
- Active: inputs, amplifiers (thermal, short-circuit), drivers thermal limiting, excursion limiting
- DSP: look ahead limiters, power supply monitoring

## **Drivers**

- High frequency (tweeter, top): soft dome, hard dome (aluminum, titanium, ceramic, diamond), ribbon, folded ribbon, compression, plasma.
- Midrange (squaker): soft dome, hard dome, hard cone, folded ribbon, compression.
- Low frequency (bass, woofer, subwoofer): paper, polypropylene, woven fibreglass, carbon fibre, Kevlar, aluminium.
- Drivers can be direct radiating or loaded (acoustical horn)

## **Cabinet**

- Wood, plastic, metal, Stone
- Sealed, bass reflex, band pass (with different orders), horns, push-pull (isobaric)



## What are we trying to achieve when designing a loudspeaker?

At the listening position, a loudspeaker should acoustically exactly reproduce the electrical input signal. Ideally the loudspeaker should have or be (in no particular order of importance):

- Flat free-field frequency response (magnitude) with opportunity for adjustment, or a response optimized for a known placement conditions (i.e. not flat free-field frequency response)
- Deep low-frequency cut-off
- No self-generated noise
- No harmonic distortion
- No intermodulation distortion
- No latency
- A flat group delay
- No resonances
- Control of the directivity
- A peak SPL suitable for the application
- Mechanically robust and mountable into the installation
- Sufficient heat management to work in a wide variety of ambient temperatures
- A suitable size and weight for the application
- Accept the given input signal format using an appropriate connector
- Self-protecting from high input signals
- No missing or under-specified features for the application
- A suitable range of mounting hardware
- A complementary set of accessories for those that need additional features and facilities

Not all of these are possible in a single loudspeaker so trade-offs are required between the parameters. Appropriate balancing of parameters can make the difference between a good and a bad loudspeaker for a particular application. In addition, the classic engineering challenge is to make a solution for a problem using the available materials within a defined cost.

## Which products should I use for my application?

Please refer to the [Product Selection Guide](#).

## Do you make specific products for different listening materials?

Our loudspeakers are all designed to have the same neutral sound quality. However, engineering trade-offs are required between design parameters which creates performance differences between products. For example small loudspeakers generally have a lower maximum SPL and a limited bass extension. Conversely, a larger loudspeaker will generally have a higher maximum SPL, a deeper bass extension and a lower distortion for a given replay level. How does this affect loudspeaker choice?

- If you listen loud, you need a larger loudspeaker.
- If you listen to bass heavy material, for example hip-hop, dance, or synthesized music, you need a loudspeaker with a deeper bass extension and sufficiently high maximum SPL.



- If to listen to material with a large dynamic range, for example orchestral, you need a loudspeaker system with a high maximum SPL.
- If you watch action movies, you need a system capable of reproducing audio down to 20 Hz.

Subwoofers can be used to increase the main loudspeakers' maximum SPL and the system's low-frequency bass extension. This can satisfy the listener's requirements without having to use very large main loudspeakers.

### **What colors are available?**

Depending on the loudspeaker or subwoofer model, the following painted cabinet colors are available:

- Black (RAL 9005)
- Anthracite (RAL 7021, a neutral dark grey color)
- Silver (RAL 9006)
- White (RAL 9016)
- RAL (any color in the RAL scheme)



Refer to the "Variants" section of the product's web page for availability. The RAL website can be found here: <https://www.ral-farben.de/ral-farben.html?&L=1>

Mounting hardware is mostly black (RAL 9005) but in some cases white (RAL 9010) is also available. Other accessories are typically only available in one color scheme.



### **What mains power voltages can be used?**

Depending on the model, the mains power voltage can be:

- Fixed (100 V, 120 V or 230 V)
- User selectable (100 V, 120 V and 230 V)
- SMPS Universal (100 V – 240V)
- Battery (12 V – 20 V)

A suitably power-rated “step-up” or “step-down” transformer can be used to allow a loudspeaker to be used with a different mains power voltage. The mains frequency can be 50 Hz or 60 Hz. An “inverter” with a suitable power rating can be used in mobile applications where only a battery power is available.

**Note:** the 230 V position also works over the range 220-240 V (voltages seen in China and the UK).

### **What is it magnetic shielding and which models have it?**

Loudspeaker drivers contain magnets. In the case of bass drivers the magnets are large and so have a large stray magnetic field. This magnetic field will interfere with any CRT screens located near the loudspeaker and so the picture is distorted. Modern flat panel screens (plasma and LCD) are not affected by this problem however there are still many CRT screens still in use. As magnetic shielding reduces the stray field, the picture distortion is reduced.

Additionally computer hard drives and other magnetic storage media, e.g. floppy disks and DAT tapes, can be corrupted in the presence of strong magnetic fields. As magnetic shielding reduces the stray field, the risk of corruption is reduced. Even so it is still recommended to keep all magnetic storage media well away from loudspeakers.

Magnetic shielding is achieved by gluing a small extra magnetic onto the back of the driver’s magnet. The polarity of this magnetic is opposite to the driver’s magnet so that it cancels the stray magnetic field. Perfect cancellation is not possible, but a sufficient reduction is possible for placement of loudspeakers net to CRT screens.

Stray magnetic fields are also produced by mains power transformers. Frame transformers have a high stray magnetic field so we use toroidal transformers or switched power supplies. These have a much lower stray magnetic field of about one-tenth of that from a frame transformer, and a number of other benefits.

All the studio loudspeaker models are magnetically shielded.

### **Can the MMD waveguide be supplied rotated to the correct orientation for my installation?**

In the case of rotatable waveguide (e.g. [O 410](#)), unfortunately we are not able to supply products with waveguides rotated in a horizontal direction. This is because all loudspeakers leave the factory for a central warehouse with their waveguides oriented in a vertical orientation. When an order is placed, any one of the loudspeakers in stock could be selected and sent to you. In products that permit waveguide rotation, rotating the waveguide is not hard to do as it has been designed to be rotated in



the field – instructions are detailed in the operating manual.

Other products may have a fixed waveguide (e.g. two-way loudspeakers), a factory-rotatable waveguide (e.g. O 500 C), or no waveguide at all (e.g. M 52, subwoofers).

### **Do I need to “burn-in” / “run-in” my loudspeakers?**

Let us think about what parts can change their properties permanently in an active loudspeaker:

- **Electronics** – There is insignificant change with use over time. Permanent changes are usually the result of poor parts selection in design, early parts failure (bathtub curve), or user abuse. Some temporary change over short periods is seen due to heating up of components, but this resets when the electronics cools down. Active loudspeakers do not suffer significantly from these temporary changes as the sensitive crossover parts are positioned before the amplifier and so do not heat up much.
- **Cabinet Mechanics** – If this changes it is not well built!
- **Drivers** – The moving mass cannot change otherwise it means that something has fallen off, or some extra mass has been added. The self-damping should not change significantly otherwise it would mean that the cone material is falling apart. These two situations would only happen in the case of loudspeaker abuse or accidental damage. The suspension and surround can change their stiffness over time; however a good design should be robust to these minor changes. Even quite large, and highly unlikely in a correctly working loudspeaker, changes (doubling or halving the stiffness) will alter the overall loudspeaker performance by less than 1 dB in the low-frequency region only. Temporary changes are seen due to the voice coil heating up (thermal compression), but this resets when it cools down.
- **Psychoacoustics** – A human's auditory memory is VERY short, less than five seconds. This fact is well-known in the academic community, so one sees controlled ABX type listening tests with fast switchers to avoid the listeners forgetting what they just heard. Typically when burning-in loudspeakers a user listens to the loudspeaker for a while, does the burn-in over a long time period (say 24 hours, or even up to a week), then listens again. Having done this they are naturally inclined to hear a difference.

Our decades of experience in servicing products has shown that products which have been in normal use for many years typically have less than 0.5 dB of variance compared to specification. The largest source of variance is seen when drivers have been replaced, therefore recalibration to production standards is advised after servicing. If the long-term specification is stable, the short-term specification should also be stable.

**Concluding**, burning-in loudspeakers came from the past when poorly-designed passive systems were common. Today we have well-designed active solutions that can be used straight out of the box. To take an analogy from the automotive industry: cars used to require a running-in period of 500 miles (800 km), these days it is not required.



### **Can the studio products be used outdoors?**

Studio products are designed to be used indoors. Most control rooms are temperature and humidity controlled, but, where the room is not, it is recommended that the operating conditions fall into the ranges mentioned below:

The loudspeakers should only be used indoors and in these ambient conditions:

- +10° C to +40° C (+50° F to +104° F), <90% relative humidity, non-condensing

During transport or storage the ambient conditions can be:

- -25° C to +70° C (-13° F to 158° F), <90% relative humidity

The loudspeakers are tested to IEC 60065. A part of this test calls for a 48 hour unpowered soak test at 30° C (86° F) and 93% relative humidity, non-condensing.

### **How does adding a subwoofer affect the system's performance?**

Adding a subwoofer to a system brings some advantages:

- Decreased LF cut-off.
- Possible decreased main loudspeaker sized for a given system maximum SPL.
- Increased replay SPL, especially when playing bass heavy material, when using a sealed cabinet design for the main loudspeakers, or when using small main loudspeakers.
- Decreased distortion for a given replay SPL.
- Decreased group delay a frequencies previously reproduced by the main loudspeakers.

...and some disadvantages:

- Increased group delay around the subwoofer / main loudspeaker crossover frequency.
- If the subwoofers are located in one place or only one subwoofer is used, previously spatial distributed bass is reproduced from a single source location – classical music engineers particularly do not favor this.

### **Why is the subwoofer crossover frequency fixed and not variable?**

Variable frequency crossovers are not practical over multiple channels in analogue products due to the expense of implementing them accurately. This is not a problem in DSP systems. Unfortunately there are conflicting requirements for selecting the crossover frequency as it should be:

- As low as possible to minimize localization of the subwoofer, which is important for off-centre subwoofer locations.
- As high as possible to take the low-frequency workload off the main loudspeakers. This reduces distortion in the main loudspeakers which results in a cleaner sound.

- As high as possible to minimize the additional group delay caused by the electronic filtering – for more on group delay see [Question: [How does adding a subwoofer affect the system's performance?](#)].

To balance these competing factors we have chosen 80 Hz as an analog bass management crossover. This also happens to be compatible with the default setting typically seen in home theatre decoders. This should lead to a better professional-to-consumer translation.

90 Hz is used in some older products which results in a lower group delay and main loudspeaker distortion but suffers from increased localization and incompatibility with the consumer market.

### How does the crossover slope affect sound quality?

As with most factors in loudspeaker design, the slope of the crossover is a trade-off between parameters. The steeper the slope:

- The better the channel separation
- The lower the distortion of the loudspeaker
- The lower the intermodulation between the different channels
- The smaller the interference frequency area in the vertical off-axis sound.

The disadvantage is that the group delay increases with increased slope.

Typically used crossover slopes are shown in the table below:

Crossover Type	Filter Order	Slope
Passive	2nd	12 dB/oct.
Active or DSP (IIR)	4th	24 dB/oct.
DSP (FIR)	8th or more	48 dB/oct. or more

Digital FIR systems have the advantage that the phase response can be flattened. This is called linear-phase and results in no group delay increase around the crossover frequencies, but this comes with the penalty that the whole of the loudspeaker's response suffers from a longer delay called latency. Therefore the FIR-controlled loudspeakers (e.g. [O 500 C](#) or [O 300 D](#) in combination with [Pro C 28](#)) can be switched between linear phase (constant group delay but higher latency) and minimum phase (non-linear group delay but low latency)

### What is "group delay" and how does it affect sound quality?

In simple terms, group delay is the time it takes for an electrical input signal to become an acoustical output. It is frequency dependant and should ideally be zero seconds at all frequencies, but this is practically impossible.

#### Latency...

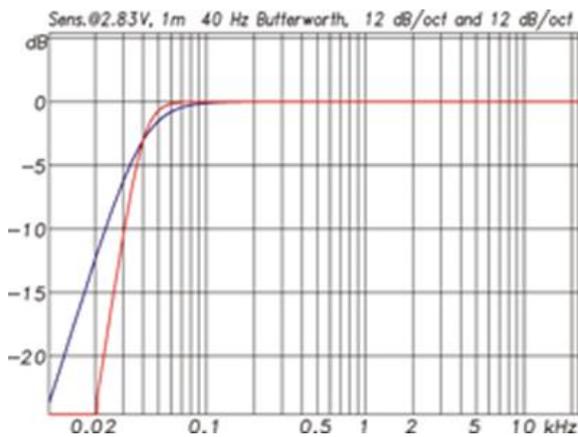
In a linear-phase system the group delay is constant – this is sometimes called latency. For example, in an [O 500 C](#) set to linear-phase mode the group delay is 80 ms at all frequencies. Whilst there are perceptual benefits in having a linear-phase sound reproduction system, this latency is not practical



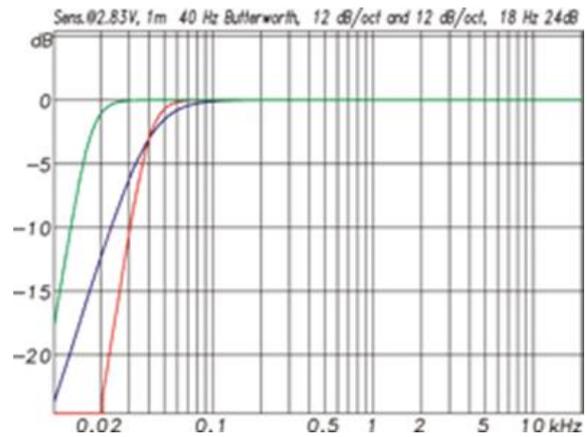
when the sound source and the loudspeakers are located in the same room as an echo will be heard. Conversely, when working with video equipment the audio signal generally leads the video signal due to video processing and so deliberate delay on the audio is required. This called lip sync. Typically flat display screens (plasma and LCD) have a two frame video delay (80 ms for a 50 Hz frames or 66 ms for a 60 Hz frames). Delays in the region of 0...8 frames may be required in broadcast stations due to additional video effects processing.

**Back to frequency-dependent delay...**

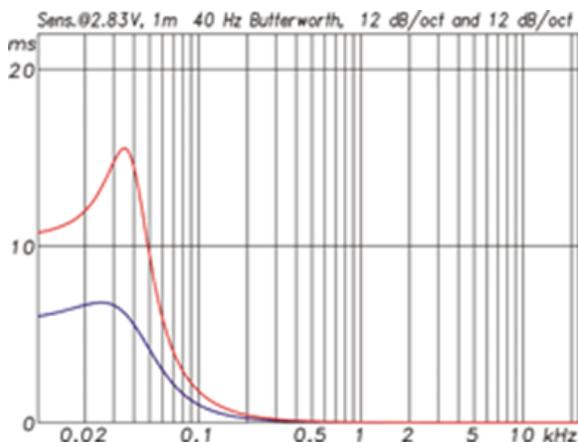
The subjective effect of excessive group delay is a “loosening” of the bass or a “less dry” bass quality. Currently there is insufficient psychoacoustic research on the threshold of group delay at low frequencies. One value is known: 2.5 ms at 100 Hz. This happens to be the same as that seen in the KH 310 A and O 410. A vented cabinet with a similar low frequency performance would have about 5 ms of group delay at 100 Hz. At higher frequencies, >1 kHz, group delay should be less than 1.6 ms (55 cm or 2 ft), which it is in most loudspeaker designs. Below are some graphs showing the theoretical frequency response and group delay of different types of enclosures (sealed and bass reflex, and a vented subwoofer).



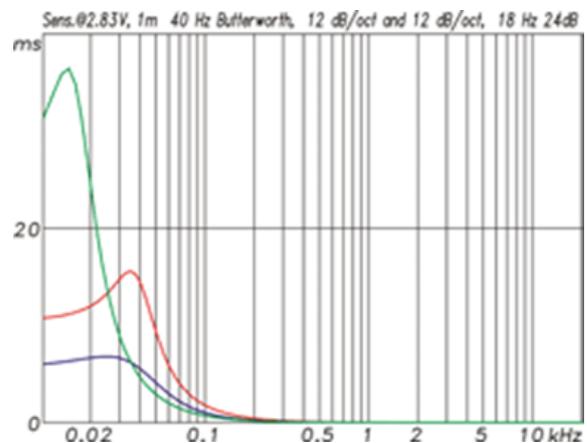
Frequency response of sealed (blue) and vented cabinets (red)



Frequency response of a vented cabinet with a lower LF cut-off (green)



Group delay of sealed (blue) and vented cabinets (red)



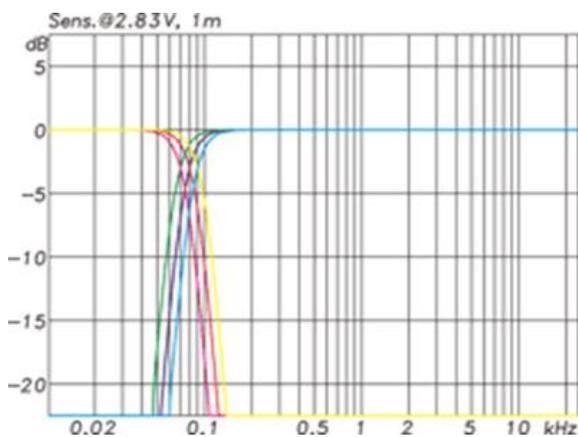
Group delay of a vented cabinet with a lower LF cut-off (green)



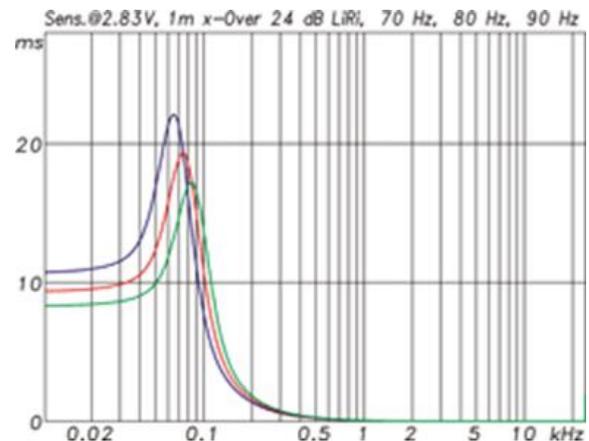
It can be seen in these graphs that a vented cabinet design must have a much deeper low-frequency cut-off for it to have the same group delay as a sealed cabinet. This is a good reason for having subwoofers with a very deep low-frequency cut-off, for example 18 Hz. When this subwoofer is added to a sealed near-field loudspeaker, the group delay performance is almost unaffected.

Adding infrasonic (high pass) filters to give very low-frequency driver protection adds to the group delay shown in the above plots. For example, a sealed cabinet with a second-order infrasonic driver protection filter with a corner frequency the same as the low-frequency cut-off of the loudspeaker will have the same group delay performance as a vented cabinet with the same low-frequency cut-off and no infrasonic driver protection filter.

An additional source of group delay is bass management filtering – see graphs below:

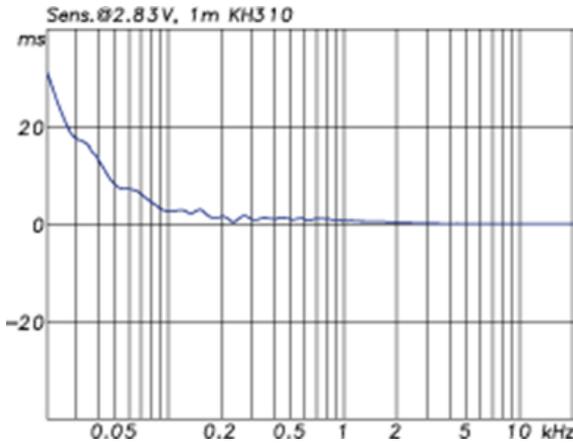


Three crossover frequencies: 70, 80, 90 Hz

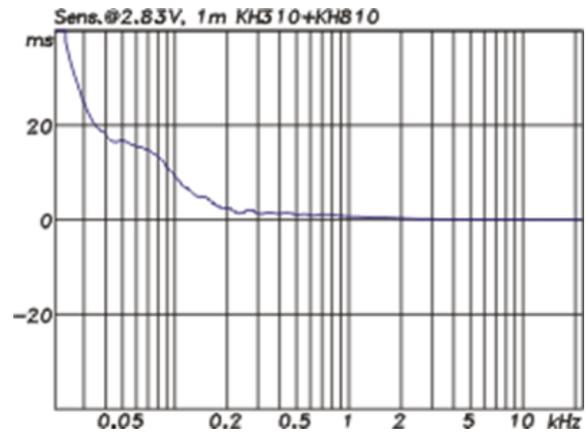


Group delay increases with reduced crossover frequency

Adding a subwoofer to a compact sealed loudspeaker increases the group delay slightly around the crossover due to the bass management crossover filtering - see graphs below around the 100 Hz region. As mentioned above, around the low-frequency cut-off of the main loudspeaker, the group delay is about the same with or without the subwoofer due to the very deep low-frequency cut off of the subwoofer. Below this the group delay continues to rise in the subwoofer as one would expect.

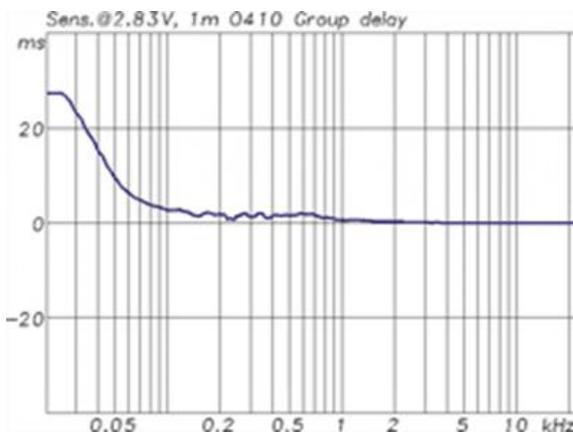


Group delay of a compact sealed loudspeaker (KH 310 A)

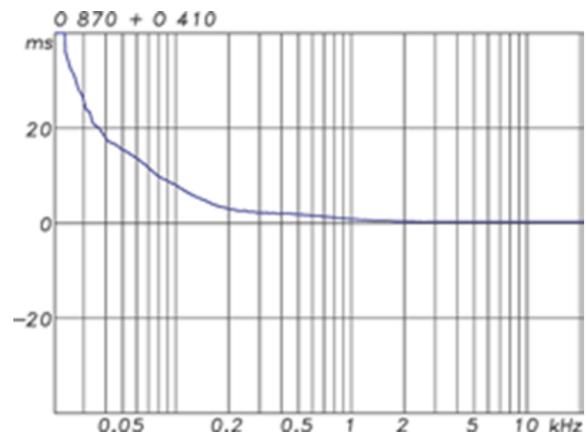


Group delay of a compact sealed loudspeaker with a subwoofer (KH 310 A + KH 810)

Moving onto a large vented cabinet, the effect in the group delay of adding the electronic filtering of the 80 Hz bass management crossover is shown in the right picture below. Now comparing the two left pictures (large vented and compact sealed), the group delay can be seen to be the same down to 35 Hz, and then it continues to rise down to the large vented loudspeaker's low-frequency cut-off.



Group delay of a large vented loudspeaker (O 410)



Group delay of a large vented loudspeaker with a subwoofer (O 410 + O 870)

So with a large vented loudspeaker we see the same group delay performance as a compact sealed loudspeaker, but with additional low-frequency extension and a higher maximum SPL. The cost is financial and a space in the room for the cabinet.

**What is distortion (linear, non-linear, harmonic, and intermodulation) and what does it mean for a loudspeaker?**

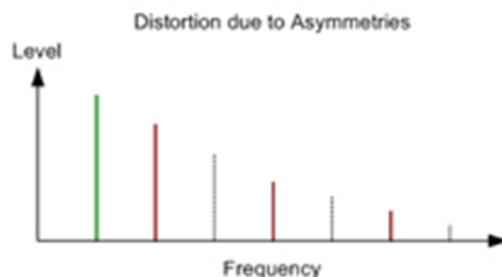
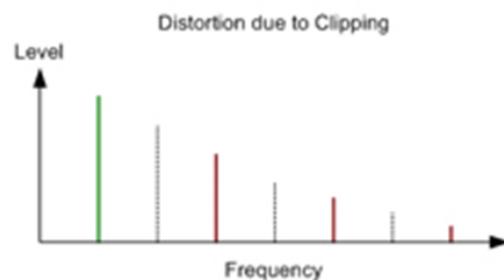
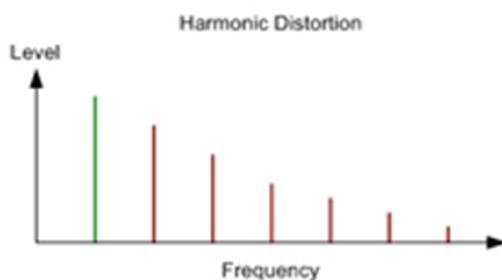
Generally, distortion is any unwanted change in the output signal when compared to the input signal. Distortion can be divided into different categories:



**Linear distortion** is an alteration of the amplitude or phase of the reproduced audio relative to the input signals. No new frequencies are generated in the reproduced audio by the loudspeaker as the shape of sine waves passed through the system is preserved. Conversely the shape of complex signals is changed, i.e. the sound changes in some way, more bass, less treble, etc. There is no linear distortion when the magnitude and phased of the frequency response is flat and linear respectively. Only a DSP based loudspeaker system can achieve this as analogue systems cannot adequately correct for the group delay at low frequencies.

**Non-linear distortion** creates new frequencies in the reproduced audio that were not present in the input signal. There are two types of non-linear distortion:

- **Harmonic frequencies** are ones that are not harmonically related to the input signal, for example, if the input signal is 1 kHz, the harmonics are 2 kHz, 3 kHz, 4 kHz, 5 kHz, etc. If the amplifier is clipping the generated harmonics are odd, for example for an input signal of 1 kHz, the newly generated harmonics are 3 kHz, 5 kHz, 7 kHz, etc. Asymmetries in the system generate even order harmonic, for the same signal we get 2 kHz, 4 kHz, 6 kHz, etc. The level of harmonic distortion frequencies do not correlate well with sound quality, for example odd order harmonics from clipping (e.g. exceeding 0 dB FS) are clearly annoying, whereas even order harmonics from asymmetries (magnetic tape saturation) can be quite pleasing. Even so harmonic distortion should be minimized in loudspeakers designed for reproduction accuracy.

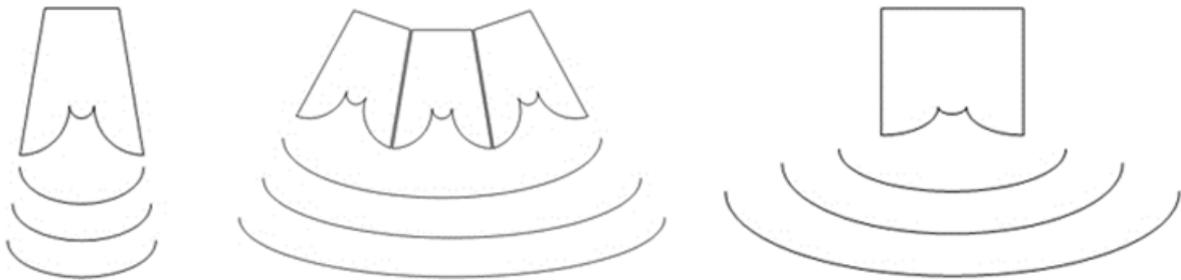


- **Intermodulation frequencies** are ones that are not harmonically related to the input signal. It is the interaction of at least two frequencies that creates intermodulation distortion (sums and differences of the frequencies in the signal). It is these additional frequencies that should be minimized. Whilst the threshold of audibility of intermodulation distortion is not academically well-understood, intermodulation distortion should be minimized in loudspeakers designed for reproduction accuracy.



**Horns are usually seen in PA applications, so why do studio loudspeakers use horns?**

Acoustical horns are used to increase the directivity of the reproduced sound into the three-dimensional listening space. Horns can also increase loading on the drivers from 1–2% to 15–20%. For PA loudspeakers, the horn decreases coverage angle but increases the SPL output for the same sized driver and amplifier. As the coverage angle is decreased for one cabinet, clustering is used to increase the coverage angle again. The very narrow directivity also minimizes interference between cabinets. The use of deep horns creates reflections off the mouth of the horn resulting in distortion.



In studio monitoring, acoustical horns are used to gently point the sound in the right direction to decrease off-axis sound reflecting off nearby surfaces, and to increase driver efficiency thereby lowering distortion. Additionally, edge diffraction is reduced resulting in a smoother midrange. Mouth reflections have a very low level as the horns are relatively wide and shallow, therefore the acoustical cost of adding a horn is insignificant.

**How do the driver protection limiters work and can I damage the loudspeakers if the protection light flashes?**

Driver protection limiters monitor each channel of the signal after the crossover (bass, treble, and midrange if present). If any one of the channel levels is too high for too long the protection system activates and attenuates the wideband input signal, thereby avoiding damage to the loudspeaker. Additionally there is another limiter for the bass driver to protect it from excessive excursion. An activated protection system is indicated by a dedicated “Protect” light or flashing logo. The system automatically resets itself once the drivers have cooled down.

The driver protection system can protect the drivers from short-term accidental high input levels. It has limited ability to protect the system from systematic abuse. If the protection system is regularly activated consider using a large loudspeaker, adding one or more subwoofers, and/or turning down the signal level.

**Why does the protection system activate at different levels in different loudspeaker units even though input signal is the same?**

There are several places where tolerances can occur in a loudspeaker:

- Electronics: response shaping, crossover, limiters, amplifiers
- Acoustics: drivers and damping

During the production trimming phase the loudspeaker is aligned to a reference curve. Trimming of up to 2–3 dB is required to compensate for the factors listed above. The greatest variance is seen in the low-frequency region. This is also the frequency region where a higher amplifier power is needed and thus the region that is most likely to cause the protection system to activate and be indicated by the “protect” light or flashing logo. So for two loudspeakers aligned to be exactly the same, differences in the sensitivity of the acoustical section (can be up to +/-2.5) require an output from the amplifier that varies over the same, but opposite, range. Less sensitive drivers require more amplifier power and thus the electronics will limit earlier to protect the amplifier.

This is illustrated in the table below with some simplified fictitious numbers. The driver sensitivity of the three loudspeakers varies over a range of 2 dB (89–90 dB/W/m), this causes the pre-amp level (represents the production trimming) to vary over the range +/-1 dB. It can be seen that with the same input level and power amplifier gain, the power amplifier can be driven up to 2 dB harder for the same dB SPL output. Therefore one can see that the protection system will be activated at a 2 dB lower level in loudspeaker A than in loudspeaker C.

Loudspeaker	Input Signal [dBu]	Pre-amp Gain [dB]	Power-amp Gain [dB]	Power-amp Output [dBu]	Driver Sensitivity [dB/W/m]	SPL Output [dB SPL at 1m]
A	0	1	20	24	89	110
B	0	0	20	23	90	110
C	0	-1	20	22	91	110

**How long can my loudspeakers be on without powering them off?**

The loudspeakers can be left on indefinitely without damaging them. However these days one should consider the environmental impact of leaving on any electrical equipment that is not being actively used. The loudspeakers do not require warming up to operate according to their specification, so powering down the loudspeaker will not incur a loss of time waiting for them to warm up again. Simply turn them off when not in use and turn them on again when required. To aid this process a 12 V trigger is available on some models. If this is not available on your loudspeakers, simply plug them into a single switched power distribution strip and turn them on and off from there.

**Can I use a new subwoofer design with an older product?**

It is possible to use a new subwoofer design with an older product design, for example, a KH 810 with the OY. Simple connect the signals to the subwoofer then into the main loudspeakers and calibrate the system in the normal way.

### **Are there plans to release center versions of the existing range?**

International standards for multi-channel systems state that the center loudspeaker should “match” the left and right loudspeakers. The best way to match to the left and right loudspeakers is to use the same model.

- For smaller products (e.g. M 52, KH 120 A, KH 120 D, KH 310 A, KH 310 D), the cabinets are sufficiently compact that a dedicated centre version is not required for most installations. Sight lines into the live room are already as good as they will get and obstruction of the sound from main monitors is already minimized. In addition, for the KH 310, a dedicated centre version would cost more and not be any shorter.
- For larger products (e.g. O 410 and O 500 C), the waveguide can be rotated to give a lower height loudspeaker.

### **How does the 12 V trigger work and what connector is required?**

#### **What plug do I use?**

The connector is called “Euroblock” and is also often referred to as “pluggable screw terminal block”. Phoenix is one manufacturer (<http://www.phoenixcontact.com>) and Altech Corp is another (<http://www.altechcorp.com>). It is commonly used in the installed sound market. The control cables are screwed onto a plug block which is then inserted into the socket.

### **What is the nominal voltage and voltage range and what current is required?**

There are two types of circuit employed in the products:

- Relay fitted to O 410:  
The 12V DC trigger input is directly connected to a relay coil. The relay is a 12 V DC type - meaning that this is the nominal voltage. The datasheet specifies a tolerance of 9.6 – 18 V DC. To be safe, assume a useable operating range of 10 - 17 V DC. The 220 Ohm relay coil requires a nominal 55 mA (45 - 77 mA) to operate.
- Solid-state interface fitted to KH 810, KH 870, O 810 and O 870:  
The 12V DC trigger input is connected to a solid-state electronic circuit. The input voltage should be 12 V DC, +/-10% (10.8 – 13.2 V DC). 18 mA is required to drive the circuit.

Both input types are protected against incorrect polarity. 12 V is typically seen on control equipment from companies such as Crestron and AMX. 5 V, seen on some equipment, is defiantly too low.

To ensure reliable switching, the total required current for a system of loudspeakers should not exceed the ability of the source to provide it.

### **Is it possible to switch on multiple monitors at the same time?**

To turn on multiple products at the same time requires two things:

- The 12 V trigger source must be able to supply the total current required of each 12 V trigger circuit – see value mentioned above. If it cannot, separate sources should be used.



- The mains power supply must be able to supply the peak current for all the connected products otherwise fuses will blow or circuit breakers will trip.

#### **Can the mains power switch be left on and the loudspeaker switch on and off remotely?**

This is exactly what the 12 V trigger is designed for, so no damage will result if the loudspeaker is powered up and down regularly in this way.

#### **Can the mains power switch be left on and the loudspeaker switched on and off using another switch?**

Leaving the loudspeaker switched on and then turning it on and off externally will not damage the loudspeaker. Large facilities are often set up in this way - one big switch turns on the entire room (speakers, mixing console, effect racks, etc.). With a suitable time delay, the 12 V trigger function can then be used to turn on the loudspeakers last.

#### **Are Neumann loudspeakers pair matched?**

Pair matching as a concept comes from passive designs where there is little opportunity to tweak each unit to be exactly the same as every other unit. Therefore two units are chosen that are quite closely matched to each other and then they are shipped as a pair. In some cases one sees active products that lack production trimmers, so these products also benefit from the efforts of pair matching.

In our active systems every monitor passes through a final test procedure where production trimmers are adjusted to compensate for tolerances in the electro-acoustical system (drivers, crossovers, etc.). The target response is flat in free-field conditions as the product is a studio monitor - a measurement device for exactly converting the electrical input signal to acoustical pressure at the listening position. All our monitors are calibrated to this standard therefore every monitor is pair matched against every other monitor of the same type.

As with any product there is the risk of transit damage or tampering. We have well tested packing designs to avoid the former but the latter is outside of our control. You can check for tampering by inspecting the product for unusual signs of interference. If you suspect anything please send a picture and/or comments to Neumann.

#### **What is the LFE-channel and how is it reproduced?**

Reproduction of the LFE-channel offers the biggest opportunity for replay error in a multichannel system. In this answer we describe first what the LFE-channel is, and then how it can be reproduced in a multichannel loudspeaker system.

#### **Firstly, the signals...**

##### **What is the LFE-channel?**

The LFE-channel comes from the movie industry as a method to transmit high-level low-frequency content from the production environment to the movie theatre, without having to compromise the signal-to-noise ratio or overload the main channels.

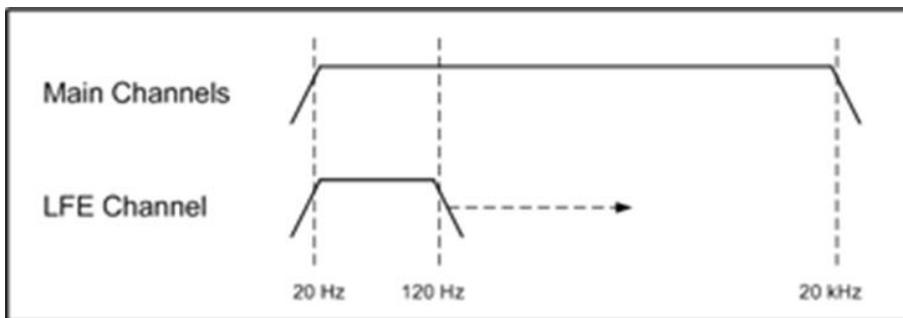
The LFE-channel signal can be correlated to (having the same phase relationship) or de-correlated from (having a different phase relationship) the main channels:

- **Correlated** signals occur when the same signal is mixed into the main channels and LFE-channel, for example a bass drum in a music mix. It is therefore important that the phase of all the channels in the reproduction system is the same so that phase-based cancellations do not occur, otherwise adding the bass drum into the LFE-channel could result in less, not more, level. The implication of this is noted in the system configurations below.
- **De-correlated** signals are used in the movie industry for the LFE channel, typically band-limited pink noise as a special effect. This, plus the fact that the overlapping bandwidth of the main loudspeakers and the subwoofer is quite small, means that the phase relationship is less critical.



### What is the frequency range of the LFE-channel?

Typically the LFE-channel contains frequency content up to 120 Hz, but higher cut-off frequencies can be found in formats such as SDDS (330 Hz) and MPEG-4 (1 kHz). Note that the upper cut off frequency is a function of the source, not the reproduction system. Although the LFE-channel specification appears to be the same in the DTS and Dolby formats, the phase responses around the upper cut-off frequency are not the same as the 120 Hz low pass filters are different. This has implications when coherent signals are played through the main channels and the LFE-channel, as the summing of the signals (electrical or acoustical) during reproduction will be different. We make no judgement here on which is right or wrong, they are what they are. The job of a reproduction system is to reproduce the electrical input signals faithfully.



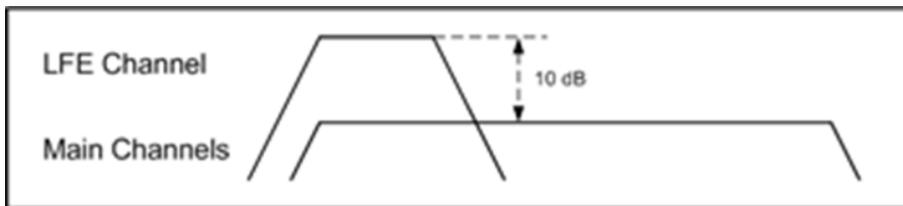
### What is the level of the LFE-channel?

For Dolby Digital and DTS multichannel signals, the LFE-channel should be reproduced at a level 10 dB higher than the main channels. This 10 dB boost must happen somewhere between the mixing console's multichannel downmix bus and the listener's ear. It can happen in any ONE of the following places:

- In the mixing console's monitor matrix
- A dedicated 10 dB amplifier inserted into the signal path
- If present, in bass management system
- If wired in this way, by a dedicated LFE-channel subwoofer set to 10 dB louder than the main channels

All decoders and DVD players should automatically add 10 dB to the LFE-channel signal after decoding. Unfortunately this is not always the case as some (usually cheaper) products on the market have low voltage power supplies that cannot output these higher signal levels. It is recommended to avoid these products.

Sometimes the source is previously mastered material, for example anything replayed by a DVD source. If this is played through one of the above systems, the 10 dB gain on the LFE-channel must be removed, i.e., the LFE-channel's gain should be the same as the main channels. This is because the decoder in the DVD player or surround sound processor inserts the 10 dB boost automatically onto Dolby or DTS encoded material.



### Phase of the LFE-channel relative to the main channels

The phase of the LFE-channel relative to the main channels is not the same due to the filtering required to limit the LFE-channel's upper cut-off frequency. The phase relationship is fixed for a particular format. Even so, the point of the loudspeaker system is to accurately reproduce the electrical input signal and so it should not make any changes to the phase of the channels relative to each other. If there is a phase difference, the result will be less bass at the listening position. To check if the relative phase of the channels is correct, turn the LFE-channel on and off. The reproduced bass in the listening room should not increase when the LFE-channel is muted. A bass drum routed to the main channels and the LFE-channel makes a good signal for this test.

The best way to achieve the same phase in the main channels and the LFE-channel is to use the same loudspeakers to reproduce them. It is easy to guarantee coherence when electrically summing the channels using a bass management system. Then the combined bass signal can be routed across the appropriate subwoofer(s) and main loudspeakers.

### Next, the Loudspeaker System...

#### **What should be the crossover frequency between the main loudspeakers and the subwoofer(s)?**

The crossover frequency between the main loudspeakers and the subwoofer should be high enough to be of benefit to the main loudspeakers, in terms of increased SPL and reduced distortion.

The crossover frequency between the main loudspeakers and the subwoofer should be low enough to ensure that the subwoofer cannot be localised.

A consequence of adding a crossover to the loudspeaker system is an increase in the group delay of the system around the crossover. A lower crossover frequency increases the group delay more than a higher crossover frequency. A system with a lower crossover frequency will have a "less dry" sound quality in the bass than a system with a higher crossover frequency which will have a "tighter" sound quality.

As loudspeaker design is usually a trade off between parameters, to balance all these factors, the crossover frequency should be in the region of 80–90 Hz.

### How many subwoofers are needed?

Typically a single subwoofer is used. Sometimes more than one subwoofer is used.

The advantages of multi-subwoofer systems are:

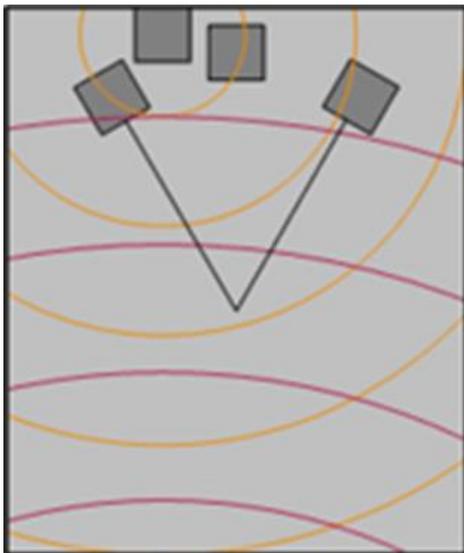
- Increased low-frequency output capacity compared to one subwoofer:
  - 2 subs → +6.0 dB SPL
  - 3 subs → +9.5 dB SPL
  - 4 subs → +12.0 dB SPL
- Decreased low-frequency distortion
- Allows for Plane Wave Bass Array™ techniques (PWBA™) to reduce side-to-side variance – see pictures below
- Multiple smaller units, which might fit better into the available space, can be used instead of single larger units

The disadvantages of multi-subwoofer systems are:

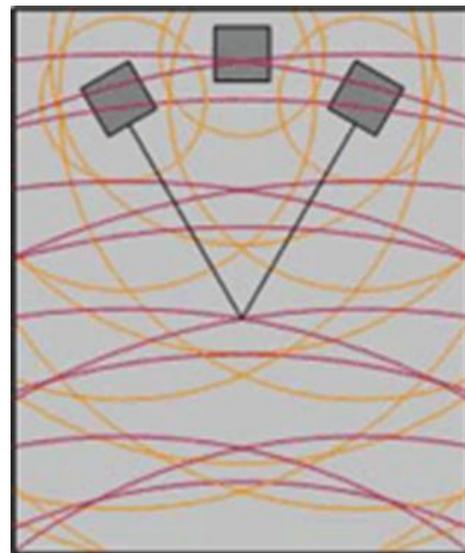
- More (different) space required
- Can be harder to set-up
- Increase budget required

### How does the low-frequency energy radiation affect sound quality?

In the examples below, only a single unattenuated back wall reflection is displayed to keep the picture simple to read:

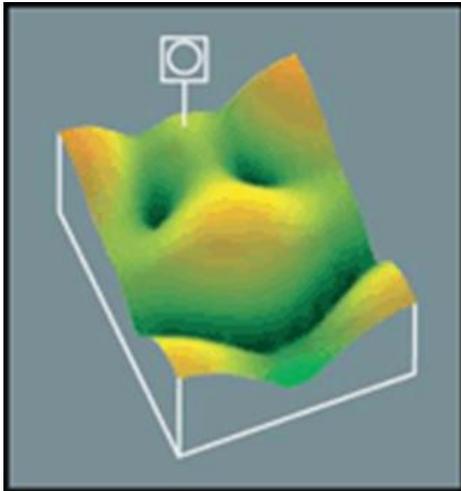


Bass reproduced by a single subwoofer has fewer, but larger, interferences

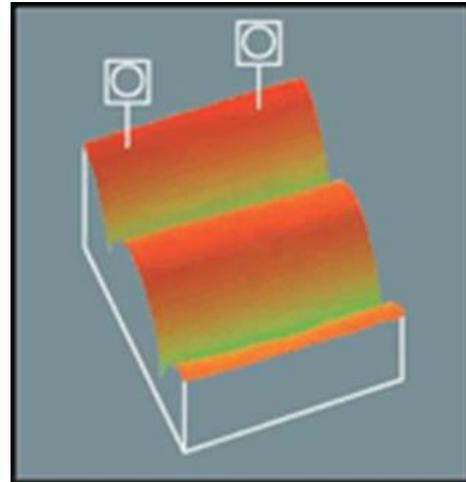


Bass reproduced by multiple full-range systems has many, but smaller, interferences

In the examples below, the sum of all horizontal plane reflections are displayed:



A single subwoofer gives large side-side variation



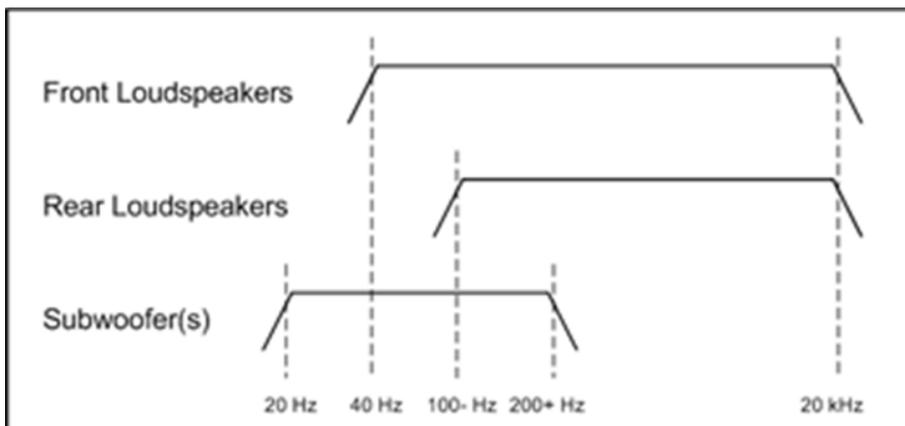
Multiple subwoofers have less side-side variation (PWBA™)

**Summary:** As we can see in the pictures above, routing the low frequency content of the main channels and LFE-channel to a set of large main loudspeakers or an array of subwoofers brings the advantage of more consistency when moving left-right across the listening area.

#### Which loudspeakers should be used?

There are many different ways to reproduce a multichannel signal:

In the professional movie industry the front channels are connected directly to the (large) main loudspeakers. The rear channels are connected to an array of loudspeakers to give good coverage over a large area.



- **Advantages:** Production and replay environments have been internationally standardised to be the same for many years. The methods are conceptually simple and well-understood by all involved.
- **Disadvantage:** Multiple loudspeakers reproducing the same signal suffer from comb filtering. Information below the LF cut-off of the main channel loudspeakers is not reproduced.



- **LFE-channel:** Connected directly to a subwoofer(s) which is calibrated to give the 10 dB boost in the listening space. The 10 dB either comes from increased gain (during production) or from the decoder (during reproduction).

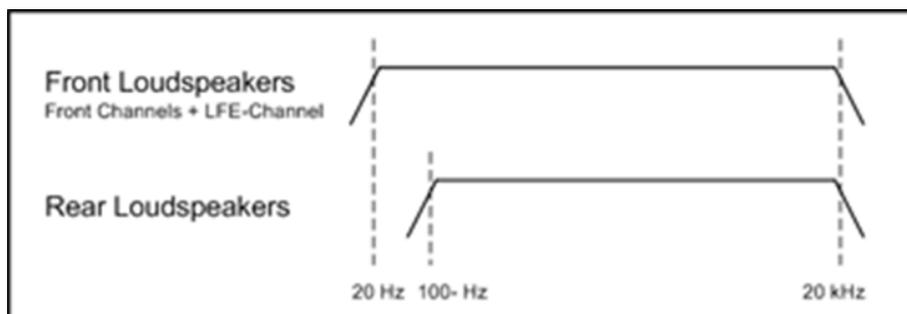
These methods are used in the domestic music and domestic movie industry (DVD and Blu-ray)...

### Systems without bass management.

- **Type A** – Same system setup as used in the movie industry (see above)
  - **Advantages:** Cheap to implement. Cabling is simple. Easy to set up.
  - **Disadvantages:** Low frequencies present on the main channels are not reproduced. Does not reduce the main loudspeakers' distortion or allow them to play louder. Main channels are only reproduced down to the low-frequency cut-off of the loudspeakers.
  - **LFE-channel:** Typically replayed by the subwoofer only

Neumann does not recommend this method.

- **Type B** – Route the LFE-channel to full-range large loudspeakers.

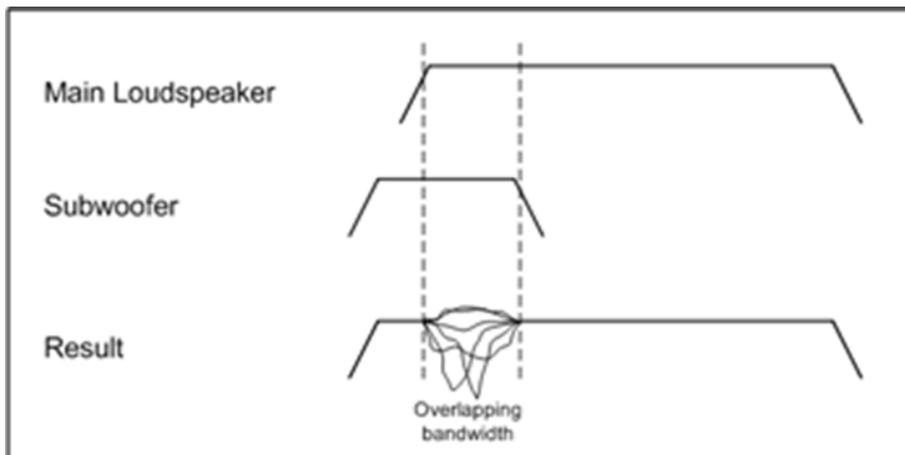


- **Advantages:** Phase relationship of the LFE-channel and the main channels are maintained. The LFE-channel should be played through all three front loudspeakers and then trimmed to give the correct level at the listening position.
- **Disadvantages:** Can only be used as a reproduction method when the main loudspeakers have a low frequency cut-off of 20 Hz or less. The LFE-channel should be routed in some way to all three front loudspeakers and then trimmed to give the correct level at the listening position.



Systems **with bass management** redirect the low-frequency content of the main channels to a subwoofer.

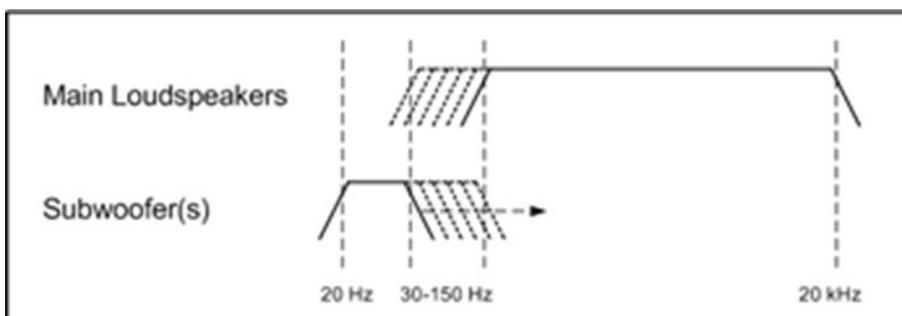
- **Type A** – Subwoofer has a variable upper cut-off frequency to extend the natural low-frequency roll-off of the main loudspeakers down in frequency:



- **Advantages:** Cheap to implement. Cabling is simple.
- **Disadvantages:** Unreliable summing (anything from +6 dB to –30 dB) in the overlapping subwoofer-loudspeaker region. Does not reduce the main loudspeakers' distortion or allow them to play louder.
- **LFE-channel:** Typically replayed by the subwoofer only.

Neumann does not recommend this method.

- **Type B** – Subwoofer has a variable upper cut-off frequency to extend the natural low-frequency roll-off of the main loudspeakers down in frequency:

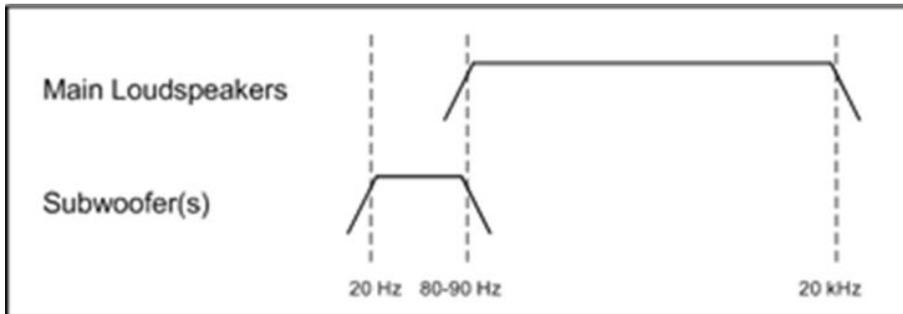


- **Advantages:** Cheap to implement. Cabling is simple.
- **Disadvantages:** Harder to setup. Does not reduce the main loudspeakers' distortion or allow them to play louder.
- **LFE-channel:** Typically replayed by the subwoofer only.

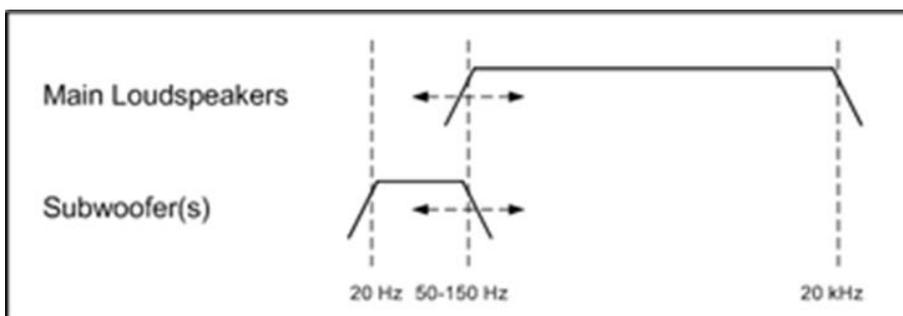


Neumann does not recommend this method.

- **Type C** – Bass management system has a fixed crossover frequency:



- **Advantages:** Good balance of all acoustical factors. Main channels reproduced by the main loudspeakers and the subwoofer(s) down to the low-frequency cut-off of the subwoofer. Reduces the work done by the main loudspeakers thereby reducing the system's distortion and/or allowing the main loudspeakers to play louder. Significantly easier to setup.
  - **Disadvantages:** Acoustic problems near the crossover frequency have to be addressed acoustically. Signal cabling has to pass through a central location for signal processing.
  - **LFE-channel:** Can be replayed by the subwoofer only or partly by the subwoofer and partly by the main loudspeaker.
- **Type D** – Bass management system has a fixed crossover frequency:



- **Advantages:** Main channels reproduced by the main loudspeakers and the subwoofer(s) down to the low-frequency cut-off of the subwoofer. Acoustical issues near the crossover can be avoided by moving the crossover frequency. Depending on the chosen crossover frequency the work done by the main loudspeakers is reduced, thereby reducing the system's distortion and/or allowing the main loudspeakers to play louder. DSP processing must be used.
- **Disadvantages:** Requires acoustical measurements to help choose an appropriate crossover frequency. It is usually better to solve acoustical problems in an acoustical way rather than electronically. A low crossover frequency (<60 Hz) brings a large increase in group delay around the crossover frequency. A high crossover frequency (>100 Hz) can lead to localisation of the subwoofer. To be practical, DSP processing must be used.

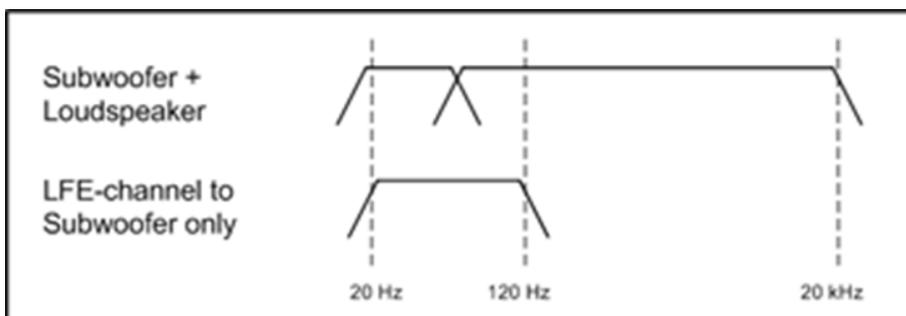


- **LFE-channel:** Can be replayed by the subwoofer only or partly by the subwoofer and partly by the main loudspeakers.

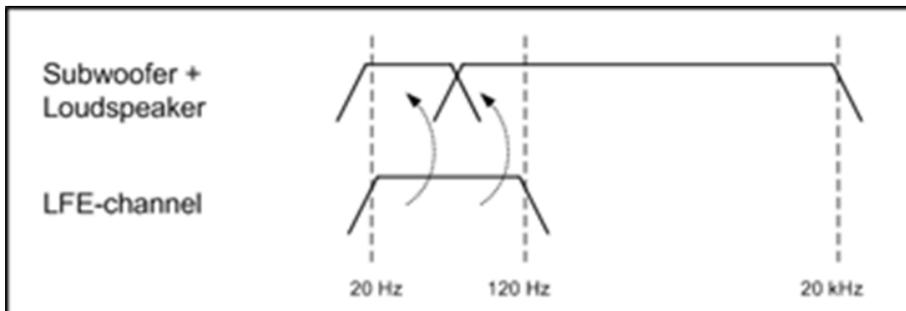
**What happens to the LFE-channel above the crossover frequency?**

If the LFE channel is played into the subwoofer and the LFE-channel has an upper cut-off with a frequency higher than the crossover frequency, some re-routing may be required:

**Option 1:** Play the LFE-channel into a wideband input on the subwoofer. No re-routing to other loudspeakers.



**Option 2:** Play the LFE-channel into a parallel input to the main channels so that it is reproduced by the subwoofer and one or more of the main loudspeakers.



The re-routing of the LFE channel above the crossover can be to the centre loudspeaker, the left + right loudspeaker, or all three front loudspeakers. Reproduction over more than one loudspeaker will require an electronic gain change to compensate for the acoustical gain.

**What is the Mathematically Modelled Dispersion™ (MMD™) Waveguide?**

A waveguide is a shape on the front panel of the loudspeaker that controls the direction in which the sound can radiate into the space around the loudspeaker. For PA systems the radiation angle is quite narrow (typically 40-80 degrees) to avoid cabinet interactions in large clusters and to point the sound in a specific direction. For a studio monitor the radiation angle is usually wider (>90 degrees) as a large listening area is required for quite short listening distances. Also there is a need for a narrowed vertical angle to reduce mixing console and ceiling reflections, so the design must be three-dimensional.



A well-designed studio monitor waveguide leads to a flatter on-axis response, smoother off-axis response, higher efficiency, lower distortion, closer phase alignment between the drivers, and reduced interaction with the room and its contents. These are all good objective goals as the result is a smoother frequency response under listening conditions. Conversely, a badly designed waveguide can be worse than no waveguide at all. Designing a good waveguide is not easy. It can either be done manually using clay (like the way cars used to be designed) or these days it is possible to make an acoustical model in a computer and run an iterative process to yield an optimal result compared to an ideal design target. The resulting design is mechanically harder and more expensive to realise as a 3D shape is more complex than a classic rectangular cabinet. The cabinet with its waveguide design is then physically built (moulded or CNC machined) and verified with real measurements in anechoic conditions. Finally, much listen confirms that the design is good.