

# Binaural Measurement, Analysis and Playback

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## Introduction

### Locating sound sources

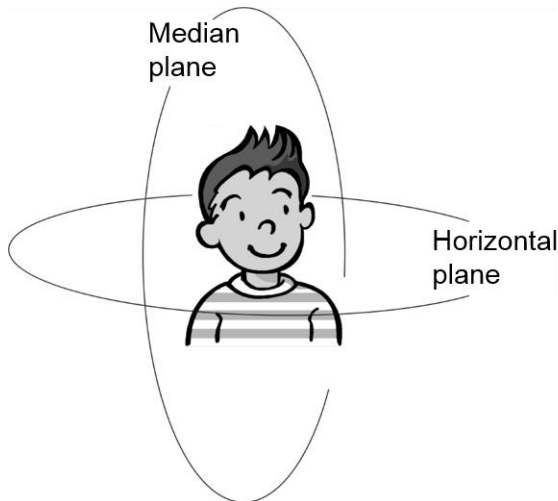


Figure 1: Median and horizontal plane

By means of their binaural hearing system, human beings are capable of determining the location of a sound. 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  $\mu$ s und 1 ms) and are not perceived consciously, the brain can interpret these differences and derive directional information. 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 differences in sound pressure level. A sound source located next to the right ear generates a higher sound pressure 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.

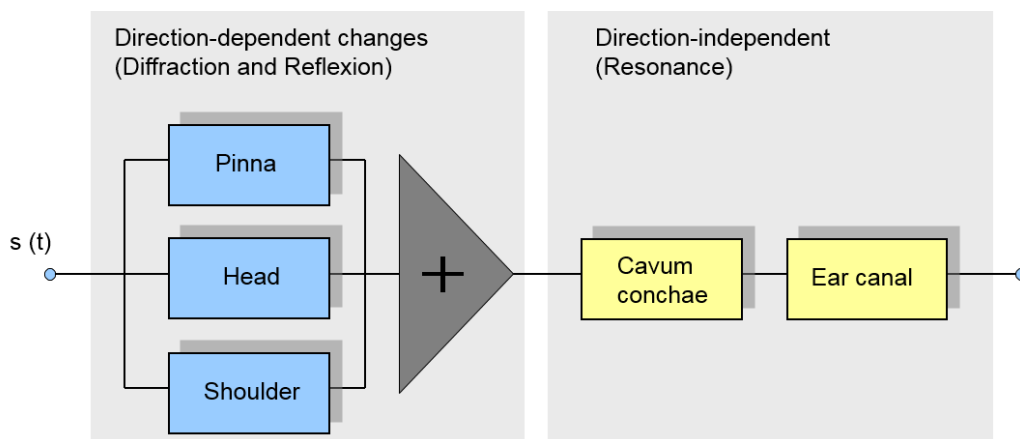
Locating sound sources in the median plane is based on a different phenomenon. When a sound source is moving along the median plane, there are no significant interaural differences. Nevertheless, the brain is capable of locating a sound source in this plane as well. This capability is based on direction-dependent 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 specific spectral differences can be interpreted as a direction by the brain, which attributes certain distortions to certain 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 to a certain extent. 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 spots of the 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 signal and their respective main causes.



**Figure 2:** Main causes of direction-dependent and direction-independent signal modifications

## Recordings with an artificial head measurement system

All the above-mentioned mechanisms have in common that they evaluate the interaural differences or sound signal 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 and reproduce the spatial position of the sound sources. 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 sound sources, making it harder to concentrate on one of them.

Recordings made with stereo microphones do contain level differences and delays caused by the spatial separation of the two microphones. However, these recordings lack the distortions of the sound field caused by the head, which are particularly 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 position between both loudspeakers.

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. In the ideal case 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 (independent of direction) 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 high dynamic range.

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:** Head measurement system 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.

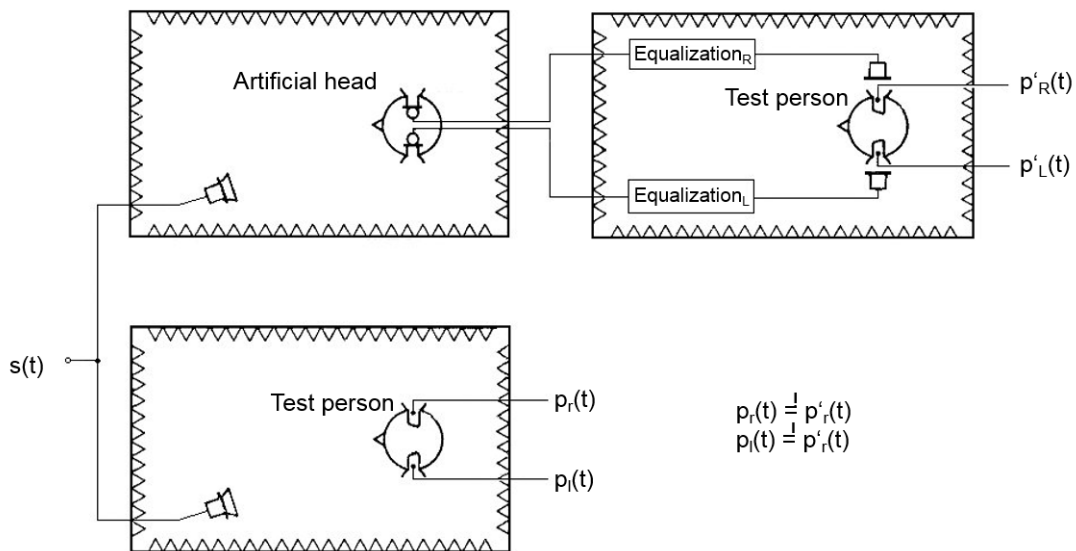


Figure 4: Demands on the playback of an artificial head recording

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 among other reasons because the membranes of the headphones 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 headphones to be compensated for.

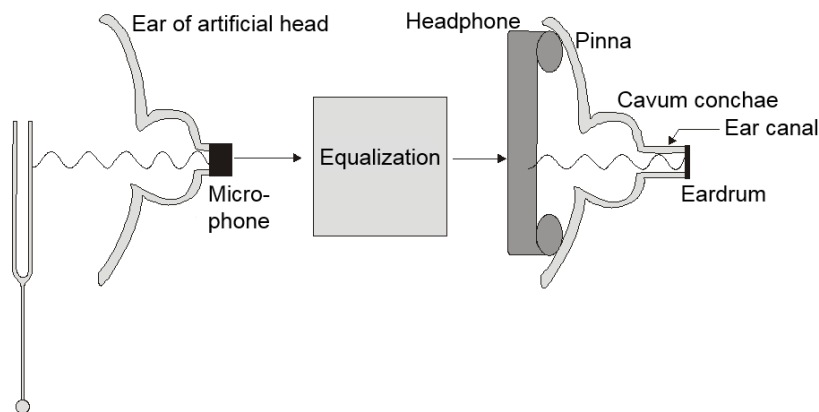
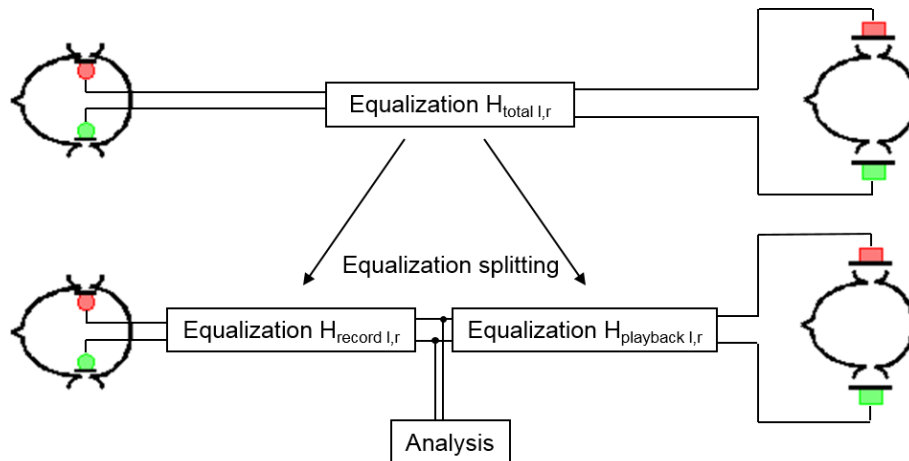


Figure 5: Equalization of an artificial head recording for authentic playback

## 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  $H_{total}$  required for the aurally accurate reproduction of the sound signals into two partial equalizations (recording equalization  $H_{record}$  and playback equalization  $H_{playback}$ ), 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  $H_{record}$  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.



**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  $H_{record}$  are available for different sound fields. The artificial head measurement systems from HEAD acoustics offer the following three recording equalizations: free field (FF), diffuse field (DF) and independent of direction (ID). To ensure an aurally accurate playback, the playback equalization  $H_{playback}$  must compensate for the recording equalization  $H_{record}$  and establish the total equalization  $H_{total}$ . Only in this way, it is guaranteed that a person listening to the artificial head recording hears the very same signals at the eardrums as if this person was present in the original sound field. In order to compensate for the different, sound-field-specific recording equalizations correctly during the playback, the playback devices of HEAD acoustics provide corresponding playback recordings.

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 at a distance of 3 meters. 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 (free field, sound exposure from the front direction at a distance of at least 3 meters). For other sound field conditions or directions of sound incidence, other equalizations must be used.

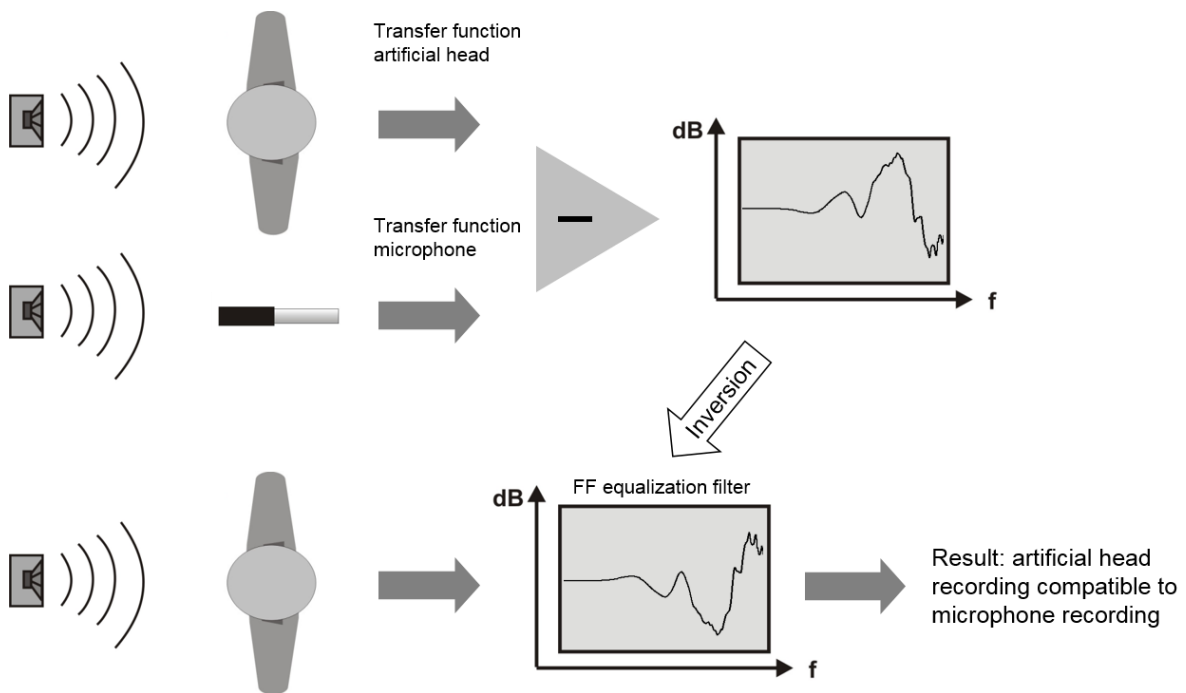


Figure 7: Measurement for determining the free field equalization (simplified)

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 acquired signal. An incorrectly equalized artificial head signal cannot be compared to a microphone signal and can lead to possible misinterpretations during analysis.

Figure 8 shows a comparison of the frequency curves of the three equalizations.

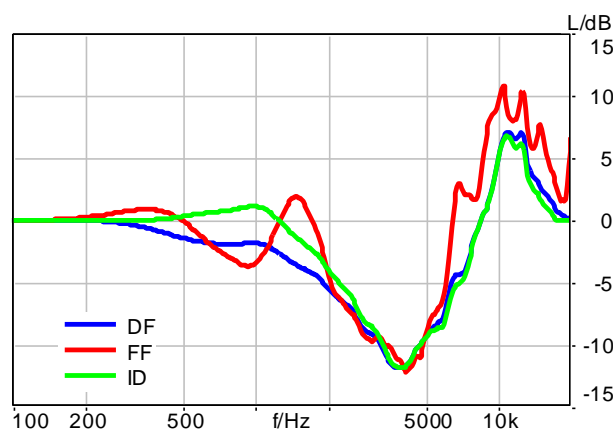


Figure 8: Frequency curves of the different equalization functions

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).



**Figure 9:** Binaural Head Microphone BHM

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 binaural head microphone recordings the ID equalization is available, i.e. only the direction-independent changes of the sound field are equalized. 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 in the recordings are possible due to the anatomical differences of people wearing the binaural head microphone.<sup>1</sup>

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 the frontend or with the recording software.<sup>2</sup> It is important to perform the equalization only once, otherwise the recording contains a measurement error that falsifies the frequency spectrum of the recording. 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.

<sup>1</sup> The application note „Comparability of recordings made with measurement systems from HEAD acoustics“ provides an overview of these differences.

<sup>2</sup> You can find detailed information in the application note „Using the BHM binaural head microphone “.



**Figure 10:** SQuadriga II with BHS II

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 SQuadriga 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.<sup>3</sup>

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 SQuadriga 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 difference of a correctly equalized artificial head recording compared to a microphone recording is (among others) that it is a two-channel measurement, whereas the microphone recording only contains one channel. 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 humans perceive and evaluate 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 humans process and interpret 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 perception-relatedly advisable to use the channel delivering the most critical analysis result regarding unwanted noise parameters (for example, a higher sharpness value than on the other channel). Various calculation rules, e.g. for the loudness, have been described in literature to merge the values for the left and the right channel to one value. However, due to the fact that these rules differ depending on the publication and highly depend on both the selection of the presented signals and the presentation form

<sup>3</sup> 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 SQuadriga II or SQobold housing to the BHM input.



(e.g. via headphones or via loudspeakers), no general recommendation can be given for a specific calculation rule.

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 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<sup>4</sup>

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 does not only influence 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 equalizer *labP2* (figure 11). 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 headphones specimen used are installed. Furthermore, this equalizer contains filters that allow possible deviations of the headphones transfer characteristics to be compensated for. The number above the headphones sockets of the *labP2* is the serial number of the headphones for which this output delivers the correct calibration and equalization.



Figure 11: Front and back of a labP2

### 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 -> Basic -> Playback -> Playback Frontend -> Default Equalization**, see figure 12).



Figure 12: Playback options for selecting the default equalization

<sup>4</sup> 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.

## 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 fixed 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: ArtemiS SUITE now offers you several protection and comfort functions. As a consequence the setting on the playback device may differ from that on the recording device. The settings for these functions can be configured on the playback settings page (**Tools** -> **Options** -> **Basic** -> **Playback** -> **Playback Level**, see figure 13).



Field	Value
Hearing Protection Level [dB(SPL)]	100 (94 + 6)
Comfort Level [dB(SPL)]	85
Playback Mode	Automatic

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 (94+6)**). 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)** 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 have 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 level. 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 accordingly. 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.

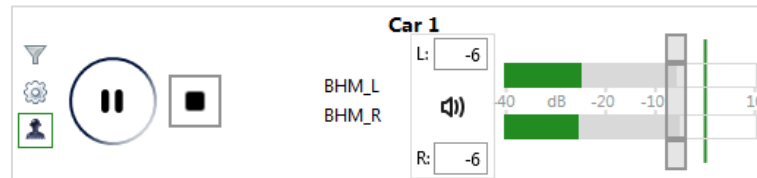
In **Normalized** mode, all signals are played at the configured **Comfort Level**, independent of the originally recorded level. Recordings whose highest levels are below the **Comfort Level** are amplified accordingly, whereas recordings whose maximum level exceeds the **Comfort Level** are attenuated. 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 binaural 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 **Comparable Levels**, when using the **Normalized** mode. Thus, 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 settings described above for influencing the playback level, the Player also provides a volume slider allowing you to adjust the playback level between -40 dB and +10 dB (see figure 14).



**Figure 14:** Player with volume slider

The slider handle is subdivided into three areas, allowing you to adjust both the left and the right channel (bar in the middle) or each channel individually (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.

### Selecting the playback level for BHM recordings

BHM recordings should also be played back via ArtemiS SUITE analysis software, *labP2*, and headphones. 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.

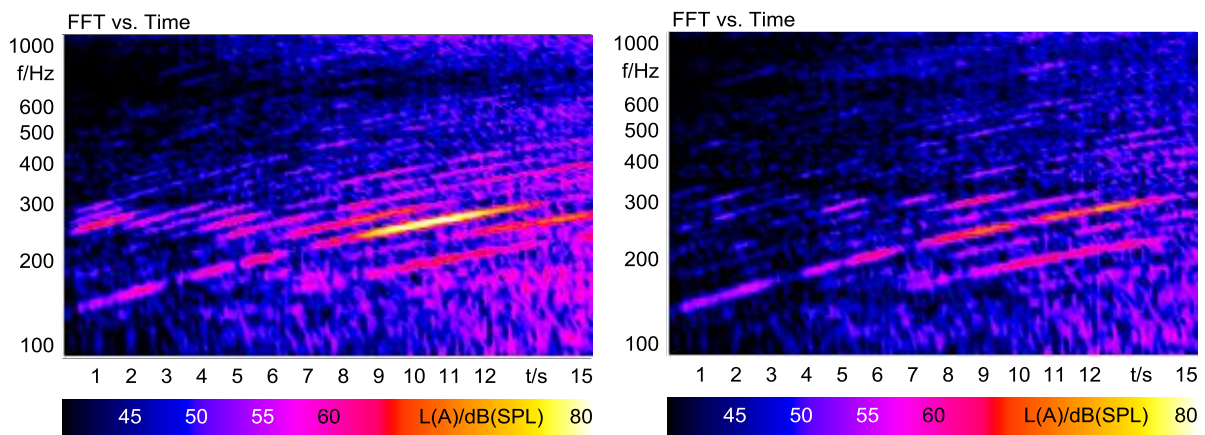


Figure 15: *FFT vs. Time* analysis of a vehicle interior noise

Figure 16 shows the result of a *Specific Prominence Ratio vs Time* analysis of the vehicle interior noise shown in figure 15. The *Specific Prominence Ratio vs Time* 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 pronounced is the respective tonal component. The *Specific Prominence Ratio vs Time* analysis is well suited for identifying resonances in signals like the one shown above.

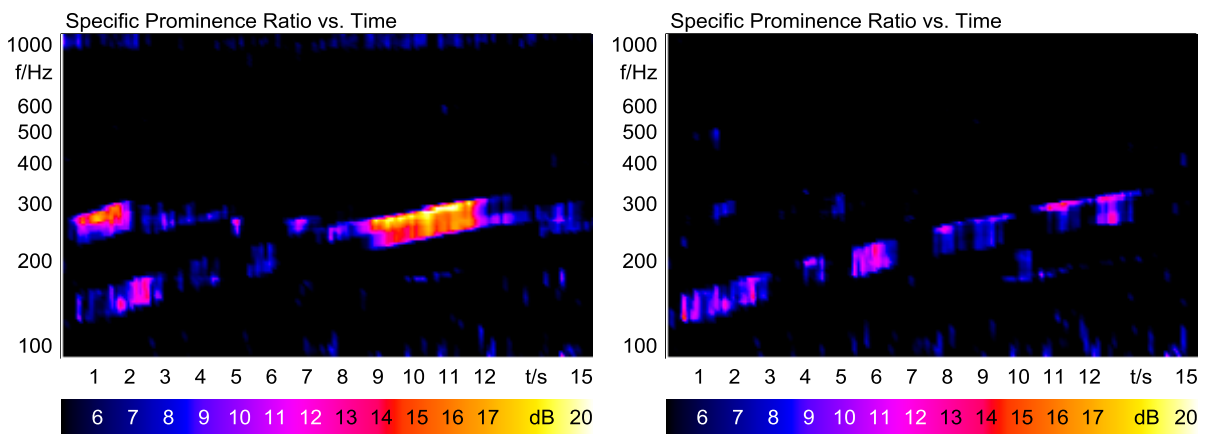
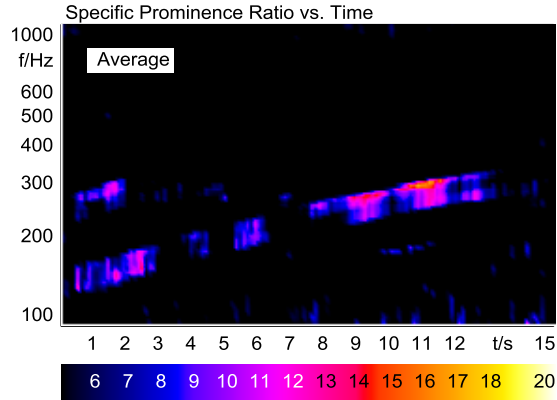


Figure 16: *Specific Prominence Ratio vs Time* analysis of the vehicle interior noise from fig. 15

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 examine particularly the left channel for further analysis of this kind.



**Figure 17:** Averaged values of the *Spec. Prominence Ratio vs Time* analysis from fig. 14

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)<sup>5</sup>.

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):

Left	Right
7,641 Pa	7,332 Pa
- 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.

<sup>5</sup> 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: 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.

- **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.

More information about how to use our binaural recording and playback devices can be found in the Application Note “Proper binaural recording and playback” which is provided in the Download Center on our website: [http://www.head-acoustics.de/eng/nvh\\_application\\_notes\\_use\\_of\\_systems.htm](http://www.head-acoustics.de/eng/nvh_application_notes_use_of_systems.htm)

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