NEUMANN.BERLIN

KH 310 II
KH 350

Active three-way loudspeaker

Instruction manual

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Contents

KH 310 II and KH 350 studio monitors

Thank you for purchasing a Neumann studio monitor. The loudspeaker features a Mathematically Modeled Dispersion™ Waveguide (MMD™), DSP acoustical controls, a digital AES3id input and output, network control capabilities, and an extensive range of mounting hardware. This allows the loudspeaker to be used in diverse acoustical conditions, with any source equipment and in a wide variety of physical locations. The KH 310 II / KH 350 represents the latest in acoustic and electronic simulation and measurement technologies to ensure the most accurate sound reproduction possible.

Depending on the level requirements in the overall system and the listening distance, Neumann's three-way systems are designed for use as near field or mid field monitors or as rear or ceiling-mounted loudspeakers in larger multi-channel systems. They can be used in project, music, broadcast, and post production studios for tracking, mixing, and mastering.

Scope of delivery

1 KH 310 II or KH 350
1 Self-adhesive feet (KH 310 II)
1 Quick guide
1 Safety instructions
1 Mains cable (EU/Korea, UK, USA or China – depending on the region)

About this instruction manual

This instruction manual describes the physical setup and autonomous operation of the loudspeaker using the controls on the rear. For information on how to control the speaker over a network using the **MA 1 – Automatic Monitor Alignment** software, please refer to the software help.

The **MA 1 – Automatic Monitor Alignment** software is available for macOS and Windows and offers the following advantages:

- Automatic adaptation to the particular listening conditions
- Calibration to the listening environment
- Optimized stereo imaging
- Perfect matching of amplitude and phase to subwoofers
- Compensation for suboptimal listening situations
- Settings for system level, logo brightness, and other features
- Rapid system reconfiguration

The **MA 1 – Automatic Monitor Alignment** software is available at www.neumann.com.

Product overview: KH 310 II

Front view

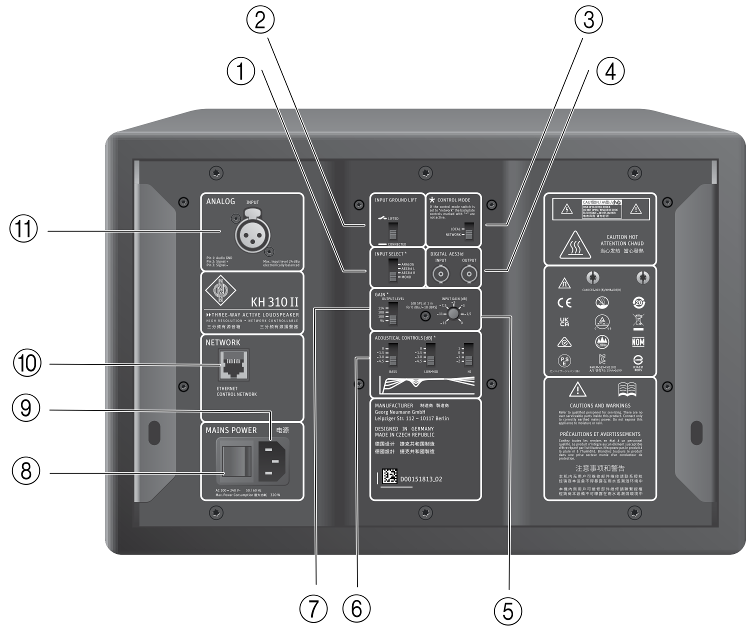
① Woofer

② Tweeter

③ Neumann logo (illuminated, status display)

④ Midrange

Rear view



① INPUT SELECT switch

② INPUT GROUND LIFT switch

③ CONTROL MODE switch

④ BNC socket | DIGITAL AES3 INPUT
BNC socket | DIGITAL AES3 OUTPUT

⑤ INPUT GAIN control

⑥ ACOUSTICAL CONTROLS switch

⑦ OUTPUT LEVEL control

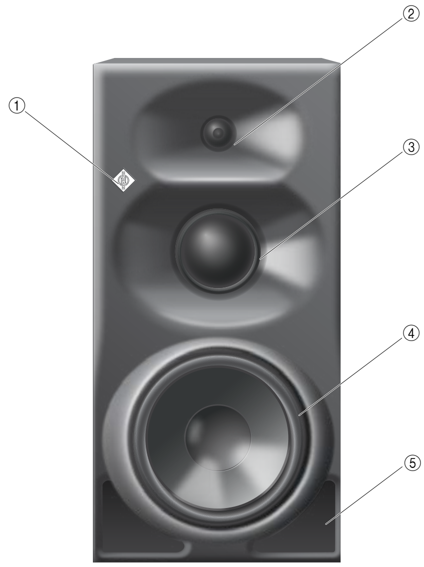
⑧ MAINS POWER switch

⑨ IEC mains socket with protective ground contact

⑩ ETHERNET CONTROL NETWORK socket

⑪ XLR3-F socket | ANALOG INPUT

Product overview: KH 350



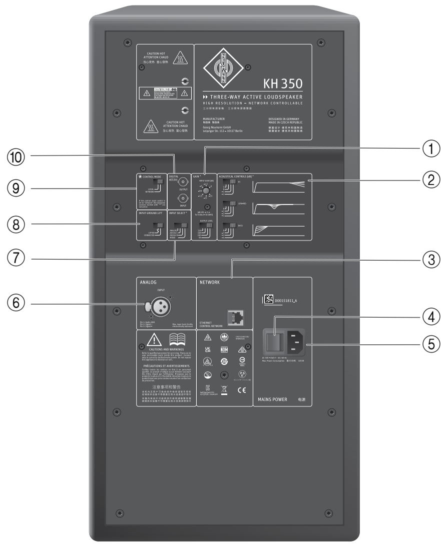
① Neumann logo (illuminated, status display)

② Tweeter

③ Midrange

④ Woofer

⑤ Front ports



① OUTPUT LEVEL | INPUT GAIN

② ACOUSTICAL CONTROLS switch

③ ETHERNET CONTROL NETWORK socket

④ MAINS POWER switch

⑤ IEC mains socket with protective ground contact

⑥ XLR3-F socket | ANALOG INPUT

⑦ INPUT SELECT switch

⑧ INPUT GROUND LIFT switch

⑨ CONTROL MODE switch

⑩ BNC socket | DIGITAL AES3id INPUT
BNC socket | DIGITAL AES3 OUTPUT

Setting up and connecting the loudspeaker

Have the product installed and connected by a specialist. Due to his/her technical training, know-how and experience as well as knowledge of relevant provisions, regulations and standards, the specialist must be able to assess assigned tasks, recognize potential hazards and ensure appropriate safety measures. The following safety and mounting instructions are addressed to this specialist.

CAUTION
Danger of injury and material damage due to tipping/dropping of the product!

If improperly mounted, the product and/or the mounting hardware (e.g. rack) can tip over or fall.

- ▶ Always have the product mounted by a qualified specialist according to local, national and international regulations and standards.
- ▶ Use the mounting systems recommended by Neumann and always provide sufficient additional protection against tipping or falling.

CAUTION
Damage to the product due to overheating!

If air cannot circulate freely behind the product, the product electronics may overheat and trigger the thermal protection system. This reduces the maximum output level and may result in damage to the product.

- ▶ The whole of the backplate should have a free flow of air to ensure sufficient cooling.
- ▶ When installing the product into tight spaces such as wall recesses, maintain an air gap of at least 5 cm around the top, rear and side panels of the product and provide sufficient air circulation. If necessary, use forced-air cooling.

For further information on setting up loudspeakers, please refer to the “Questions & Answers” section on the product page at www.neumann.com.

For more information on building systems using Neumann loudspeaker products, please refer to the Product Selection Guide at www.neumann.com.

Preparing the loudspeakers

CAUTION
Risk of staining surfaces!

Some surfaces treated with varnish, polish or synthetics may suffer from stains when they come into contact with other synthetics. Despite a thorough testing of the synthetics we use, we cannot rule out the possibility of staining.

- ▶ Do not place the loudspeaker on delicate surfaces.

To place the loudspeaker on a flat surface:

- ▶ Attach the supplied self-adhesive feet to the bottom of the loudspeaker (KH 310 II).

  This reduces the risk of scratching the surface and acoustically isolates the loudspeaker from the surface. Alternatively, use a soft, non-slip surface.
  In the case of strongly resonating substrates, additional acoustic decoupling (e.g. layers of foam) may be useful. An ideal decoupling effect is achieved when the loudspeaker swings slowly back and forth when tapped, at a frequency that is significantly lower than the lower cut-off frequency of the loudspeaker. Make sure that the loudspeaker is still stable, cannot tip over, and that there is nothing on the loudspeaker that can fall off.

Preparing the room

- Arrange all acoustically relevant surfaces and objects symmetrically on either side of the listening axis of the room (left/right).
- Minimize the sound that is reflected back to the listening position by using angled surfaces and/or acoustic treatment (diffusion, absorption).

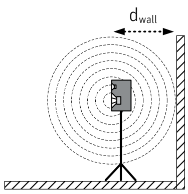
This product has been optimized for use in recording studios. In order to not affect the quality of playback, make sure that the product is used in an EMC environment, ideally without electromagnetic radiation from other devices.

Positioning the loudspeakers

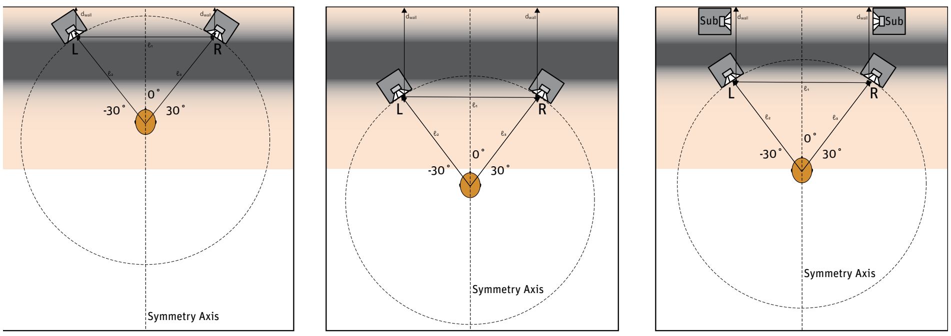
- Carry out the following steps very accurately, since the more accurate the physical arrangement of the loudspeakers in the room, the more accurate the reproduction will be at the listening position.

Spacing

- Observe the recommended distances between the loudspeakers and your listening position:
  - Recommended listening distance for KH 310 II: 1.0 m – 2.5 m
  - Min./max. listening distance for KH 310 II: 0.75 m – 6.0 m
  - Recommended listening distance for KH 350: 1.2 m – 3.0 m
  - Min./max. listening distance for KH 350: 0.75 m – 6.0 m



- Avoid positioning the loudspeaker at a distance (d_{wall}) of 0.8 to 3 m from the wall behind the loudspeaker. When using loudspeakers with a bass-managed subwoofer, avoid distances (d_{wall}) of 0.8 m to 1 m from a solid wall behind the loudspeaker. Similarly, avoid these distances from solid side walls or a solid ceiling. Respecting these positioning limitations reduces the chances of dips in the low frequency response (comb filtering) caused by strong reflections.



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Installation angles

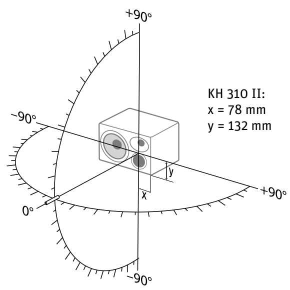
- Print out the “Installation angles” diagram that can be found at the end of this operating manual.
- Place the diagram at the listening position or the center of the listening area.
- Using a tape measure, place the loudspeakers at the same distance from the center of the diagram “Installation angles”. To ensure good imaging, do this at an accuracy of at least 1 cm.
- If the loudspeakers cannot be placed at the same distance from the listening position, compensate for distance differences > 1 cm by delaying closer loudspeakers by 30 µs/cm. The **MA 1 – Automatic Monitor Alignment** software automatically compensates for different distances between the loudspeakers and the listening position.
- Position your loudspeakers as follows:

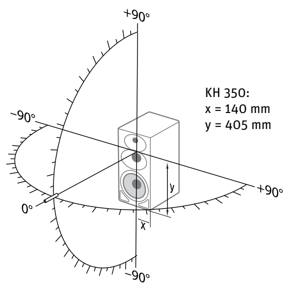
System	Position and orientation
2.0 (stereo)	±30°
5.1	ITU-R BS.775-1: 0°, ±30°, ±110° (center, front left/right, surround left/right)
	ANSI/SMPTE 202M: 0°, ±22.5°, arrays to the surround left and to the surround right, plus optional subwoofer(s)
Dolby Atmos 7.1.4	0°, ±30°, ±100°, ±135° (center, front left/right, side surround left/right, rear surround left/right) ±45°, ±135° (45° vertical) (left/right top front, left/right top rear)

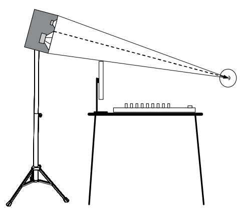
The acoustical axis of the loudspeaker is between the midrange and tweeter drivers.

- ▶ Always point the acoustical axis, in the horizontal and vertical planes, towards the listening position.

The acoustical axis is a line perpendicular to the loudspeaker's front panel along which the microphone was placed to tune the loudspeaker's crossover when the monitor was designed. Pointing the acoustical axis, in the horizontal and vertical planes, toward the listening position or center of the monitoring area will give the best measured and perceived transient response quality.







- ▶ Position the loudspeaker so that there is a direct line of sight from the listening position to the tweeter, midrange, and woofer.
- ▶ Make sure that the bass reflex ports (KH 350) are not blocked or partially covered.
- ▶ Keep sharp edges away from the port outlets, as they can cause air noise.

Connecting audio signals

▶ Always use good quality cables to achieve the maximum cable lengths shown below:

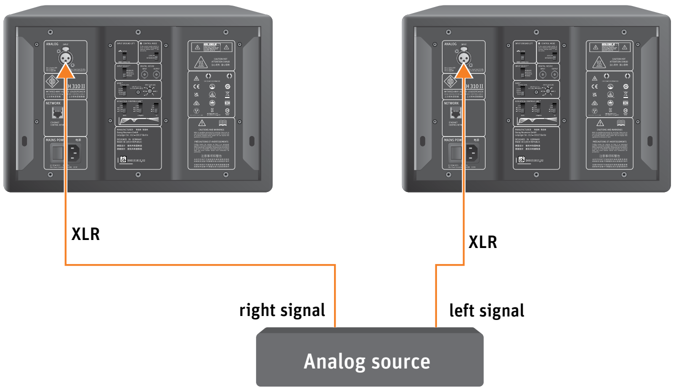
Signal (connector)	Cable Length	Connection method
Analog (XLR)	Up to 100 m	Directly to the ANALOG INPUT socket (XLR) (see below)
Analog (jack)	Up to 100 m	Via an adapter (jack–XLR) to the ANALOG INPUT socket (XLR) (see below)
Analog (RCA)	Up to 10 m	Via an adapter (RCA–XLR) to the ANALOG INPUT socket (XLR) (see below)
Digital (BNC)	Up to 200 m	Direct connection with DIGITAL INPUT socket (BNC) (see below)
Digital (XLR)	Up to 50 m	Via an adapter (XLR–BNC) to the DIGITAL INPUT socket (BNC) (see below)

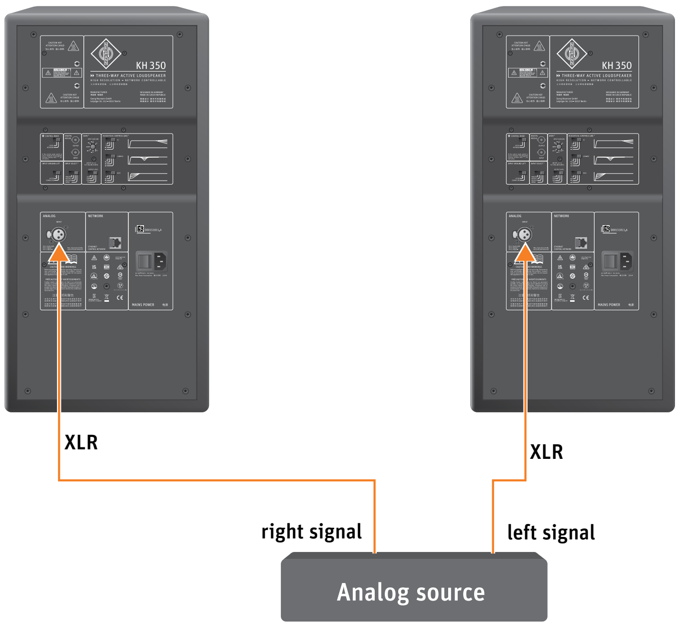
▶ If possible, use a balanced, analog signal connection (XLR, stereo jack) to prevent interference in the cable during an analog connection.

Connecting XLR or RCA cables

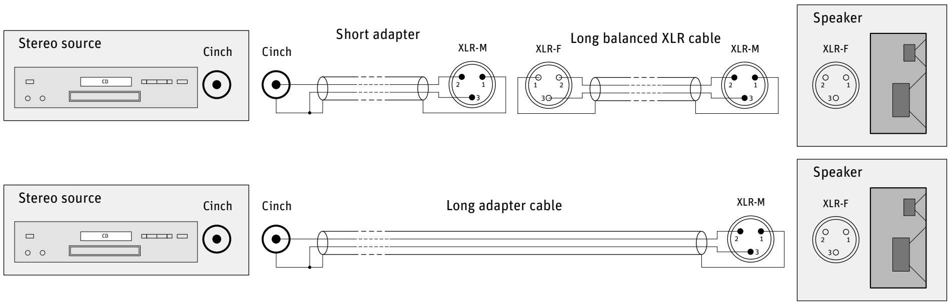
Connecting analog signals to the loudspeaker

▶ Connect the left and right output of your analog audio source to the XLR input sockets of the respective loudspeaker.





- Use an XLR adapter (not supplied) to connect unbalanced cables (e.g. RCA cables).
- Use this adapter directly at the source and connect the adapter via a properly wired balanced XLR cable to the loudspeaker. The connection of pin 3 to ground should be as close as possible to the source to maximize hum rejection on the cable.
- Use the following wiring if you want to make your own RCA-to-XLR cable:



The level delivered by devices with RCA outputs (-10 dBV) is usually less than the studio level (+4 dBu):

- If necessary, use active unbalanced-to-balanced converters in order to be able to connect devices with unbalanced signals.
- We do not recommend passive unbalanced to balanced converters using transformers. They usually limit low frequency levels and increase high frequency distortion.

Connecting digital signals to the loudspeaker

Connecting AES3 cables

- Connect the digital AES3id or S/PDIF output signal of your audio source to the DIGITAL AES3id INPUT socket on the first loudspeaker (see figure below).

i The loudspeaker only supports non-encoded AES3id and S/PDIF signals. Encoded signals such as MP3, DTS or Dolby Digital are not supported. The cable connection must be unbalanced with a characteristic impedance of 75 Ω.

i Only one cable is needed for uncompressed AES3 and S/PDIF digital signals (single-wire mode). They contain two audio channels: “subframe A” and “subframe B.” Usually, the audio channels are:

Subframe A	Subframe B
Left	Right
Center	LFE
Side surround	Side surround
Rear left	Rear right

A clock input is not required because loudspeakers are not audio sources and the converters are clocked to a very stable internally generated clock source.

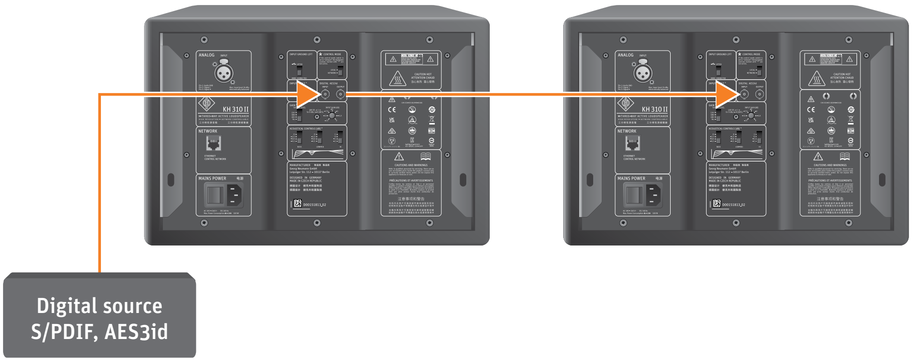
i If the signal source is based on internal digital signal processing, it is recommended that you choose a digital connection between the signal source and the loudspeaker. This eliminates the need for additional signal conversion from digital to analog in the source and from analog to digital in the loudspeaker. This also applies if you are connecting to an upstream DSP subwoofer (e.g. KH 750 DSP, KH 810 II). This should be connected to the loudspeakers via its digital output.

Note that the supplied digital signal often has a maximum signal level, and that the level in the source is often not adjustable. Therefore, before making the digital connection, set the OUTPUT LEVEL switch on the loudspeaker to 94 dB SPL and the INPUT GAIN knob to -15 dB.

- Connect the DIGITAL AES3id OUTPUT socket on the first loudspeaker to the DIGITAL AES3id INPUT socket on the second loudspeaker.
- Set the INPUT SELECT switch on the left loudspeaker to AES3id L and the INPUT SELECT switch on the right loudspeaker to AES3id R. To do so, the CONTROL MODE switch must be in the **Local** position.

Right speaker:
Switch INPUT SELECT: AES3id R

Left speaker:
Switch INPUT SELECT: AES3id L



Source	Ideal	Alternative
Professional interface	XLR	-
Consumer device	DI box → XLR	RCA → XLR adapter
TV	RCA → XLR	Toslink → DAC →XLR
Smartphone	Interface	DI box

Right speaker:
Switch INPUT SELECT: AES3id R

Left speaker:
Switch INPUT SELECT: AES3id L

The settings of the first loudspeaker in the signal chain are not looped through to the second loudspeaker. Each loudspeaker must be configured separately.

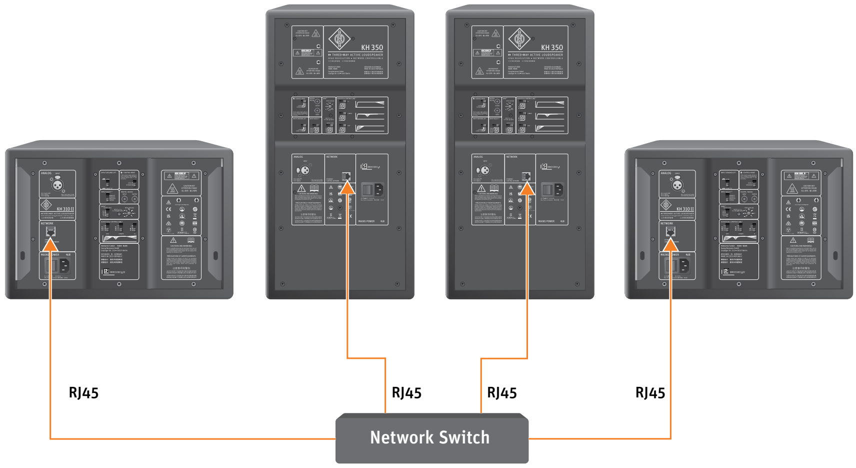
When using a daisy chain connection as shown here, the signal from the second loudspeaker is not delayed relative to the first loudspeaker.

The output of the final loudspeaker does not have need to be terminated with an additional 75 ohms. It is terminated internally.

Connecting network cables

To use the functionality of the separately available MA 1 software, the loudspeaker must be connected to a computer via a network. To do this, use a standard Ethernet cable (Cat 5 or higher; not included with delivery) to connect directly to the network port on the computer or to a network to which the machine is also connected.

For further information, refer to the explanations in MA 1.

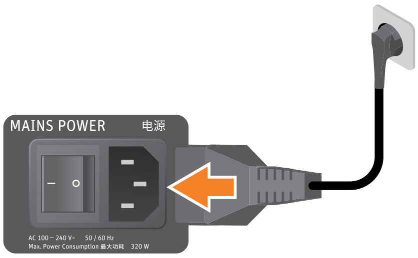


This device supports Sennheiser Sound Control (SSC) based on TCP and IPv6. The static IPv6 address of the device can be determined with mDNS. For more information about SSC, see the Sennheiser website.

Connecting/disconnecting the loudspeaker to/from the power supply system

To connect the loudspeaker to the power supply system:

- Make sure that the on/off switch is set to “0”.
- Connect the IEC connector on the supplied power cable to the power supply socket.



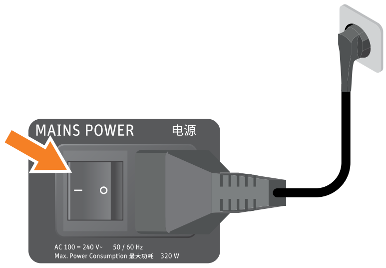
- Connect the wall plug on the mains cable to a suitable wall outlet.

To completely disconnect the loudspeaker from the power supply system:

- Set the on/off switch to “0”.
- Pull the wall plug out of the wall outlet.

Configuring and using the loudspeaker

Switching the loudspeaker on/off



▶ Set the on/off switch to:

- “I” to switch on the loudspeaker. The Neumann logo lights up solid red while the DSP system boots up. After approximately 5 seconds it turns white indicating the loudspeaker is ready to be used. Using the software **MA 1 – Automatic Monitor Alignment** allows you to dim the logo or switch it off completely after the boot phase is complete.
- “OFF” to switch off the loudspeaker. The Neumann logo switches to red for a short moment and then goes off.

There is a 5 second delay before sound can be heard from the loudspeaker. This suppresses any noise that would be generated by connected devices that are switched on at the same time. Conversely, switching off the loudspeaker immediately mutes the audio.

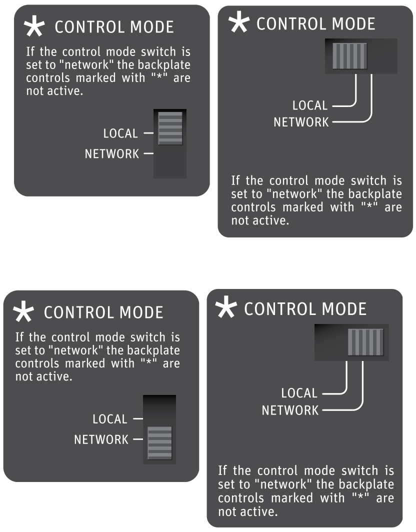
If the devices are switched on or off individually rather than centrally, the following order is recommended:

Switch-on sequence: 1. audio source → 2. subwoofer →3. loudspeaker
Switch-off sequence: 1. loudspeaker → 2. subwoofer →3. audio source

Functionality of the Neumann logo

Action	Logo indication
Firmware activities	
Loudspeaker is booting up, muted	Solid red for approx. 5 seconds
Loudspeaker is shutting down, muted	Brief red light
Loudspeaker boot up error	Red flashing (fast)
Loudspeaker firmware is being updated	Solid pink
Loudspeaker resetting to factory default settings	Pink flashing (very fast)
Loudspeaker is being reset	Red flashing (fast)
Normal operation	
Loudspeaker switched on and ready for operation (can be dimmed using MA 1 – Automatic Monitor Alignment software)	Solid white
Subwoofer system output level is muted (mute activated in MA 1)	Solid pink
Protection and standby	
Peak limiter active	Brief red light in time with the pulse peaks
Thermolimiter active	Solid red as long as level reduction is active
Overload protection	Solid red as long as the loudspeaker is muted to prevent overheating
Other protection system is activated (takes priority over other indications)	Red
Calibration with the MA 1 – Automatic Monitor Alignment software	
Identifying the loudspeaker	Flashing pink (2 Hz)
Loudspeaker is selected	Solid white

CONTROL MODE switch



The CONROL MODE switch toggles between local mode and network mode.

Local mode: operation on the device

If the switch is in the LOCAL position, the loudspeaker will respond only to the controls on the rear. It is operated as described in this manual. The loudspeaker does not respond to network commands from the MA 1 – Automatic Monitor Alignment software.

Network mode: operation via the network

If the CONTROL MODE switch is in the NETWORK position, the loudspeaker will respond only to network commands from the **MA 1 – Automatic Monitor Alignment** software. The controls marked ***** on the back of the loudspeaker are ignored.

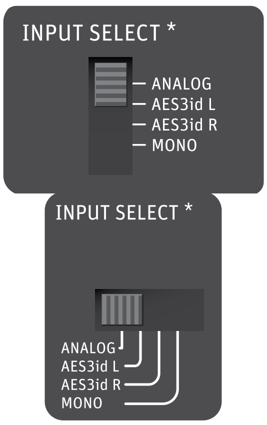
All settings made in MA 1 and transferred to the loudspeaker remain permanently stored in the loudspeaker. This is also the case if the network cable has been removed, the loudspeaker has been switched off and on again, and/or the CONTROL MODE switch has been temporarily set to the Local position.

By switching from network control to LOCAL, you can easily switch between a configuration set in the **MA 1 – Automatic Monitor Alignment** software and settings made directly on the loudspeaker.

This is useful if you want to temporarily use the loudspeaker in a location other than the studio environment you configured.

If the MA 1 – Automatic Monitor Alignment software has not been connected to the loudspeaker before, the default settings will be used in Network mode.

INPUT SELECT



The INPUT SELECT switch toggles between analog and digital input.

Set the INPUT SELECT switch to ANALOG if you intend to feed in an analog signal via the XLR socket labeled ANALOG INPUT.

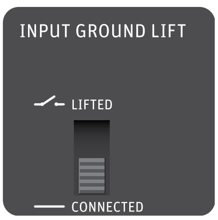
Set the INPUT SELECT switch to AES3id L, AES3id R or MONO if you intend to feed in a digital AES3id or S/PDIF signal via the BNC socket labeled DIGITAL AES3id INPUT.

Set the INPUT SELECT switch on the left loudspeaker to AES3id L and the INPUT SELECT switch on the right loudspeaker to AES3id R for normal stereo mode.

If the loudspeakers are used as surround speakers in larger systems, the channels must be configured based on how they are assigned in the source. In this case, channel A (digital subframe A) corresponds to the left channel and channel B (digital subframe B) corresponds to the right channel.

Switch position	Meaning
ANALOG	XLR socket ANALOG INPUT is active
AES3id L	Digital subframe A, BNC socket DIGITAL AES3id INPUT is active
AES3id R	Digital subframe B, BNC socket DIGITAL AES3id INPUT is active
MONO	Digital subframe A is added to digital subframe B with 4.5 dB attenuation, BNC socket DIGITAL AES3id INPUT is active

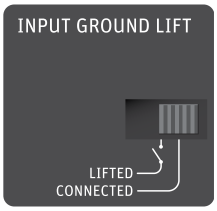
GROUND LIFT



The GROUND LIFT switch internally separates pin 1 of the XLR input socket from the chassis ground of the loudspeaker electronics.

If there is humming or buzzing noise coming from the loudspeaker, first search for the cause of the noise:

▶ Disconnect all input cables from the loudspeaker.



If the noise is no longer audible, it is probably coming from the audio source or input signal cabling. It may be possible to eliminate the noise by disconnecting the ground from the input signals (activating ground lift).

To activate the ground lift:

▶ Reconnect the signal cables and set the GROUND LIFT switch to LIFTED.

ATTENTION
Life-threatening danger!

For safety reasons, the electronics chassis ground is always connected to the ground contact of the power supply.

▶ Never disconnect the ground contact of the power cable from ground.

▶ Even if ground lift is enabled, the power supply ground connection remains connected to the device chassis.

Resetting the loudspeaker settings

You can use the **MA 1 – Automatic Monitor Alignment** software to configure the loudspeaker beyond what is available on the backplate.

To reset these internal loudspeaker parameters to the factory settings:

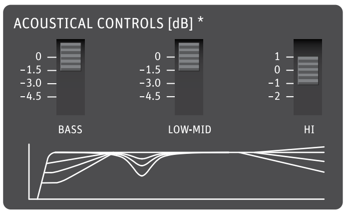
▶ Switch on the loudspeaker.
While the front Neumann logo is solid red during the boot up phase, quickly move the CONTROL MODE switch between the top and bottom switch position until the logo turns white for a few seconds. The logo then flashes red until all values are reset to the default settings. The logo then glows white.

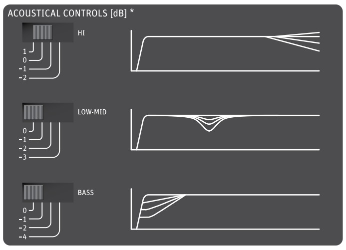
Firmware update

Firmware updates are performed via the **MA 1 – Automatic Monitor Alignment** software. When you open the software, it scans the network for loudspeakers and checks if the firmware is up to date. If a firmware update is required, you will be notified. Follow the instructions shown on screen. It takes approximately 10 seconds per loudspeaker to perform the update.

i The Neumann.Control iPad app is no longer supported and is not compatible with this loudspeaker. Please do not use this app in connection with the loudspeaker.

Adapting loudspeakers to the environment using the switches on the backplate





When the BASS, LOW-MID and HI switches on the ACOUSTICAL CONTROLS panel are set to 0, the loudspeaker is designed to have a flat frequency response in anechoic conditions.

The frequency response will change in your monitoring environment. The loudspeaker's frequency response will also change based on its position in the room. The same loudspeaker installed in different positions in the same room may require different acoustical control settings. In a symmetrical installation, left/right pairs (front or back) will probably have the same acoustical settings.

The loudspeaker is designed to optimally interact with the monitoring environment. The best results can be achieved in conjunction with the MA 1 – Automatic Monitor Alignment system (available separately).

- ▶ If the loudspeakers are not calibrated using MA 1, we recommend calibrating them for the listening conditions using an external measurement system.
- ▶ Repeat this measurement whenever you make changes to the layout of your studio and adjust the ACOUSTICAL CONTROLS accordingly.
- ▶ At your listening position, determine the frequency response of the loudspeakers, one after the other.

i These switches are not intended for adjusting the sound to your preferences during playback. They are used to adapt the loudspeaker to the acoustic conditions of the room and the position in the room to provide as neutral a response as possible.

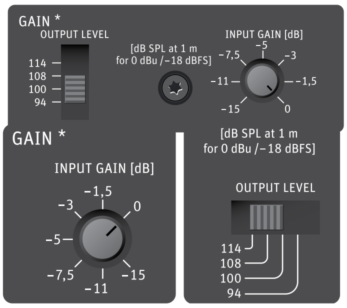
Switch	Function	Possible settings
BASS	Compensates for acoustical loading in the low frequency range due to nearby large solid boundaries (e.g. walls).	KH 310 II: 0, -1.5, -3.0, -4.5 KH 350 : 0, -1, -2, -4
LOW-MID	Compensates for acoustical loading in the low-mid frequency range due to large, reflecting objects (e.g. mixing consoles, tables or screens) in the vicinity of the loudspeaker.	KH 310 II: 0, -1.5, -3.0, -4.5 KH 350 : 0, -1, -2, -3
HI	Compensates for insufficient or excessive high-frequency damping in the room.	+1, 0, -1, -2 dB

The following settings can be used as a starting point for further adjustment:

Loudspeaker position	Switch		
	BASS	LOW-MID	HI
In a corner	KH 310 II: -4.5 KH 350: -4	KH 310 II: -3.0 KH 350: -2	-
Next to an acoustically solid wall (e.g. brick, concrete)	KH 310 II: -3 KH 350: -2	-	-
Next to an acoustically soft wall (e.g. drywall)	KH 310 II: -1.5 KH 350: -1	-	-
Free standing in an untreated room	-2 dB	-	-1 dB
Free standing in a well-treated room	-	-	-
In a small room with strong side wall reflections	KH 310 II: -4.5 KH 350: -4	-	-
Near a small desktop or small reflecting surface*	-	KH 310 II: -3.0 KH 350: -2	-

Loudspeaker position	Switch		
	BASS	LOW-MID	HI
Near a large desktop or large reflecting surface*	-	KH 310 II: -4.5 KH 350: -3	-

* Use these settings in addition to one of the top settings.



Adjusting the volume level

This section describes how to correctly calibrate the listening level in the studio using pink noise, level measurement, and by adjusting the input and output levels of the loudspeaker.

The values described comply with the current standards set forth in the ARD/IRT guidelines, SMPTE, and standard studio reference levels.

Preparation:

Make sure that the following conditions are met:

- The listening position has been defined (sweet spot).
- The quiet noise level in the room is sufficiently low.
- A sound level meter according to IEC 61672 is available and set to:
  - **C weighted**
  - **Slow**
- On all loudspeakers used, set the OUTPUT LEVEL switch to the lowest value of 94 dB SPL and the INPUT GAIN knob to −15 dB.

Test signal:

Use a **broadband pink noise** with calibrated RMS level:

- **Europe (Broadcast / ARD/IRT): −18 dBFS RMS**
- **USA (SMPTE / film): −20 dBFS RMS**

Play back the signal **for each loudspeaker** individually.

Implementation:

- **Play a test signal**
  - Play the pink noise at the appropriate level (−18 / −20 dBFS).
- **Measure the sound pressure level**
  - Position the sound level meter at the listening position (ear height, 0° orientation).
  - Record the level in **dB SPL(C), Slow.**
- **Level out the loudspeaker**
  - Adjust the **OUTPUT LEVEL switch** and the **INPUT GAIN control** of the loudspeaker to achieve the desired reference level.

Recommended reference level:

	Level display		Analog input level		Digital level	Listening level (for each loudspeaker)
	dB	%	dBu	VRMS	dBFS	dB SPL(C)
Permitted maximum level (ARD/IRT)	0	100	6	1.55	-9	91-92
Reference level (ARD/IRT)	-9	35	-3	0.55	-18	82-83
Studio standard USA (film/SMPTE)	0	100	4	1.23	-20	85
Studio standard USA (music/K system)	-	-	4	1.23	K-20: -20 K-14: -14	K-20: 85 K-14: 77
Consumer	-	-	-7.78	0.32	-	-

- Make sure that the RMS level of the test signal was measured correctly. Many DAWs display peak values instead of RMS.
- The value per loudspeaker is taken as the measurement, not the combined volume.
- When playing back via both channels, the level increases by approximately +3 dB if the noise is uncorrelated.

If your signal source is not used as a reference for the listening level, make sure that the level of the source is higher and the level on the loudspeakers is lower. This reduces the self-noise from the source and any extraneous signals in the input line.

Configuring standby mode

Standby means that the network interface, signal processing circuitry and power amplifiers are all powered down. Standby mode is automatically deactivated when a sufficiently large audio signal is detected at the analog input. The time taken to resume normal operation and hear sound is 5 seconds.

When the INPUT SELECT switch is in the ANALOG position, the loudspeaker switches to standby mode after 20 minutes without an input signal or with only a very low input level.

For the standby function to be activated, the incoming signal must not cause the loudspeaker's output level to exceed an internally specified threshold for the specified standby time. This therefore depends on both the amplitude of the input signal and the level setting on the loudspeaker itself. The level threshold is approx. 20 dB SPL at a distance of 1 m in free field conditions.

Even if there is no input signal, however, interference and crosstalk in the signal source or input line can cause the loudspeaker to wake up or prevent the loudspeaker from switching to standby mode in the first place.

In positions AES3id L, AES3id R, and MONO of the INPUT SELECT switch, the loudspeaker switches to standby mode if no valid clock signal from a digital source is present for more than 20 minutes. The loudspeaker wakes up from standby mode as soon as a valid digital signal clock is present again.

The sampling rate of the AES3id signal must not change while standby mode is active.

The power consumption of the loudspeaker is significantly reduced in standby mode (from over 20 W to less than 0.5 W). This not only reduces the energy required and thus the associated costs, but also the heat loss, which reduces the temperature increase in the room and leads to a lower load on the electronic components in the device, thereby contributing to its longevity.

We therefore recommend that you leave the automatic standby function enabled at all times, including in NETWORK mode.

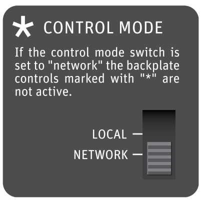
Standby in LOCAL mode

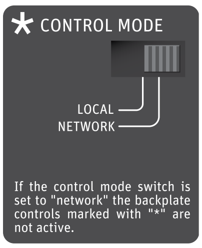
▶ Set the CONTROL MODE switch to **Local**.

The idle time for standby mode is 20 minutes.

The output level depends on the positions of the OUTPUT LEVEL switch and the INPUT GAIN knob on the back of the loudspeaker. Set them as required for your setup.

Standby in network mode





▶ Set the CONTROL MODE switch to **Network**.

In the Network switch position, the standby behavior is determined by the settings in the MA 1 – Automatic Monitor Alignment software.

Customizing standby behavior

Standby is too sensitive

If standby is too sensitive, the loudspeaker will not enter standby mode when it is supposed to, or it will wake up from standby mode when it is not supposed to.

Possible reasons:

The source may contain extraneous noise or noise peaks that wake the loudspeaker up or prevent it from going into standby. This can also cause loudspeakers with the same settings to behave differently.

You can determine if extraneous noise or noise peaks are responsible for the behavior as follows:

▶ Set the CONTROL MODE switch to LOCAL.

▶ Set the OUTPUT LEVEL switch to 114 dB to listen for possible extraneous noise.

▶ Listen carefully to identify noise peaks.

▶ Alternatively, you can record the loudspeaker output signal with a microphone and analyze the recording.

▶ Try to find out whether switching other devices in the house on or off has an effect (e.g. refrigerator or similar).

You can check that the standby feature is working correctly by connecting a short XLR cable to the loudspeaker input without connecting a device to the other end of the cable.

▶ In backplate mode, set the OUTPUT LEVEL switch to 100 dB.
The loudspeaker should enter standby mode after 20 minutes.

i Since the loudspeaker continuously monitors the input signal, peaks that come from the source or are induced in the cable can also prevent the loudspeaker from switching to standby mode. Make sure that there are no signal peaks coming from the source or induced in the cable that can wake up the loudspeaker.

Standby is not sensitive enough

If standby is not sensitive enough, the loudspeaker goes into standby mode when it is not supposed to, or does not wake up from standby mode when it is supposed to.

Possible reasons:

The standby threshold is above the signal level. If the input and output levels of the loudspeaker are set very low, but the source has a high level, this can result in a very quiet output signal.

LOCAL mode:

▶ In backplate mode, increase the source output level or the input and output levels of the loudspeaker in order to raise the sound pressure level above the standby threshold.

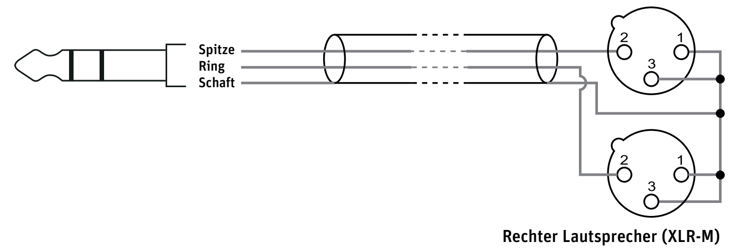
Network mode:

▶ In network mode, increase the source output level or the input and output levels of the loudspeaker in order to raise the sound pressure level above the standby threshold.

In general, you should set the loudspeaker level as low as possible (e.g., the INPUT GAIN knob to 0, the OUTPUT LEVEL switch to 94 or 100 dB SPL) and the level of your source as high as possible to obtain the best possible signal-to-noise ratio and minimize induced extraneous noise.

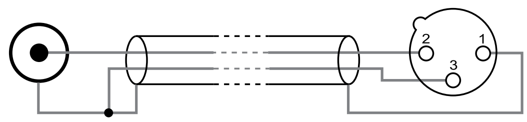
Ideally, the source should be connected to the loudspeaker using a balanced XLR cable (XLR to XLR or jack to XLR). If only an unbalanced source is available, you should connect it as shown in the following figures.

Headphone output (TV or hi-fi system): miniature jack (3.5 mm) or jack (6.3 mm):



Verbindungen zwischen Stereo-Klinke (unsymmetrisch) und XLR (symmetrisch)

RCA line output from a television (if the output level is adjustable) or RCA output (pre-amp) from an AV receiver. One cable is required per speaker:



Verbindungen zwischen Cinch (unsymmetrisch) und XLR (symmetrisch)

Cleaning and maintaining the loudspeaker

CAUTION
Damage to the product caused by liquids!

Liquids entering the product can cause a short-circuit in the electronics and damage or even destroy the product.

▶ Keep all liquids away from the product!

▶ Before cleaning, disconnect the product from the power supply.
▶ Use a soft, dry, and lint-free cloth to clean the product.
▶ Be careful not to accidentally adjust the controls when cleaning.
▶ Under no circumstances should you touch the loudspeaker diaphragms.

Troubleshooting

Problem	Cause	Solution
The Neumann logo is off, no sound is heard from the loudspeaker.	The loudspeaker's main internal fuse has blown.	Have the product checked by an authorized Neumann service partner. There is no plug fuse in the device.
The Neumann logo is off or not clearly visible, but sound is heard from the loudspeaker.	The Neumann logo is switched off or dimmed.	Connect the loudspeaker to the MA 1 – Automatic Monitor Alignment software and adjust the logo settings there.
There is a hum or buzz coming from the loudspeaker when an analog audio cable is connected.	A cable is defective, the cabling is bad, there is ground loop in the installation or the level of the audio source is too low.	Check the cabling, especially if unbalanced cabling has been used – see the cable wiring diagram on page 9. Use gold-plated connectors. Do not run signal cables parallel to power cables. Set the output level of the loudspeaker as low as possible and set the output level of the audio source as high as possible, without causing it to clip. Set the GROUND LIFT switch to the Lift position.
The loudspeaker sounds very "thin" in the bass. The low frequency response is very low.	Incorrect wiring in the audio cable or adapter.	Check the cabling, especially if unbalanced cabling has been used – see the cable wiring diagram on page 9.
	One loudspeaker is out of phase with the other. This results in cancellation in the low frequency range.	Check the cabling, especially if unbalanced cabling has been used – see the cable wiring diagram on page 9. Check the settings in the signal source.
Standby is too sensitive or not sensitive enough.	Incorrect standby settings, or noise peaks / extraneous noise in the source.	Check whether there is noise or interference spikes in the source or input line – see „Configuring standby mode" on page 20.
The loudspeaker is behaving abnormally. None of the above will solve the problem.	The microcontroller has frozen or contains corrupt data.	Reset the microcontroller: see "Resetting the loudspeaker settings."

For further information, please refer to the "Questions & Answers" section on the product page at www.neumann.com.

Technical data

For a complete list of specifications, see the loudspeaker's product page at www.neumann.com.

Product characteristics	KH 310 II	KH 350
Power supply	100 to 240 V~, 50/60 Hz	
Power consumption standby / idle / full load	<0.3 W / 26 W / 320 W	
Dimensions (H x W x D)	253 x 383 x 292 mm 10" x 15 1/8" x 11 1/2"	545 x 280 x 363 mm 21 1/2" x 11" x 14 1/4"
Weight	13 kg	20 kg
Drivers (woofer, midrange, tweeter)	210 mm (8 1/4"), 75 mm (3"), 25 mm (1")	210 mm (8 1/4"), 75 mm (3"), 25 mm (1")
Temperature		
Operation and storage, unpacked	+10°C to +40°C	
Transport and storage, packed in original packaging	-25°C to +60°C	
Relative humidity		
Operation and storage, unpacked	Max. 75% (non-condensing)	
Transport and storage, packed in original packaging	Max. 90% (non-condensing)	

Acoustical measurements

Additional specifications, such as acoustical measurements, can be found on the product page for the loudspeaker at www.neumann.com.

Trademarks

Neumann® is a registered trademark of Georg Neumann GmbH. The following are trademarks of Georg Neumann GmbH:

- Plane Wave Bass Array™
- PWBA™
- Neumann.Control™

iPad® is a trademark of Apple Inc., registered in the U.S. and other countries.

Other company, product, or service names mentioned in this operating manual may be the trademarks, service marks, or registered trademarks of their respective owners.

Accessories for KH 310 II

Product	Description
LH 25 II	Mounting bracket for mounting on the wall/ceiling and on stands
LH 28	Tripod stand adapter (for 35 mm tripods)
LH 29	TV spigot (lighting stand adapter)
LH 37	Subwoofer adapter
LH 41	Base plate
LH 43	Surface mounting plate
LH 45	"L"-shaped wall bracket
LH 46	Adjustable ceiling drop adapter
LH 47	Mounting adapter plate
LH 48	Tripod adapter plate
LH 67	Table stand for KH 310 II
GKH 310 II	Protective grille

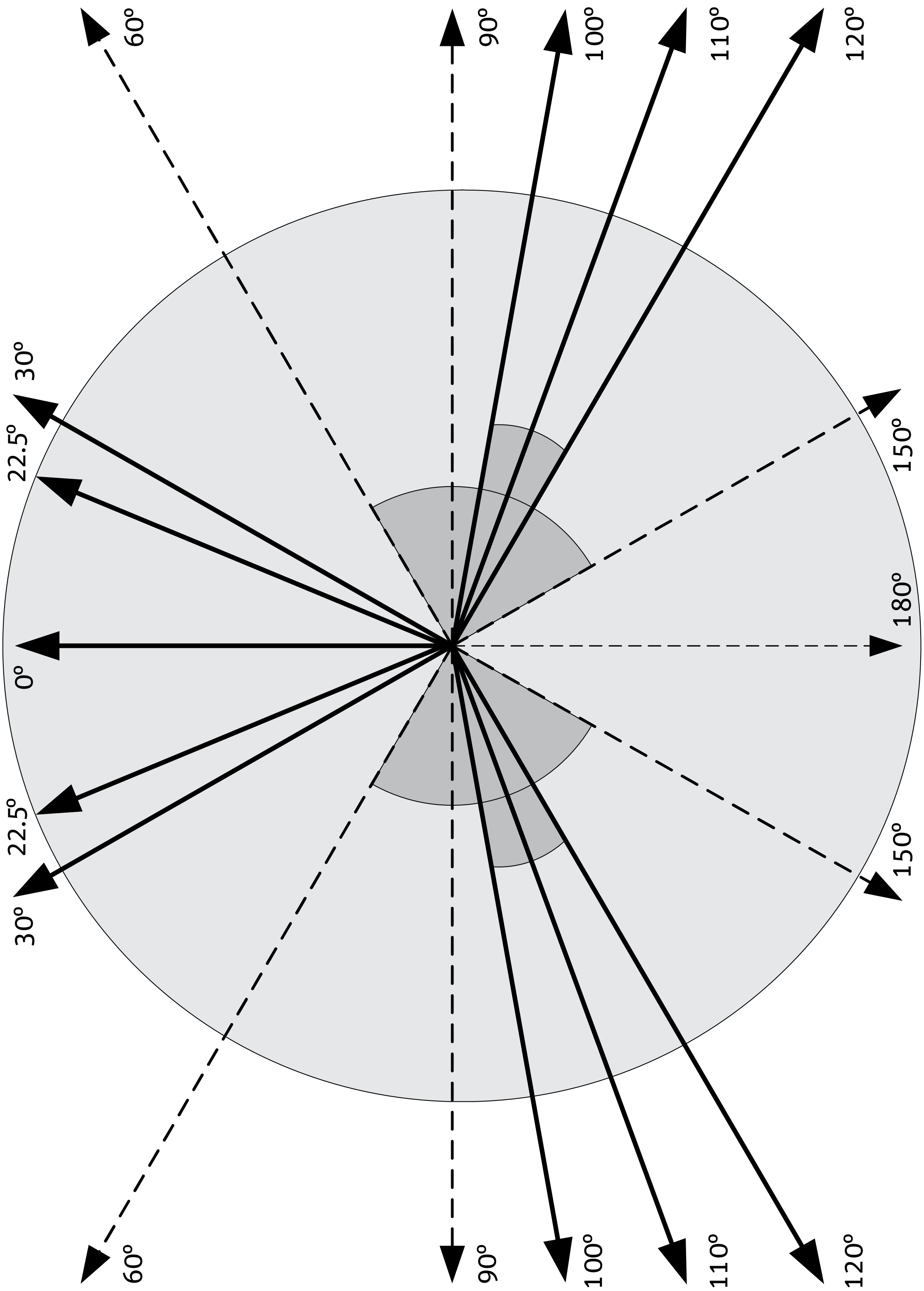
i For a full list of mounting hardware, see the **Mounting Matrix** at www.neumann.com

Accessories for KH 350

Product	Description
LH 28	Tripod stand adapter
LH 29	TV spigot (lighting stand adapter)
LH 30	Mounting bracket
LH 37	Subwoofer adapter
LH 41	Base plate
LH 43	Surface mounting plate
LH 45	"L"-shaped wall bracket
LH 46	Adjustable ceiling drop adapter
LH 47	Mounting adapter plate
LH 48	Tripod adapter plate
GKH 350	Protective grille

i For a full list of mounting hardware, see the **Mounting Matrix** at www.neumann.com

Installation angles



Technical information & glossary

Absolute level

In Europe, the absolute level of 0 dBu is –18 dBFS (EBU standard R68). In the US, +4 dBu is –20 dBFS (SMPTE standard RP155). These dBu values should lead to the following sound pressure levels:

Application	Sound pressure level
Film	85 dB(C)
Broadcast	79 dB(C) (reference level)
Music	No defined reference levels

Near field loudspeakers can be as close as 1 m from the listening position, whereas loudspeakers in a Dolby® certified movie mixing room should be at least 5 m from the listening position.

In the examples below, it is assumed that the listener is inside the room radius and thus the sound field decays according to 20 $\log_{10}$ (*r*) (level reduction of 6 dB as distance doubles). However, this may not always be the case.

Absolute voltage level of input signal	0 dBu (0.775 V)	+4 dBu (1.23 V)
Setting SUBWOOFER GAIN | INPUT GAIN ⑫	–1 dB	–5 dB
Setting SUBWOOFER GAIN | OUTPUT LEVEL ⑪	100	100
Listening distance [m] (dB change)	5 m (–14 dB)	5 m (–14 dB)
Measured output level in dB SPL at 1 m	85 dB SPL	85 dB SPL
Maximum input signal before activation of the protection system	17 dBu	17 dBu

Absolute acoustic level calibration for signal channels is generally achieved using a sound pressure level meter set to “C-weighted” and “Slow”. Play a broadband pink noise test signal set to –18 dBFS (Europe) or –20 dBFS (USA) on the meters of the mixing console and measure the sound pressure level at the listening position. Then adjust each channel’s source level, not the loudspeakers and subwoofer(s) so that the above stated sound pressure levels are achieved.

Acoustical axis

The acoustical axis is a line perpendicular to the loudspeaker’s front panel along which the microphone was placed to tune the loudspeaker’s crossover when the monitor was designed. The acoustical axis runs between the mid-range and tweeter.

Acoustical output level

The loudspeaker provides an output level of 100 dB SPL at a distance of 1 m under free-field conditions when an analog input signal with a level of 0 dBu is applied in its frequency response range. The output level of the loudspeaker changes depending on its position in the room.

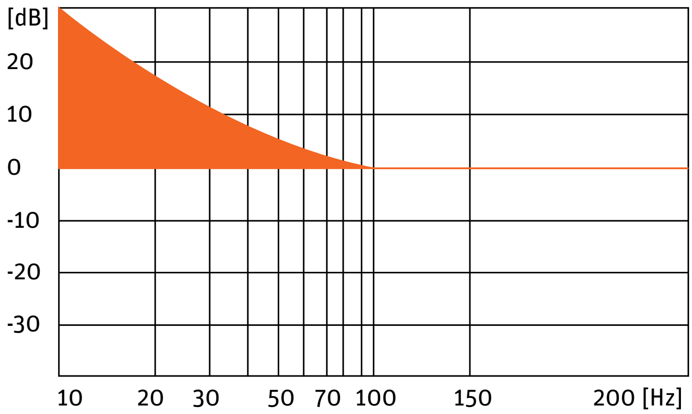
In the free field, the output level decreases by 6 dB each time the distance doubles. The conditions in a room are significantly different. In this case, a considerably lower level decrease is to be expected.

Acoustical response

Neumann loudspeakers are designed to have a flat frequency response in anechoic rooms. When a loudspeaker is installed in a listening environment, the frequency response changes. This depends on the loudspeaker's location and the dimensions and behavior of the room and is likely to be different for the same loudspeaker type positioned in different places in the same room. Even moving the loudspeaker by as little as 50 cm can drastically alter the frequency response, necessitating adjustments to restore a linear response.

The best way to achieve this is to use the MA 1 – Automatic Monitor Alignment software, which optimally adapts the loudspeaker to the environment.

One factor that affects the frequency response of the loudspeaker is room gain. This causes the level at low frequencies to rise by approx. 12 dB/octave in enclosed spaces. The cut-off frequency of this effect is inversely proportional to the room size. So, the smaller the space, the higher the frequency at which the level rises at low frequencies.



Acoustical controls

The acoustical controls on the backplate of the loudspeakers are low-order DSP filters designed to compensate for some of the acoustical issues commonly found in listening environments. Their settings depend on the installation location and the acoustical properties of the room. They are not intended for adjusting the sound to personal taste, but for adapting the loudspeaker to the acoustic conditions of its installation location in order to achieve the most neutral reproduction possible. This is generally not achieved when all acoustical controls are set to the "0" position.

Bass management

Bass management refers to the handling of low-frequency signal components of the channels available on the subwoofer.

Bass management separates the low frequencies (<80 Hz) of all available main channels from the higher frequencies (>80 Hz), adds them up, and passes them to the subwoofer.

The signal components above 80 Hz are forwarded directly to the connected loudspeakers.

Furthermore, the low-frequency effects channel LFE is also added to the bass component of all channels without filtering and is reproduced via the subwoofer. This thus transmits the frequency range of all main input channels to the cut-off frequency of 80 Hz, as well as the complete spectrum of the LFE channel.

Free field	The term "free field" is used to refer to a space or area in which sound propagates exclusively in a direct manner – without significant reflections from walls, the floor, or the ceiling. Typically, this will be an open outdoor environment or a specially designed anechoic room. Free-field measurements enable a precise assessment of the acoustic behavior of a loudspeaker, since only direct sound is evaluated. This makes it possible to determine frequency responses, radiation characteristics, and sound level distributions under idealized, reproducible conditions. Consequently, specifications such as "free-field frequency response" always refer to this anechoic measurement situation.
	A free-field environment is not a desired listening condition in studio acoustics, as it is not representative of the acoustic conditions under which the program material will ultimately be played back. Instead, in addition to various other aspects, the acoustics in the playback room should provide a constant, room-volume-dependent reverberation time down to low frequencies and minimize direct reflections.
Loudspeaker	In this manual, loudspeaker refers to wide-range studio monitors. In conjunction with the subwoofer, they act as a satellite.
	When operating with the subwoofer and with bass management enabled, the signal on each channel is split into the low frequency (<80 Hz), which is played back through the subwoofer, and the high frequency (>80 Hz), which is heard through the loudspeaker.
Power amplifier	The loudspeaker's power amplifiers feature high efficiency, resulting in lower power loss and, consequently, reduced heat dissipation. They operate in bridged mode, which further reduces distortion.
LFE channel	"Low Frequency Effects" (Dolby®) or "Low Frequency Enhancement" (dts®). The LFE channel has a limited bandwidth. Because of the limited frequency range of the LFE channel, it is referred to as ".1" when describing, for example, a 5.1 or 7.1.4 system. The designation "LFE channel" and the format specifications (5.1, 7.1, 9.1.4, etc.) always refer to the source channels and not to the loudspeaker arrangement.
Protection system	A comprehensive protection system prevents damage to the subwoofer if excessive levels are requested. The logo display will then change from white to red, indicating that the protection system is active. In this case, reduce the input signal level. If this happens regularly, use a larger subwoofer with a higher possible maximum level or add more subwoofers to the system to increase the LF headroom.
	The protection system consists of peak limiters that prevent the amplifier from starting to distort when the levels are too high and the woofer from reaching its maximum excursion and becoming distorted or damaged. It also incorporates thermal protection circuitry that protects the driver from excessive continuous stress and protects the electronic system from overheating. The protection system cannot protect the subwoofer during excessive use, e.g., prolonged operation of the subwoofer while the logo is glowing red. Avoid excessive use of the subwoofer so as to not compromise the product's long service life.
Subwoofer	The subwoofer is the bass loudspeaker described in this manual, through which the bass component of all main channels and the LFE channel are played back.

Subwoofer crossovers

Using filters, the crossover divides the input signal of each main channel into two bands for reproduction by the subwoofer or the main loudspeakers. With Neumann subwoofers, the crossover frequency between subwoofers and loudspeakers for the main channels is 80 Hz, and the slope is 24 dB/octave. This frequency was chosen to achieve the best compromise between conflicting requirements:

on the one hand, a high crossover frequency to relieve the main loudspeakers of their low-frequency duties while reducing distortion and having the lowest possible group delay, and on the other hand, the need for a low crossover frequency to minimize the chances of localizing the subwoofer, which provides greater flexibility when positioning the subwoofer in the room. 80 Hz is the best compromise for Neumann studio loudspeakers and subwoofers.

In addition, choosing 80 Hz ensures compatibility with the replay conditions commonly found in consumer products.

Drivers

The electrical signal is converted into audible vibrations by the drivers or loudspeaker chassis. Three drivers are used to reproduce the different frequency ranges: woofer, midrange, and tweeter.