Using the BHM binaural head microphone

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 Binaural Head Microphone (BHM) and the artificial head HMS (HEAD Measurement System). 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 on the driver's seat in a vehicle

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 microphones. 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.[1]
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 one specific BHM only. The serial number of the matching BHM can be found on the BEQ II.1 housing.
   The HEAD Recorder software 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

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

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.

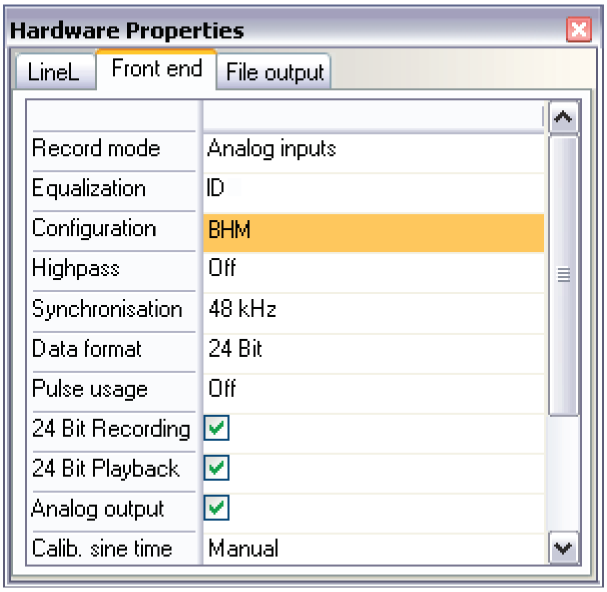


Figure 2: Hardware settings in the HEAD Recorder

2. Another possibility is to configure the equalization via the software used. To do this in the HEAD Sensor Explorer, the custom equalization filter for the BHM must be included in the "Filter File" field when defining the BHM sensor (see figure 3). 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 the HEAD Recorder software, 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.

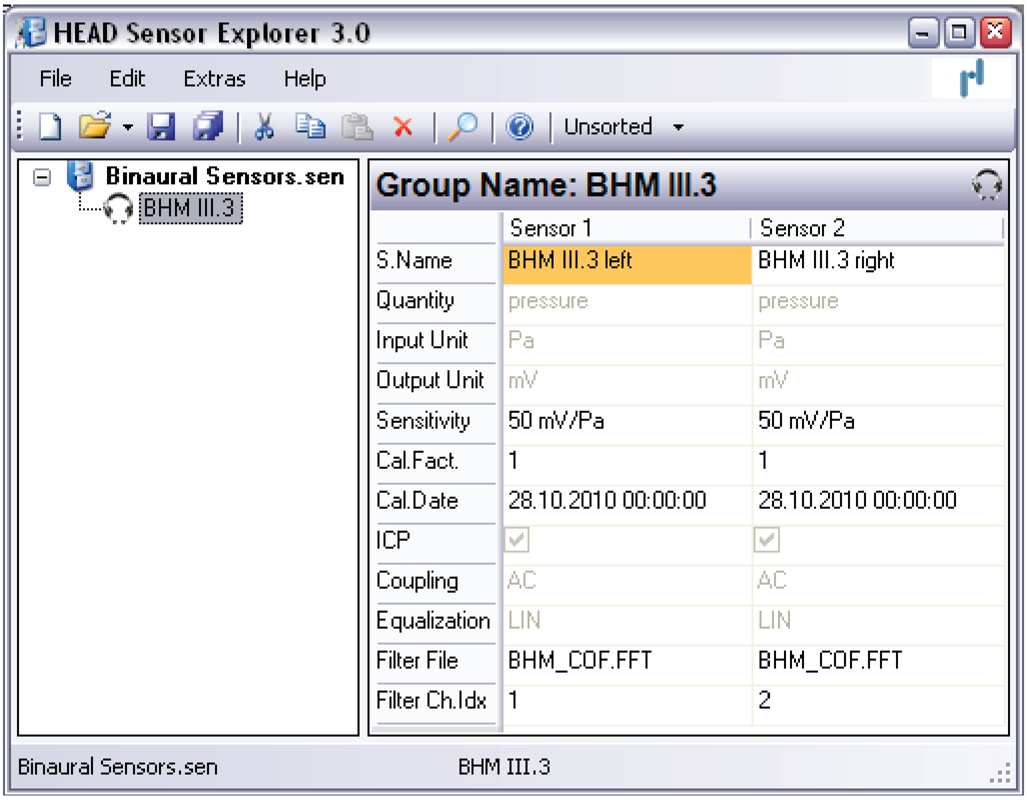


Figure 3: Definition of a BHM sensor in the Sensor Explorer

3. If the equalization has not been performed already during the recording, the signal can also be equalized later in the ArtemiS 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 position[2].

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 red curve shown in figure 4 (more than 15 dB at 10.5 kHz for the right channel).

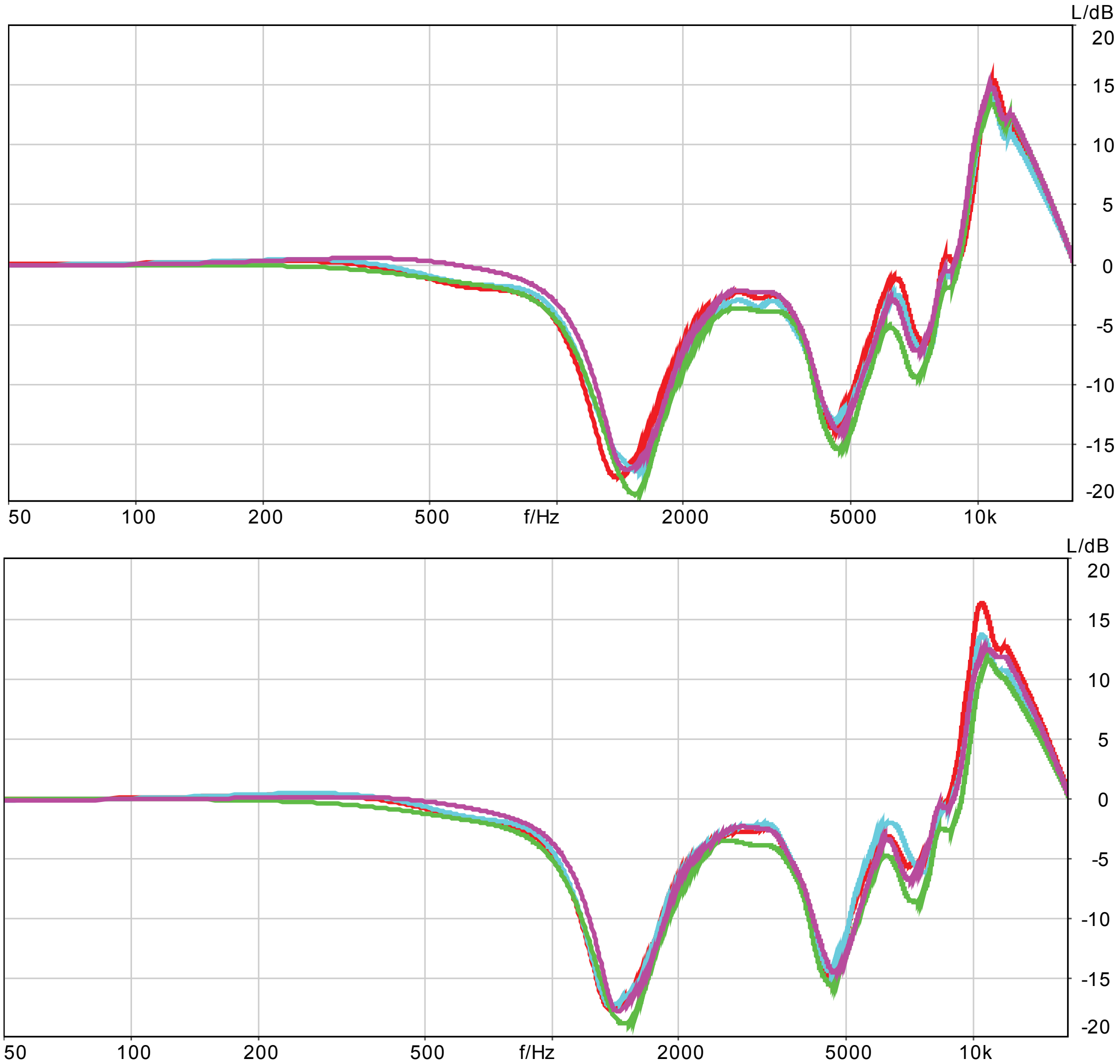


Figure 4: Individual equalization curves of four different BHM specimens

[2] 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.

Using the equalization curves

This strong amplification is necessary to ensure a correct equalization of the BHM recordings and the comparability of measurements and 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 is 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

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, online filtering in ArtemiS is the solution of choice. By using a suitable filter, the interfering signal components can be eliminated in a controlled way, allowing the listener to concentrate on the relevant signal.
ArtemiS offers the online filtering function in the Mark Analyzer, for example. After loading the signal into a Mark Analyzer with the desired analysis functions, the filter editor can be opened to select the desired filter. You can choose between a band-stop filter and a parametric band-pass filter[3]. 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.

[3] In order to use a parametric band-pass filter, you need the ATP 09 tool pack.

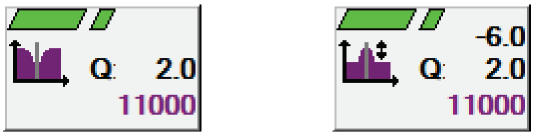


Figure 5: Band-stop filter Parametric band-pass filter

Of course, a signal can also be filtered permanently and saved as a new file. To do so, copy the filter used for online filtering to the Filter Pool. Afterwards, the time domain signal with the activated filter can be saved as a new file e.g. via the File Export or Wave Export functions. This provides a filtered audio signal that can be embedded in a PowerPoint® presentation, for example.

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

Sound impression in the original sound field

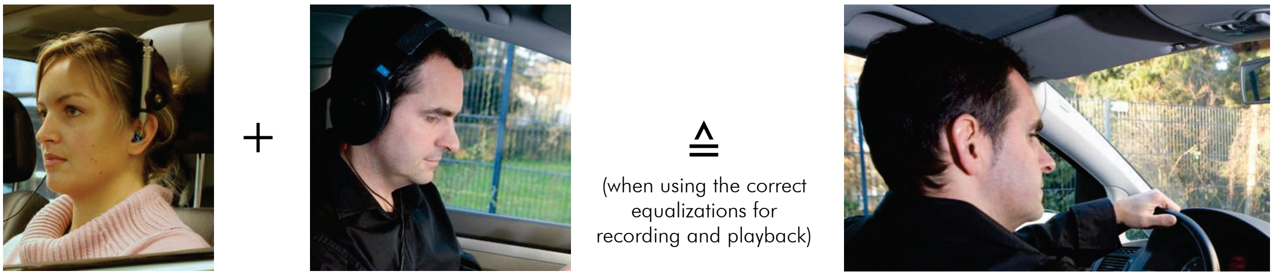


Figure 6: Recording and playback of a BHM recording

For the playback of BHM recordings, HEAD acoustics offers a programmable equalizer (PEQ). The PEQ is programmed with the required ID filter for a correct playback equalization of the BHM recording, resulting in a sound impression resembling the original sound field. The easiest way to play back a BHM recording is via the ArtemiS analysis software or the HEAD Data Portal, a PEQ and a headphone.

The level setting on the PEQ must match the measurement range of the recording. This can be achieved automatically or manually. If the two channels of the BHM measurement have different measurement ranges, a manually configured PEQ must be set to the higher of the two ranges. The channel with the lower measurement range is converted automatically by ArtemiS to ensure level-accurate playback of both channels. A detailed description of how to configure a PEQ IV or PEQ V can be found in the following section.

Application example: BHM recording in a vehicle[4]

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.

Analysis of the BHM recording

Figure 5 shows the FFT vs. time analysis 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.

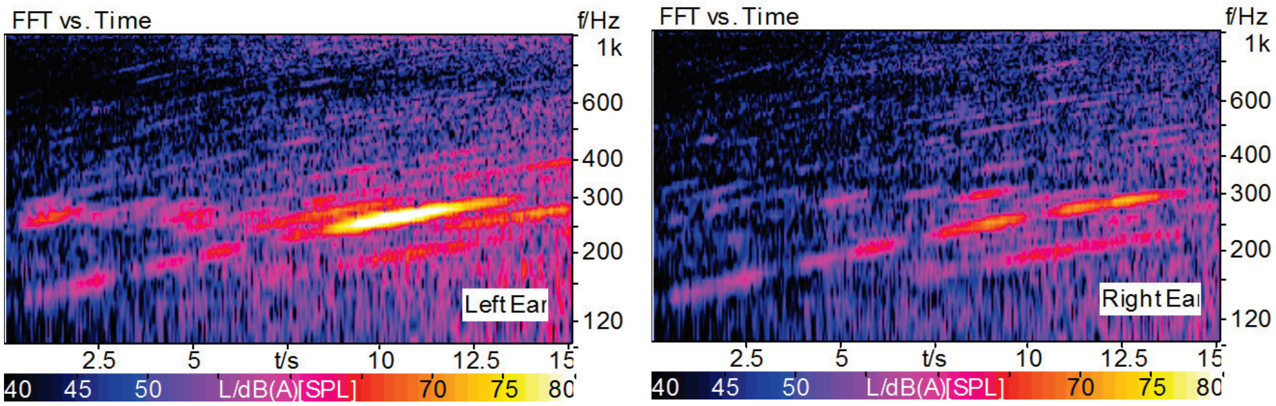


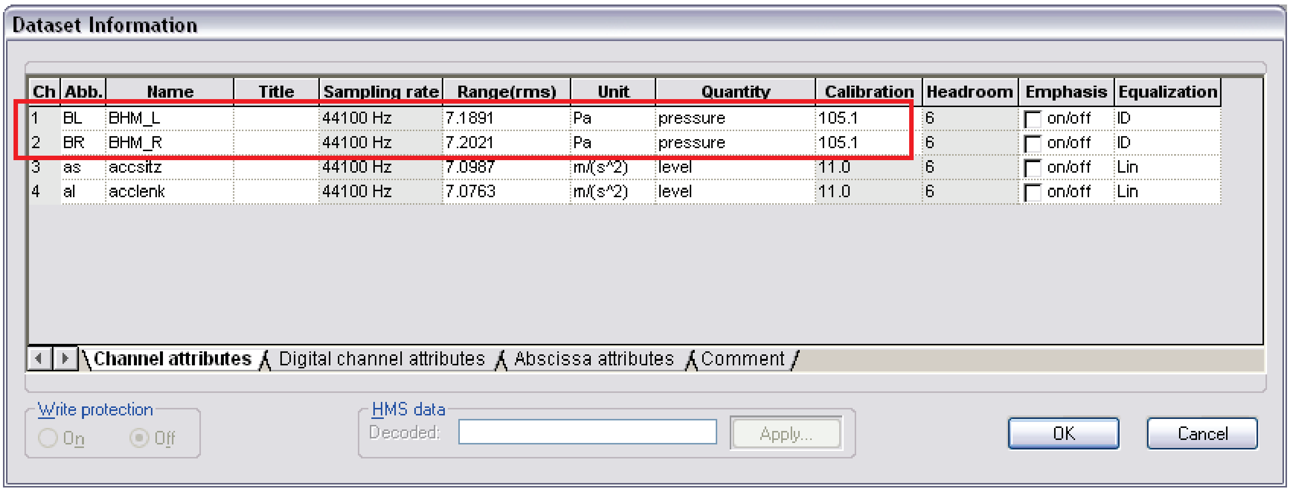
Figure 7: FFT vs. time analysis of vehicle interior noise

In order to verify whether the analysis results match the actually perceived acoustic impression, it is necessary to listen to the sound file.

Playback of the BHM recording

For the playback of the BHM file in this example, ArtemiS and a PEQ unit are used. The measurement range of the recording is 105.1 dB for both channels. This value is contained in the dataset information and can be read with ArtemiS (see figure 6).[5]

[4] This section is an excerpt from the Application Note "Binaural measurement, Analysis and Playback", which is included here as a supplement and clarification of the information on the BHM.
[5] To open the dialog shown here, keep the [Shift] key pressed while clicking on the "Dataset Info" button in the Properties dialog of the time domain signal.



Ch	Abb.	Name	Title	Sampling rate	Range(rms)	Unit	Quantity	Calibration	Headroom	Emphasis	Equalization
1	BL	BHM_L		44100 Hz	7.1891	Pa	pressure	105.1	6	on/off	ID
2	BR	BHM_R		44100 Hz	7.2021	Pa	pressure	105.1	6	on/off	ID
3	as	accsitz		44100 Hz	7.0987	m/(s^2)	level	11.0	6	on/off	Lin
4	al	acclenk		44100 Hz	7.0763	m/(s^2)	level	11.0	6	on/off	Lin

Figure 8: Dataset information about the BHM recording

If the "Auto" setting in the "Calibration" section of the "Settings/Audio Properties" dialog is selected, several possibilities are available for the correct playback:

- Playback with HMS data via a PEQ IV: If the option "Generate HMS / RPM pulse" is enabled, the level and equalization information contained in the file is sent to the PEQ IV as HMS data for the playback. If the PEQ is set to "Auto", the correct setting is configured automatically in one of the 10 dB steps (84 dB, 94 dB etc.). In our example with a measurement range of 105.1 dB, the PEQ would be automatically set to the next higher range, i.e. 114 dB, and to ID equalization. To make sure that the playback is not too loud, the signal level is recalculated accordingly by ArtemiS.
- Playback without HMS data via a PEQ IV: For a playback without HMS data, i.e. with the "Generate HMS / RPM pulse" option disabled, the level need not be set to one of the 10 dB steps predefined in the HMS data standard. Instead, the PEQ is manually set to the next possible whole-numbered level value matching the measurement range, in this case 105 dB.
- Playback via PEQ V (via USB): The PEQ V does not require HMS data for the automatic configuration. If the PEQ V is set to "Auto", the playback takes place either in the known 10 dB steps (if the option "Generate HMS / RPM pulse" is active) or with the next possible whole-numbered level value matching the measurement range (if the option "Generate HMS / RPM pulse" is disabled). In both cases, the level range of the file is recalculated suitably by ArtemiS, so that the file is played back with the correct level.

Do you have any questions for the author? Contact us at imke.hauswirth@head-acoustics.de.
We look forward to your feedback!