Introduction

Locating sound sources

By means of their binaural hearing system, human beings are capable of determining the location a sound is originating from. This locating of sound sources is possible in the horizontal as well as in the median plane. In the horizontal plane, locating is based on the evaluation of interaural differences. As soon as a sound source is not located directly in front of the head, the time the signal needs to reach the listener’s ears is slightly different for each ear, causing an interaural delay. Even though these differences are very small (between 20 μs and 1 ms) and are not perceived consciously, the brain can derive directional information from these differences. Whenever a sound arrives earlier at one ear, this means for our brain that the sound is coming from that side. This delay reaches its maximum when the sound source is located to the left or to the right of the listener. The interaural delay is considered the most important parameter for locating sound sources. In addition, shadowing effects cause interaural sound level differences. A sound source located next to the right ear generates a higher sound level in this ear than in the left one, because the head influences the propagation of the sound waves. These level differences, are also interpreted by the brain and used for locating the source. For spatial orientation, both effects are always used. However, experiments with artificial signals have shown that each of the two effects alone allows the location of a sound source to be determined as well.

Locating sound sources in the median plane is based on a different phenomenon. When a sound source is moving along the median plane, there are no interaural differences. Nevertheless, the brain is capable of locating a sound source in this plane as well. This capability is based on direction-dependent filtering of the sound caused by the anatomic shape of the auricles, the head, the shoulders and the upper body.
Depending on the direction, the frequency spectrum of the signal reaching the ear is distorted in a specific way. These spectral differences can be interpreted as a direction by the brain, which attributes certain distortions to specific directions. Compared to the locating of sound sources in the horizontal plane, the locating of results in the median plane has a less precise resolution.

The binaural hearing also allows separation of sound sources, suppression of interfering noise and selection of sources. This can be illustrated, for example, with the cocktail party effect: During a party with many guests having conversations in small groups and with a relatively high overall noise level, it is still possible to concentrate on one speaker and “blanking out” the other ones. This capability is based on the spatial separation of the sound sources, which in turn is only possible because the hearing system is able to locate the individual sound sources.

**Direction-dependent and direction-independent changes of the sound field**

As described above, in order to locate a sound source, the brain not only needs the interaural differences, but also the direction-dependent changes of the sound signal caused by the physical presence of the person in the sound field. These direction-dependent changes are mainly caused by the auricles, the head and the shoulders.

In addition to these direction-dependent changes, the auricle cavity (cavum conchae) and the ear canal cause changes that are independent of the direction the sound is coming from. Unlike the direction-dependent changes, which are based on diffraction and reflection, the direction-independent changes are caused by resonance. Figure 2 shows these different changes of the sound field and their respective causes.

![Figure 2: Causes of direction-dependent and direction-independent signal modifications](image-url)
Recordings with an Artificial Head

All the above-mentioned effects have in common the fact that they evaluate the interaural differences or sound field modifications caused by the physiology of hearing. Our hearing system depends on this information for determining a reasonable spatial location of the sound source. Only a sound recording preserving this information can create a real spatial impression during playback. Recordings made with a single microphone do not contain this information and therefore cannot be evaluated accordingly by the brain. Due to the lack of spatial separation of the sound sources, the brain cannot isolate the individual sources, making it harder to concentrate on one of them.

Recordings made with stereo microphones contain level differences and delays caused by the spatial separation of the two microphones. However, these recordings still lack the distortions of the sound field caused by the head, which are important for locating sound sources in the median plane. This means that a stereo recording does not allow sound sources to be located accurately, but only a rough determination of their direction in the horizontal plane.

A complete spatial reproduction of a sound field is only possible with an artificial head recording. Such a recording contains two channels (left and right ear), which include not only the required interaural differences, but also the required distortions, because the artificial head distorts the sound field in a similar way as a real person. An artificial head recording allows listeners to perceive a sound as if they were present in the original sound field.

The objective of head-related stereophony is the distortion-free measurement, transmission and reproduction of sound events at the human eardrums. The principle of using an artificial head for the measurement is basically simple, yet a lot of problems occurred with early artificial head measurement systems (e.g. high inherent noise, low dynamic range, lack of calibratability). In 1982, HEAD acoustics GmbH presented HMS I (HEAD Measurement System), the first calibratable artificial head measurement system with an authentic reproduction of the head and the auricles and transfer characteristics comparable to those of the human hearing system. Since 1989, the artificial head measurement system HMS II is available, which has a simplified geometry, which can be described mathematically, and a representative directional characteristic. The simplified geometry accounts for the relative positions of all acoustically relevant parts of the body, and thanks to its mathematic describability, it allows, for example, the calculation of the ID equalization.

The digital artificial heads HMS III and HMS IV (figure 3) are the result of the further development of the artificial head measurement technology by HEAD acoustics GmbH. Thanks to its improved technology, e.g. 24-bit technology, these artificial head measurement systems have a very low inherent noise and a dynamic range comparable to that of human hearing.

Using these new artificial head measurement systems, sound events can be recorded so that they contain all required information for locating the sound source.

Figure 3: HMS IV
Equalization of an Artificial Head Recording

Artificial head recordings should be played back on a playback system adapted to the artificial head, which should ensure that the listeners’ impression is exactly the same as if they had perceived the original sound event directly (see figure 4). To fulfill this requirement, the signals \( (p_r(t), p_l(t)) \) present at the ears of a listener in the sound field must equal the signals \( (p'_r(t), p'_l(t)) \) at the ears of a listener hearing the signals recorded by the artificial head in the same sound field.

![Figure 4: Demands on the playback of an artificial head recording](image)

For such a playback, however, it is required to filter the signals recorded with the artificial head prior to the playback. This filtering is also called equalization. The equalization is required because the membranes of the headphone cannot be placed in the same position as the microphone membranes in the artificial head. Figure 5 shows this in a simplified scheme. The sound that has already crossed one (artificial) auricular cavity before being recorded by the microphone in the artificial head is sent through another (human) auricular cavity when played back with headphones. Furthermore, the connection of the headphones to the ear and the different terminations (eardrum ↔ microphone), cause the sound field in the ear to be modified. Using equalization, these effects can be compensated for, so the signal reaching the eardrum when listening to an artificial head recording is the same as if the listener was present in the original sound field. In addition, the equalization allows possible irregularities in the transfer characteristics of the headphone to be compensated for.

![Figure 5: Equalization of an artificial head recording for authentic playback](image)
Equalization Interface

In order to analyze artificial head signals in a way that is compatible to conventional measurement technology (standard microphone recordings), a suitable interface must be provided. HEAD acoustics products split the equalization $H_{\text{total}}$ required for the aurally accurate reproduction of the sound signals into two partial equalizations ($H_{\text{record}}$ and $H_{\text{playback}}$), so the required interface is available between these two steps. At this interface, the artificial head signal has only been filtered by the recording equalization $H_{\text{record}}$ in order to make it comparable to conventional microphone recordings. This signal can then be used for the signal analysis. The splitting scheme is shown in figure 6.

![Diagram](image)

Figure 6: Splitting of the equalization for analysis purposes

To make sure that the signal at the interface point is equivalent to a conventional microphone recording, different recording equalizations $H_{\text{record}}$ are available for different sound fields. The artificial head measurement system from HEAD acoustics offers the following three recording equalizations: FF (free field), DF (diffuse field) and ID (independent of direction). To ensure that each recording equalization $H_{\text{record}}$ combined with the playback equalization $H_{\text{playback}}$ results in the correct total equalization $H_{\text{total}}$, there are different types of playback equalizations corresponding to the different recording equalizations. For an aurally accurate playback, the playback equalization $H_{\text{playback}}$ must compensate for the $H_{\text{record}}$ equalization and supplement it so that the total equalization $H_{\text{total}}$ is achieved. Only in that way, is it guaranteed that a person listening to the artificial head recording hears the very same signals at the eardrum as if this person was present in the original sound field.

The free field as well as the diffuse field are sound fields with exactly specified conditions. However, these conditions are hardly ever met in real-life situations. Therefore, HEAD acoustics developed the patented ID equalization. The ID equalization removes only the direction-independent components of the transfer function, caused by resonance, from the artificial head signal. The FF and DF equalizations also remove the direction-dependent distortions from the signal. The FF and DF equalizations were determined by extensive measurements, whereas the ID equalization is based on mathematical calculations.

Figure 7 (next page) schematically shows the basic procedure for determining the FF equalization. To determine the free field equalization for an artificial head, the head is placed in a free field and exposed to white noise coming from the front direction. Then the same measurement is repeated with a measurement microphone replacing the artificial head. The two resulting spectra of the artificial head and the microphone recording are subtracted from each other. The result is an FF equalization filter, which allows an artificial head recording under the given sound field conditions to be filtered so that it is equivalent to a microphone recording. Of course, the equalization only works correctly if the prescribed sound field conditions are fulfilled. For other sound field conditions or directions of sound incidence, other equalizations must be used.
So, using the FF equalization, an artificial head recording made in a free field with sound coming from the front can be equalized so that artificial head signal can be compared to a corresponding conventional microphone recording. Accordingly, the DF equalization can be used to equalize an artificial head recording made in a diffuse field with sound coming from all sides. In sound fields matching neither a diffuse field nor a free field, the ID equalization should be used. Using the wrong equalization, i.e. one that does not match the original sound field conditions and directions of sound incidence, will degrade the stored signal. An incorrectly equalized artificial head signal cannot be compared to a microphone signal and leads to misinterpretations during analysis. Figure 8 shows a comparison of the frequency curves of the three equalizations.

Figure 8: Frequency curves of the different equalization functions

An artificial head recording made with the correct recording equalization is largely comparable to a conventional microphone recording and can be examined with signal analysis software such as ArtemiS SUITE.
Binaural Recordings with Other Recording Devices

It is not always possible to use an artificial head for making a binaural recording. For example, the interior noise of a real vehicle as perceived in the driver’s position cannot be recorded with an artificial head during a test drive. The driver’s position must be taken by a real person operating the vehicle. For such situations, the Binaural Head Microphone (BHM) was developed (figure 9).

![Binaural Head Microphone BHM](image)

Such a recording device consists of two probe microphones worn by the driver in a similar way as headphones, with the microphone probes protruding into his ears. Via the probes, the two microphones record the sound pressure level at the driver’s ear canal entrances. Instead of the artificial head, the driver’s body causes all the necessary distortions of the sound field. That way, a binaural sound recording is achieved that is comparable to an artificial head recording.

A binaural head microphone recording must be equalized in the same way as an artificial head recording to make sure that people listening to the recording get the same impression as if they were present in the original sound field. For the binaural head microphone recording, this equalization is again split into a recording part and a playback part in order to provide an interface to conventional measurement technology. However, only the direction-independent changes of the sound field are equalized. For other sound field conditions, such as FF and DF, an artificial head should be used. The ID equalization for the binaural head microphone is based on measurements made with several different test subjects wearing the microphone and was designed so that an ID-equalized binaural head microphone recording is comparable to an ID-equalized artificial head recording. Minor differences are possible due to the anatomical differences of people wearing the binaural head microphone.

As with the artificial head, the equalization for the binaural head microphone is split, so that an interface to conventional measurement technology is available. Unlike the artificial head, where the equalization is achieved directly by the built-in electronics, a recording with a BHM binaural head microphone from HEAD acoustics requires the equalization to be performed either with a binaural equalizer (BEQ) or with the recording software. It is important to perform the equalization only once, otherwise the frequency spectrum of the recording would be misrepresented. The equalized binaural recordings can then be analyzed just like an artificial head recording.

Another binaural recording device from HEAD acoustics is the Binaural Headset BHS II. This headset is a binaural recording and playback unit that can be connected e.g. to SQuadriga II (figure 10, next page) or SQobold.
BHS II recordings can be ID-equalized just like binaural head microphone recordings and are then comparable to an ID-equalized artificial head recording. However, the differences between the BHS II recording and the artificial head recording can be larger than between a BHM recording and an artificial head recording. This is due to the different design of the devices and the different way they are worn. If the BHS II is connected to SQquadriga II or SQobold via the Headset input, the BHS II recordings are equalized automatically, so the BHS recording is immediately available with the correct equalization and ready to be analyzed just like an artificial head recording.¹

If the BHS II is connected to two Line/ICP channels via an adapter, the user has to ensure the application of the correct equalization filters. For stand-alone recordings with SQquadriga II or SQobold you can transfer the filters to the front end and activate them for the appropriate channels. If you record in front-end mode using a recording software, you can perform the equalization by means of the recording software. Then the equalization filters must be assigned during the sensor definition or in the channel list. In this case, too, it is important to apply the equalization only once.

Analysis of an Artificial Head Recording

The obvious difference of a correctly equalized artificial head recording compared to a microphone recording is that it is a two-channel measurement. To simplify the analysis of the artificial head recording, users often average the two artificial head channels; however, this is not advisable in most cases. Real-life artificial head recordings are almost always dichotic signals, i.e. the signals from the two ears are different. When the human brain perceives such different signals, they are not simply averaged. Examinations of the annoyance caused by dichotic noise signals have shown that the signals are considered to be more annoying as the interaural level differences become larger. Regarding other signal aspects, the arithmetic mean value does not necessarily represent the entire sound impression, because the calculation of the mean value can lead to the effect that a negative value on one channel is compensated by a positive value on the other channel (see also the application example in the appendix of this Application Note). This is not how the human brain processes and interprets sounds in most cases.

Therefore, in the analysis of artificial head signals, both channels should be examined initially. If a comparison shows that there are only minor differences between the channels, it is sufficient to use only one of the two channels for the subsequent examinations. If there are major differences, it can be advisable to use the channel delivering the most prominent analysis result regarding unwanted noise parameters (for example, a higher sharpness value than on the other channel).

In addition to the signal analysis, it is always important to listen to the artificial head signals. A comparison between the live impression and the results of the signal analysis shows which analysis

¹ Note that the automatic equalization can only work correctly if the equalization filters are matched specifically to the connected BHS II specimen. To ensure this, you should only connect the BHS II specimen whose serial number is stated on the SQquadriga II or SQobold housing to the Headset input.
best represents the real-life impression. Furthermore, such a comparison shows which of the two channels should get special attention.

**Playback of Binaural Recordings with ArtemiS SUITE (version 6.0 or higher)**

For the playback of binaural recordings, it is important to adapt the playback level and the playback equalization to the settings used for the recording. As described above, the playback equalization $H_{\text{playback}}$ must be chosen so that together with the recording equalization $H_{\text{record}}$, it results in the total equalization $H_{\text{total}}$. The chosen equalization filter not only influences the playback level, but also the spectral distribution, i.e. the sound of the recording.

For the playback of binaural recordings, HEAD acoustics offers the programmable headphone equalizer labP2 (figure 11). Like its predecessor PEQ V, it is programmed with all required filters for playback equalization, so that binaural 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.

![Figure 11: Front and back of a labP2](image)

**Selecting the playback equalization**

If the playback is started in ArtemiS SUITE, the correct playback equalization is enabled automatically. For this purpose, ArtemiS SUITE reads the equalization information stored in the recording file and passes it to the playback device. If the file does not contain such information, ArtemiS SUITE automatically sets the playback device to a default equalization setting specified for this case. You can select the default equalization via *Tools -> Options -> Playback -> Playback Frontend -> Default Equalization*, see figure 12).

![Figure 12: Playback options for selecting the default equalization](image)
Selecting the playback level for artificial head recordings

Besides the correct equalization setting, the selection of the playback level is essential for a correct, calibrated playback. For artificial head recordings, the level range is predefined in 10-dB steps (84 dB, 94 dB etc. plus a reserve of 6 dB(SPL) called “headroom”). This level range corresponds to the setting that must be selected on the playback device for an accurate playback level.

With ArtemiS SUITE 6.0, the playback functionality was enhanced significantly, which is why the setting on the playback device may differ from that on the recording device. ArtemiS SUITE now offers you some protection and comfort functions. The settings for these functions can be configured on the playback settings page (Tools -> Options -> Playback -> Playback Mode, see figure 13).

Figure 13: Playback settings with protection and comfort functions

In the **Hearing Protection Level** field, you can specify a maximum sound pressure level in dB(SPL), which must not be exceeded during the playback. The first value in this field specifies the maximum level (in the figure: **100 dB(SPL)**). This value is composed of 94 dB(SPL) (level range set on the playback device) + 6 dB(SPL) (headroom).

In the next field, you can enter a **Comfort Level**. This value specifies the playback level you feel comfortable with. Valid values are between 50 dB(SPL) and the configured **Hearing Protection Level**.

The playback device is automatically set to the **Hearing Protection Level** by ArtemiS SUITE (example: A **Hearing Protection Level** setting of **100** (94+6) dB(SPL) causes a setting of **94** on the playback device).

Furthermore, two playback modes are available to choose from:

In **Automatic** mode, ArtemiS SUITE determines the highest signal level in the file to be played. As long as this level is within the level range you selected, the recording is played with the aurally accurate level. If the highest level exceeds the configured level range, the file is automatically played at a reduced. For this purpose, the ratio of the maximum occurring level to the configured level range is determined, and the playback level for the entire file is reduced by that factor. This means that the playback level is no longer aurally accurate, which is indicated to the user by the artificial head icon in the player being grayed out. Furthermore, the message **Reduced Playback Level** is displayed in the status bar.

In **Normalized** mode, all signals are played at the configured **Comfort Level**, independent of the originally recorded level. Recordings whose highest level is below the **Comfort Level** are amplified accordingly, whereas recordings whose maximum level exceeds the **Comfort Level** are attenuated. This allows you to compare recordings with different recording levels independent of their volume (e.g. sounds from comparable products recorded at different distances from the source). Playback in **Normalized** mode is not aurally accurate.

The **Automatic** mode can only be used for airborne sound signals. For all other signal types (e.g. structure-borne sound, acceleration, voltage), ArtemiS SUITE automatically uses the **Normalized** mode.

In artificial head recordings, level differences between the left and the right ear are always retained regardless of the playback mode. Any kind of adaptation always uses the channel containing the higher maximum level in this case. The other channel is modified by the same factor, so that its relative level differences to the louder channel are retained for the playback.

For playback via a playlist, you can also activate the additional function **Relative**. In this mode, ArtemiS SUITE determines the highest level within the entire playlist and a correction factor for adapting the playback volume so that this maximum level matches the configured **Comfort Level**. All files in the playlist are amplified or attenuated accordingly, so that the relative level differences between the
recordings in the playlist are retained. The absolute playback level is not aurally accurate in this playback mode.

In addition to the setting described above for influencing the playback level, the Player interface also provides a volume slider allowing you to adjust the playback level between -40 dB and +10 dB (see figure 14).

![Figure 14: Player interface with volume slider](image)

The slider handle is subdivided into three areas, allowing you to adjust both the left and the right channel (dark gray bar in the middle) or each channel individually (light gray areas on the outside). When you move the mouse pointer on the slider, you are also offered a control field where you can enter the desired level change numerically. As soon as you change the playback level with the slider, the circle around the Play/Pause button turns orange, and the warning **Auralization Modified** is displayed in the status bar.

**Selecting the playback level for BHM recordings**

BHM recordings should also be played back via ArtemiS SUITE analysis software, a labP2 or a PEQ V, and a headphone. Since only ID equalization is available for BHM recordings, the playback device is always set to ID mode.

The playback level is adjusted as described in the previous chapter, i.e. the two modes **Automatic** and **Normalized** are available for BHM recordings, too. So you can again choose between aurally accurate playback or – if the recording is too loud – reduce the playback volume to your **Comfort Level**.

If the two channels of the BHM measurements have different signal ranges, the channel with the lower range is automatically recalculated by ArtemiS SUITE, so that aurally accurate playback is possible for this channel, too.

**Playback of binaural recordings with BHS II and SQuadriga II or SQobold**

Since the BHS II headset is a combined recording and playback unit, not only the recording, but also the playback takes place directly via the BHS II, if it is connected to SQuadriga II or SQobold via the Headset input. This combination allows not only BHS II recordings, but also other binaural recordings to be played back, for example artificial head recordings. For an accurate playback of artificial head recordings not recorded with ID equalization, FF and DF equalization can be selected for playback as well.

In stand-alone mode, recordings stored on the memory card can be played back directly with SQuadriga II or SQobold. Once the function **Auto** in the **Headphone** menu is selected, the equalization and level settings for the playback are adjusted automatically by the front end. Furthermore, it is also possible to select the settings for the equalization and the level manually. If the automatic setting has been disabled, you must set the equalization mode and the signal range for the playback so that they match the settings during the recording in order to achieve aurally accurate playback.

If SQuadriga II or SQobold is used as a playback device for ArtemiS SUITE, the settings are made by ArtemiS SUITE, which automatically configures the correct equalization mode and signal range as described in the previous chapter.
Application Example

BHM recording in a vehicle cabin

Due to its mixture of sound-reflecting and sound-absorbing materials, the interior of a vehicle is neither a genuine free field nor a diffuse field. Therefore, the ID equalization is the method of choice for recordings in the passenger compartment of a car. 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 15 shows the FFT vs. time analysis of the vehicle cabin 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.

![FFT vs. Time Analysis](image)

Figure 15: FFT vs. time analysis of a vehicle interior noise

Figure 16 shows the result of a Specific Prominence Ratio analysis of the vehicle interior noise shown in figure 15. The Specific Prominence Ratio analysis is used for identifying and quantifying tonal components in a signal. For this purpose, the signal power is determined, for example, in a third-octave band, which is then related to the mean value of the powers in the neighboring third-octave bands. The higher the resulting ratio, the more tonality is contained in the sound. The Specific Prominence Ratio analysis is well suited for identifying resonances in signals like the one shown above.

![Specific Prominence Ratio Analysis](image)

Figure 16: Spec. Prominence Ratio analysis of the vehicle interior noise from fig. 15

This analysis, again shows that the resonance is much more prominent in the left ear than in the right one. An averaging of the two channels as shown in figure 17 reduces the significance of the analysis considerably. While the tonal component is still visible in the averaged analysis, its prominence is...
reduced significantly. In this case, it would therefore be advisable to use the left channel for further analysis of this kind.

![Figure 17: Averaged values of the Spec. Prominence Ratio analysis from fig. 14](image)

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
In this example, ArtemiS SUITE and a labP2 module are to be used for playing the BHM recording. For the playback, ArtemiS SUITE 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)².

For a playback with a labP2, 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):
  
<table>
<thead>
<tr>
<th>Left</th>
<th>Right</th>
</tr>
</thead>
<tbody>
<tr>
<td>7.641 Pa</td>
<td>7.641 Pa</td>
</tr>
<tr>
<td>no level adaptation</td>
<td>no level adaptation</td>
</tr>
<tr>
<td>44.1 kHz</td>
<td>44.1 kHz</td>
</tr>
</tbody>
</table>

  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.

- **Playback mode: Automatic, Hearing Protection Level: 110 (104+6) dB(SPL)**
  (i.e. selected maximum level < 108.6 dB(SPL):
  
<table>
<thead>
<tr>
<th>Left</th>
<th>Right</th>
</tr>
</thead>
<tbody>
<tr>
<td>7.641 Pa</td>
<td>7.382 Pa</td>
</tr>
<tr>
<td>-4.6 dB</td>
<td>-4.6 dB</td>
</tr>
<tr>
<td>44.1 kHz</td>
<td>44.1 kHz</td>
</tr>
</tbody>
</table>

  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 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 warning in the status bar (**Reduced Playback Level**).

- **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):
  
<table>
<thead>
<tr>
<th>Left</th>
<th>Right</th>
</tr>
</thead>
<tbody>
<tr>
<td>7.641 Pa</td>
<td>7.641 Pa</td>
</tr>
<tr>
<td>no level adaptation</td>
<td>no level adaptation</td>
</tr>
<tr>
<td>44.1 kHz</td>
<td>44.1 kHz</td>
</tr>
</tbody>
</table>

  The sound pressure level is calculated with the following formula:

  \[ L = 20 \cdot \log \left( \frac{\hat{p}}{p_{10 \text{Pa}}} \right) \]

  where \( \hat{p} \) is the root mean square of the sound pressure amplitude \( p \), which can be calculated with the formula \( \hat{p} = \frac{p}{\sqrt{2}} \) in case of sine-shaped signals.

  

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

- **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):
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.

Do you have any questions or comments? Please write to imke.hauswirth@head-acoustics.de. We look forward to receiving your feedback!