

Using the BHM binaural head microphone

Introduction	1
Recording with a binaural head microphone	2
Equalization of a BHM recording	2
Individual equalization curves	5
Using the equalization curves	5
Post-processing in ArtemiS SUITE	6
Playback of a BHM recording	7
Application example: BHM recording in a vehicle	7

Introduction

HEAD acoustics offers a range of binaural audio sensors for recording sound events in different environments in a way that allows aurally accurate playback. This means that the recordings are made in a way that ensures that they – when played back with the correct playback equalization – give a listener the same impression as if he were present in the original sound field.

These binaural sensors include, for example, the artificial head HMS (HEAD Measurement System) and the Binaural Head Microphone (BHM). The BHM allows binaural recordings to be made in places where an artificial head measurement system cannot be used. This is the case, for example, with recordings to be made on the driver’s seat in a moving vehicle. Here it is impossible to use an artificial head measurement system. Instead, a binaural head microphone can be worn by the driver, which measures the sound directly at the entrance of the driver’s ear canals (see figure1).



Figure 1: Using the BHM

Recording with a binaural head microphone

A frequent application for a binaural head microphone is recording in a vehicle. In order to obtain reproducible results, the following must be observed when making recordings with the BHM:

Particularly in a complex acoustic environment as is a vehicle cabin, the positioning of the microphones has a significant influence on the recording. For example, the signal level can be very different if the car seat is moved slightly forward or backward. Basically, neither of the two recordings is wrong, as both represent the sound pressure level at the respective location of the microphones¹. However, to ensure reproducible results, the recording position must be documented in detail and must be restored exactly for repeated measurements that are supposed to be comparable.

Another important factor is who wears the BHM. Different acoustic properties of the people wearing the BHM and how they put it on may differ as well. To make comparable recordings, e.g., for a listening test, all recordings of a test series should be made by the same person. Recordings made this way can be compared easily even by inexperienced listeners.

Another important precondition is that the person wearing the BHM must try to avoid any interfering noise during the recording. Unlike an artificial head, which does not produce any sound, the human body is in almost permanent motion, so interfering noise can be caused by friction of clothing, movements of the BHM cable or even the ticking of a watch on the wearer's wrist. While the wearer might not even notice these sounds during the recording, they can be very disturbing when listening to the recording later and might even lead to the necessity to repeat the recording.

For an optimal signal level of the recording, it is important to configure a suitable measurement range for the BHM in the recording software, as is done with any other measurement microphone. This ensures that the recorded signal is not overmodulated (clipped) and that the sound to be recorded is not covered by the inherent noise of the measurement chain.

An experienced measurement engineer will quickly develop the required routine to meet all these conditions.

Equalization of a BHM recording

As with artificial head recordings, BHM recordings must be equalized for analysis and playback. For analysis, the BHM signal is equalized during the recording. The goal of this recording equalization is to obtain a signal for the analysis that is comparable to a measurement microphone recording. That way you can analyze a signal that does not contain the acoustic characteristics of the specific recording sensor, but rather an equalized signal that can be examined just like a measurement microphone recording.²

For a correct equalization of the BHM during the recording, the ID equalization profile is available. The ID equalization (Independent of Direction) filters the direction-dependent components of the transfer function out of the signal. This equalization was developed, because sound fields in the real world rarely comply with the conditions of the standardized diffuse field or free field. The ID equalization should be used for all sound fields that are neither a diffuse field nor a free field, e.g., for the sound field within a vehicle cabin. The equalization filter is individually generated for each BHM specimen at the factory and can be used in various ways.

1. The easiest possibility is using the BEQ II.1 binaural equalizer. The BEQ II.1 can be programmed with the custom equalization curve of a BHM, so that a level-accurate and ID-equalized BHM signal is present at the output. Programming is done at the factory by HEAD acoustics, e.g., when a BHM and a BEQ II.1 are shipped together. However, the programming can also be performed later. To obtain a correctly equalized signal, it is important to use the specific BHM, which the BEQ II.1 was programmed for. Each BEQ II.1 can be programmed for

¹ The differences caused by the fact that the BHM was worn by different persons are described in the Application Note "Comparability of Recordings made with Measurement Systems from HEAD acoustics".

² A detailed description of the recording equalization can be found in the Application Note "Binaural measurement, Analysis and Playback".

one specific BHM only. The serial number of the matching BHM can be found on the BEQ II.1 housing.

The ArtemiS SUITE Data Acquisition Module (HEAD Recorder) can be configured for using a BHM along with a correctly programmed BEQ II.1 as follows: In the **View** menu of the HEAD Recorder user interface, select **Hardware Properties**. In the dialog that appears, select the **BHM** item under **Configuration** (see figure2). This will automatically set the **Equalization** parameter to **ID**. Furthermore, the appropriate sensors are connected automatically in the channel list of the HEAD Recorder. With this configuration, the ID equalization programmed on the BEQ II.1 is used for the recording, and the BHM signal is equalized correctly.

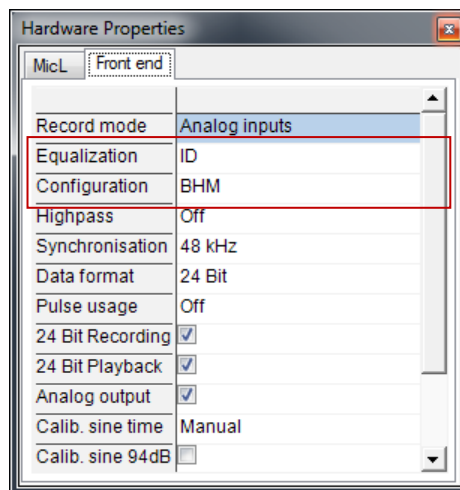


Figure 2: Hardware settings in the HEAD Recorder

- Another possibility is to configure the equalization via the software used. To do this in the HEAD Sensor Explorer³, the custom equalization filter for the BHM must be included in the **Filter File** field when defining the BHM sensor (see figure 3, next page). This custom equalization filter file is included on the equalization CD shipped with the BHM and must be entered in the corresponding fields of the Sensor Configuration for both channels separately. In recordings made with ArtemiS SUITE Data Acquisition Module (HEAD Recorder), this custom filter with the ID equalization is automatically used for the BHM channels. An additional equalization activated via the **Equalization** field would cause double filtering and distort the recorded signal. To prevent this, this field is grayed out automatically as soon as an equalization filter has been entered in the **Filter File** field. Furthermore, when generating a new BHM sensor, the sensitivity of the microphones can be entered in the field **Sensitivity**. The correct value can be found in the calibration certificate, which was delivered with the BHM.

³ As of ArtemiS SUITE 7.0 you can also create BHM sensors in a sensor library. Before you can use a sensor library created with ArtemiS SUITE in the channel list of the HEAD Recorder (as of version 7.0) you have to import the library. One advantage of using sensor libraries is that the equalization curves needed for the correct recording are saved with the sensor library, thus after the sensor creation the EQU file is not needed anymore.

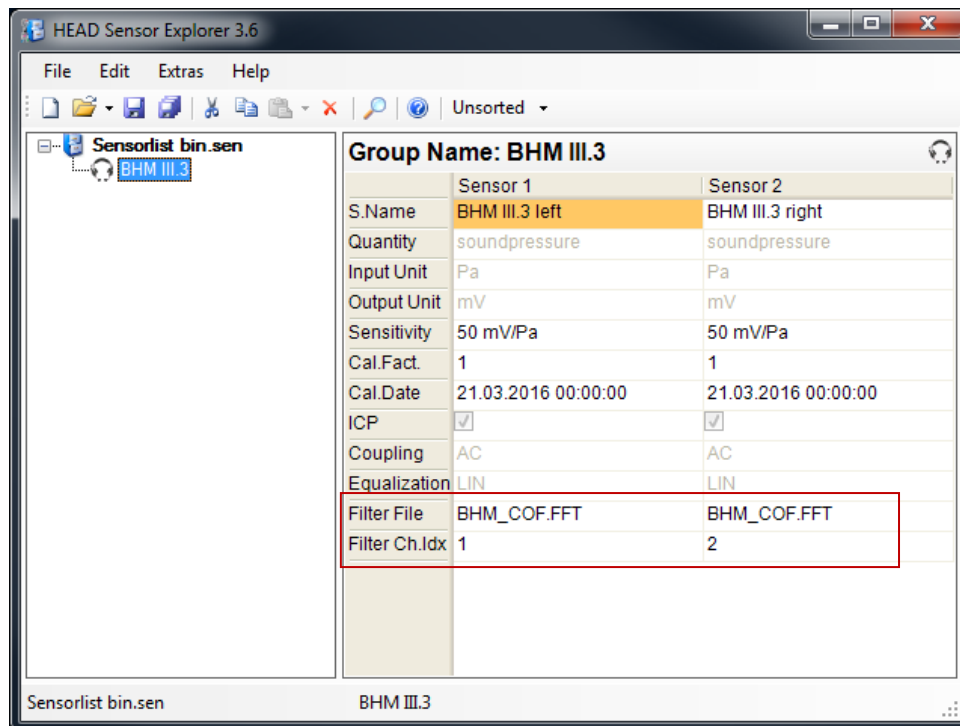


Figure 3: Definition of a BHM sensor in the Sensor Explorer

3. If the equalization has not been performed already during the recording, the signal can also be equalized later in ArtemiS SUITE analysis software using the custom equalization filter. However, such a retroactive equalization should remain an exception. If the equalization is not applied during the recording, the recording level may not be chosen optimally, so that the retroactive equalization can cause clipping of the recorded signal. If the measurement range chosen is too high, the dynamic range of the signal is not optimally used. Furthermore, a retroactive equalization of recordings in 16-bit format can increase background noise. Due to the amplification in the range between 10 and 12 kHz by the ID equalization filter, not only the actual signal is amplified, but also the undesired quantization noise, which is stronger in 16-bit recordings than, for example, in recordings in 24-bit format. In recordings of low signal levels, the amplified noise can then become noticeable. Therefore, especially when recording sound with a low volume, it is recommended to choose the 24-bit format and to equalize the signal during the recording.

For all equalization variants described above, it is important that only the custom equalization filter created specifically for your BHM specimen is used. Only this ensures a correct equalization and makes the measurement comparable to an ID-equalized artificial head measurement or a measurement made with a measurement microphone.

To verify the comparability between an ID-equalized artificial head recording and an ID-equalized BHM recording, it is important to take care of the positions of the sensors. The microphones of the artificial head and the openings of the BHM must be placed in exactly the same position⁴.

⁴ The BHM equalization was developed to ensure the correct equalization of a BHM worn by an adult person, not by an artificial head. Therefore, for a comparative measurement, the BHM must be worn by a real person and must not be placed on the artificial head.

Individual equalization curves

Some examples of individual equalization curves of four different BHM specimens are shown in figure 4. These diagrams clearly show that recordings made with a BHM are significantly amplified around 11 kHz by the equalization filters. This peak is determined by the physical properties of the BHM, such as the tubes and microphones used.

Depending on the BHM specimen, this peak can be very strong, as in the black curve shown in figure 4 (more than 15 dB at 10.5 kHz for the right channel).

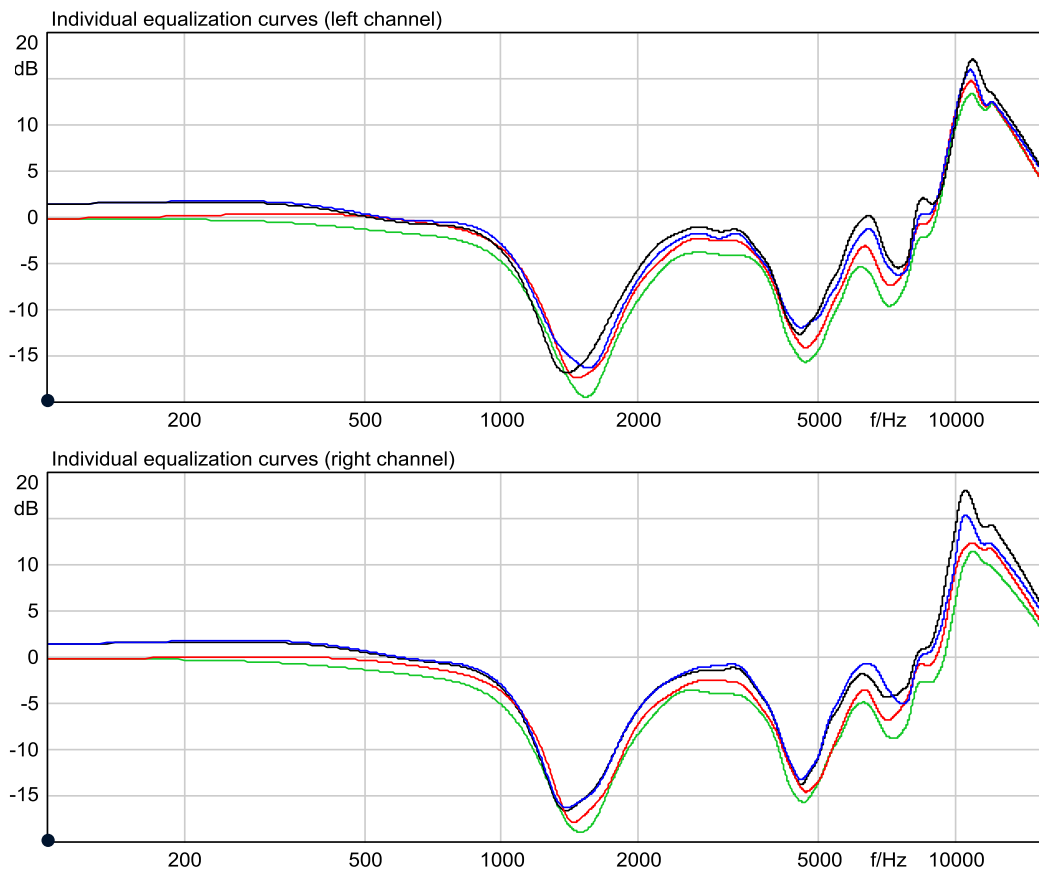


Figure 4: Individual equalization curves of four different BHM specimens

Using the equalization curves

This strong amplification is necessary to ensure not only a correct equalization of the BHM recordings, but also the comparability of both an ID-equalized artificial head recording and a measurement microphone recording. It will not compromise the signal quality in most application cases.

However, with certain signal types, this strong amplification can lead to undesired side effects. This is the case if a very weak signal is recorded with a very high measurement range. In this case, the stronger background noise due to the high range is amplified around 11 kHz to such a degree that it becomes clearly perceptible as interfering noise when listening to the recording later.

This effect can be reduced by suitable means. However, since noise components are always and inevitably recorded along with sound signals due to the physical properties of the microphones, the measurement chain and the digitization of the signals, the effect cannot be avoided completely. The necessary measures described in the following correspond to those for improving the signal quality with retroactive equalization (page 4, section 3).

To keep the unwanted noise components as low as possible, it is important to choose the correct measurement range for the recording. If the range is too small, signal clipping will occur in the recording. If the measurement range is too high, the available dynamic range is not optimally utilized, leading to

an unnecessary amplification of background noise. This will make weak signal components disappear in the noise floor of the recording, and the equalization filter will amplify the noise in certain frequency bands.

The undesired quantization noise is higher with recordings in 16-bit format than, for example, in 24-bit format. When recording weak signals, the amplified noise can sometimes interfere significantly. Therefore, the 24-bit format is recommended especially for recordings of weak signals.

Post-processing in ArtemiS SUITE⁵

If a recording situation requires a high measurement range even when recording weak signals (e.g., if signals with very high and very low levels are to be recorded in rapid succession) and the undesired effect mentioned above occurs, Playback Filters in ArtemiS SUITE are the solution of choice. By using an 11 kHz filter, the interfering signal components can be eliminated in a controlled way, allowing the listener to concentrate on the relevant signal.


ArtemiS SUITE offers the Playback Filters in the Mark Analyzer. After loading the signal into a Mark Analyzer with the desired analysis functions, activate the Playback Filters by clicking on the  button and afterwards select the desired filter. You can choose between a band-stop filter and a parametric band-pass filter. With the band-stop filter, the interfering signal components can be eliminated completely. However, if the signal also contains useful components around 11 kHz, the relevant signal will be significantly modified by such a filter as well. This can be avoided by using a parametric band-pass filter, where the attenuation is adjustable. With such a filter, the attenuation can be set to 6 dB, for example, thus reducing the effect on the relevant signal.



Figure 5:

Band-stop filter

Parametric band-pass filter

In addition to the Playback Filters, a signal can certainly be filtered permanently, too, and saved in a new file. To do so, drag & drop the filter used for the playback to the Filter Pool. Afterwards, the time domain signal with the activated filter can be saved as a new file by using the HDF export. The filtered time signal can certainly be exported as wave file and embedded into a PowerPoint⁶ presentation or the like.

⁵ The descriptions in this Application Note refer to version 9.0 of ArtemiS SUITE. The general proceeding also applies to other versions. However, the scope of functions and the user interface may differ.

⁶ PowerPoint is a registered trademark of the Microsoft Corporation.

Playback of a BHM recording

To play back a BHM recording, playback equalization must be applied. It ensures that the signal played through a headphone gives you the same acoustic impression as if you were present in the original sound field.

BHM recording in a vehicle



+

Playback via PEQ and headphone



△
(when using the correct ID equalizations for recording and playback)

Sound impression in the original sound field



Figure 6: Recording and playback of a BHM recording

For the playback of BHM recordings, HEAD acoustics offers the programmable head-phone equalizer *labP2*. Like its predecessor PEQ V, it is programmed with all required ID filters for playback equalization, so that BHM recordings can be played back with the correct equalization in order to achieve an acoustic impression resembling the original sound field.

However, an accurate playback level and the correct equalization can only be guaranteed if the *labP2* unit has been calibrated at the factory and the resulting custom equalization filters for the headphone specimen used are installed. Furthermore, this equalizer contains filters that allow possible variations of the headphone transfer characteristics to be compensated for. The number above the headphone sockets of the *labP2* is the serial number of the headphone for which this output delivers the correct calibration and equalization.

The easiest way to playback a BHM recording is to use the analysis software ArtemiS SUITE, a *labP2* and a headphone. With ArtemiS SUITE 6.0, the playback functionality was enhanced significantly. Now, the playback equalization as well as the level are set automatically taking into account the selected playback, protection and comfort parameters. The following application example illustrates the setting possibilities that are available as of ArtemiS SUITE 6.0. A complete description of the setting possibilities for playing binaural signals can be found in the application note "Binaural Measurement, Analysis and Playback"⁷.

Application example: BHM recording in a vehicle

In the following example, a BHM with ID equalization was used for the recording. The recording was saved directly to the hard disk of the computer and is now available with the correct equalization for analysis and playback.

Figure 7 shows the analysis *FFT vs. Time* of the vehicle interior noise recording. The FFT analysis clearly shows that the signal is passing a resonance between 8.5 and 13.5 seconds. This resonance is stronger on the left channel than on the right one.

⁷ You can download this application note in our Download Center:
http://www.head-acoustics.de/eng/nvh_application_notes_use_of_systems.htm.

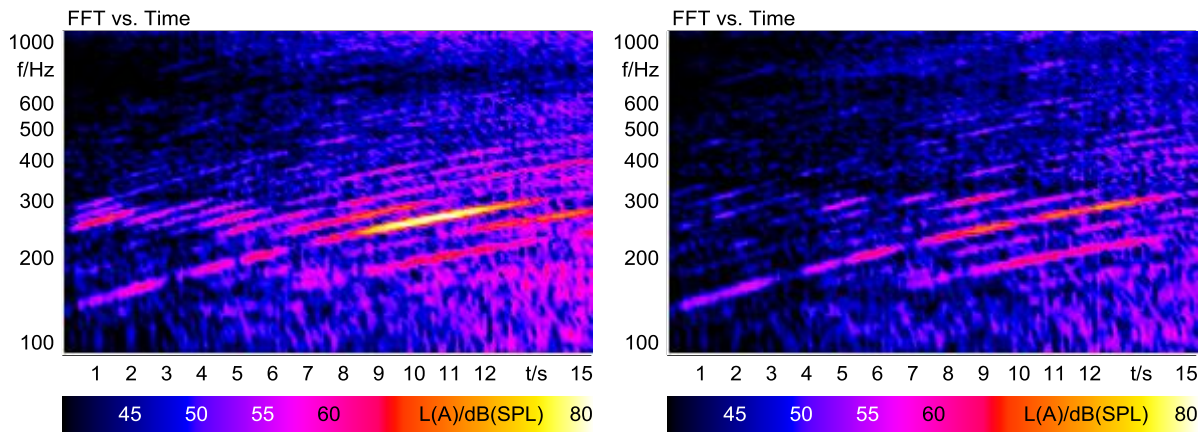


Figure 7: Time dependent FFT analysis of vehicle interior noise, left diagram: left channel, right diagram: right channel

In this example, ArtemiS SUITE and a *labP2* module are to be used for playing the BHM recording. For the playback, ArtemiS SUITE⁸ determines the maximum sound pressure amplitude in the recording and displays it in the info tile of the player. In our example, the value is 7.641 Pa. That is equivalent to a sound pressure level of 108.6 dB(SPL)⁹.

Depending on the selected **Hearing Protection Level**, **Comfort Level** and **Playback mode**, there are the following possibilities:

- **Playback mode: Automatic, Hearing Protection Level: 120 (114+6) dB(SPL)**
(i.e., selected maximum level > 108,6 dB(SPL):

Left	Right
7,641 Pa	7,332 Pa
- no level adaptation -	- no level adaptation -
44,1 kHz	44,1 kHz
ID	ID

ArtemiS SUITE sets the *labP2* to **114** and **ID** equalization. No level adaptation is necessary, the playback takes place with the correct signal level and equalization. This is indicated by the green frame around the artificial head icon in the Player.

- **Playback mode: Automatic, Hearing Protection Level: 110 (104+6) dB(SPL)**
(i.e., selected maximum level < 108.6 dB(SPL):

Left	Right
7,641 Pa	7,332 Pa
-4,6 dB	-4,6 dB
44,1 kHz	44,1 kHz
ID	ID

ArtemiS SUITE sets the *labP2* to **104** and **ID** equalization. Since the maximum sound pressure level in the recording exceeds the configured maximum level, the signal level of the entire file is reduced for the playback. The reduction factor is: $\Delta L = 104 \text{ dB(SPL)} - 108.6 \text{ dB(SPL)} = -4.6 \text{ dB(SPL)}$

Since the playback of the entire file is subject to the level reduction, the relative level differences within the recording are retained. The playback takes place with the correct equalization, but not with the original signal level. This is indicated by the artificial head icon being grayed out and a corresponding warning in the status bar.

- **Playback mode: Normalized, Hearing Protection Level: 120 (114+6) dB(SPL), Comfort Level: 110 dB(SPL)** (i.e., selected normalization level > 108,6 dB(SPL):

Left	Right
7,641 Pa	7,332 Pa
+1,4 dB	+1,4 dB
44,1 kHz	44,1 kHz
ID	ID

ArtemiS SUITE sets the *labP2* to **114** and **ID** equalization. The playback volume is increased so that the highest level occurring in the recording matches the selected normalization level of 110 dB(SPL). This means that the level is increased by $\Delta L = 110 \text{ dB(SPL)} - 108.6 \text{ dB(SPL)} = 1.4 \text{ dB(SPL)}$.

The playback takes place with the correct equalization, but not with the original signal level. This is indicated by the artificial head icon being grayed out and a corresponding warning in the status bar.

⁸ The descriptions in this Application Note refer to version 9.0 of ArtemiS SUITE. The general proceeding also applies to other versions as of ArtemiS SUITE 6.0. However, the scope of functions and the user interface may differ.

⁹ The sound pressure level is calculated with the following formula: $L = 20 \cdot \log\left(\frac{\tilde{p}}{2 \cdot 10^{-5} \text{ Pa}}\right)$, where \tilde{p} is the root mean square of the sound pressure amplitude \hat{p} , which can be calculated with the formula $\tilde{p} = \frac{\hat{p}}{\sqrt{2}}$ in case of sine-shaped signals.

- **Playback mode: Normalized, Hearing Protection Level: 110 (104+6) dB(SPL), Comfort Level: 100 dB(SPL)** (i.e., selected normalization level < 108,2 dB(SPL):

Left	Right
7,641 Pa	7,332 Pa
-8,6 dB	-8,6 dB
44,1 kHz	44,1 kHz
ID	ID

ArtemiS SUITE sets the *labP2* to **104** and **ID** equalization. The playback volume is reduced so that the highest level occurring in the recording matches the selected normalization level. This means that the level is reduced by $\Delta L = 100 \text{ dB(SPL)} - 108.6 \text{ dB(SPL)} = -8.6 \text{ dB(SPL)}$. The playback takes place with the correct equalization, but not with the original signal level. This is indicated by the artificial head icon being grayed out and a corresponding warning in the status bar.

Do you have any questions or remarks? Please write to:
imke.hauswirth@head-acoustics.de.

We look forward to receiving your response!