Binaural Measurement, Analysis and Playback

Contents

Introduction

Locating sound sources

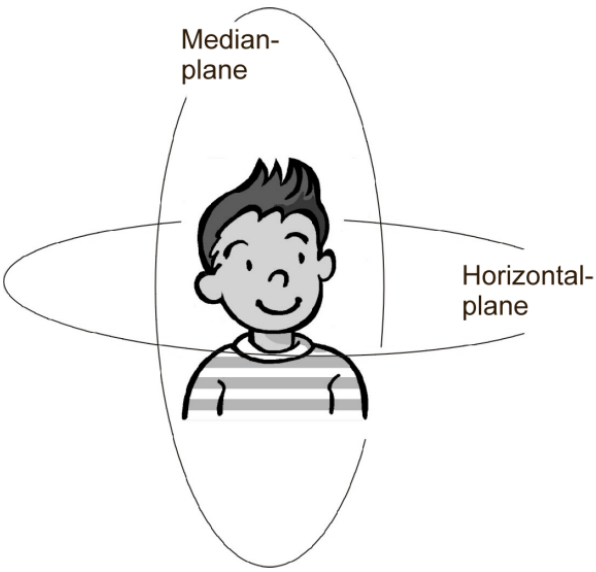


Figure 1: Median and horizontal plane

By means of their binaural hearing system, human beings are capable of determining the location 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 (considerably less than 0.0001s) 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 system also allows the separation of sound sources, the suppression of interfering noise and the 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.

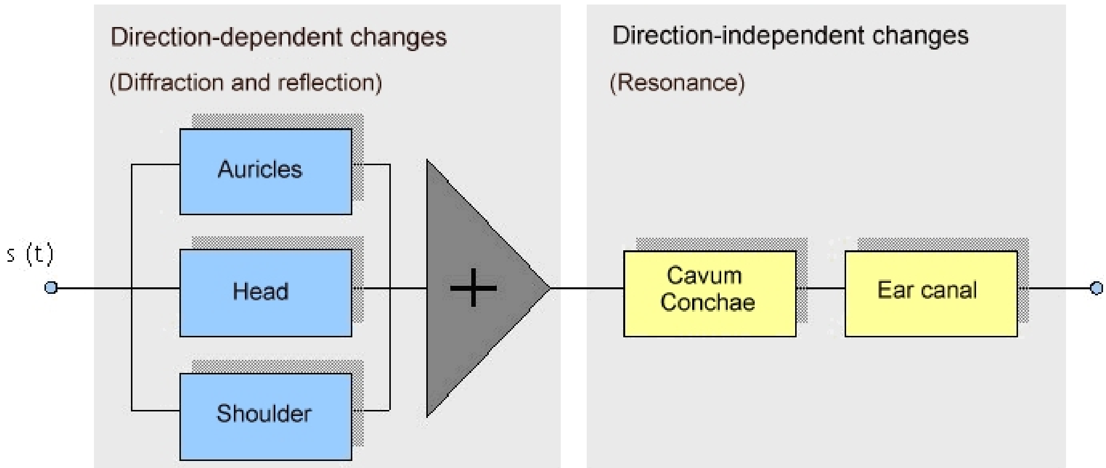


Figure 2: Causes of direction-dependent and direction-independent signal modifications

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 24-bit technology, this artificial head measurement system has 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.

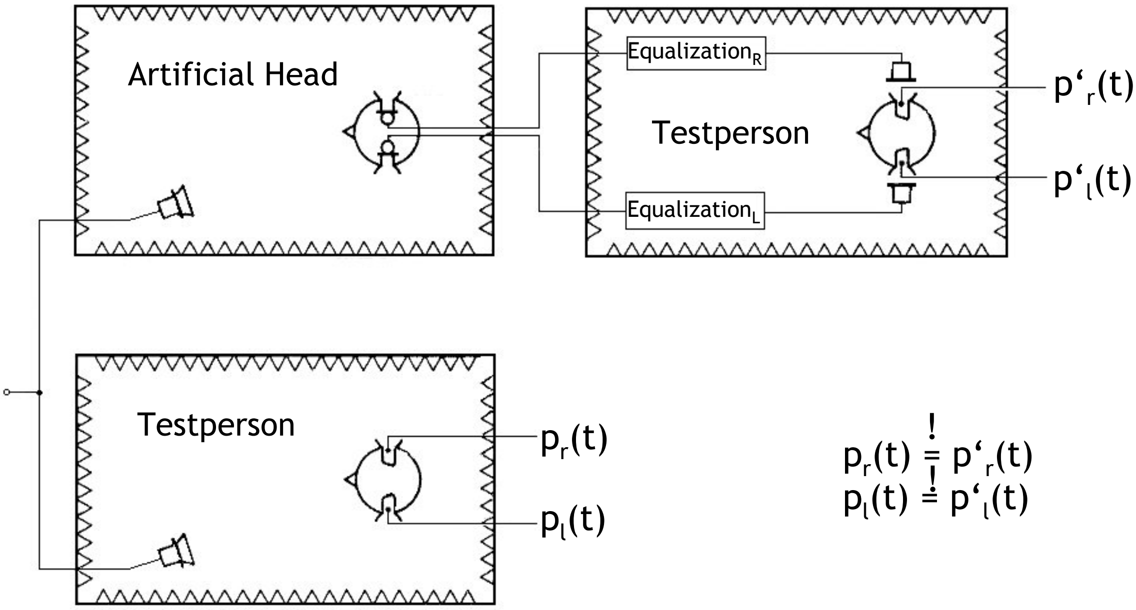


Figure 4: Demands on the playback of an artificial head recording

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.

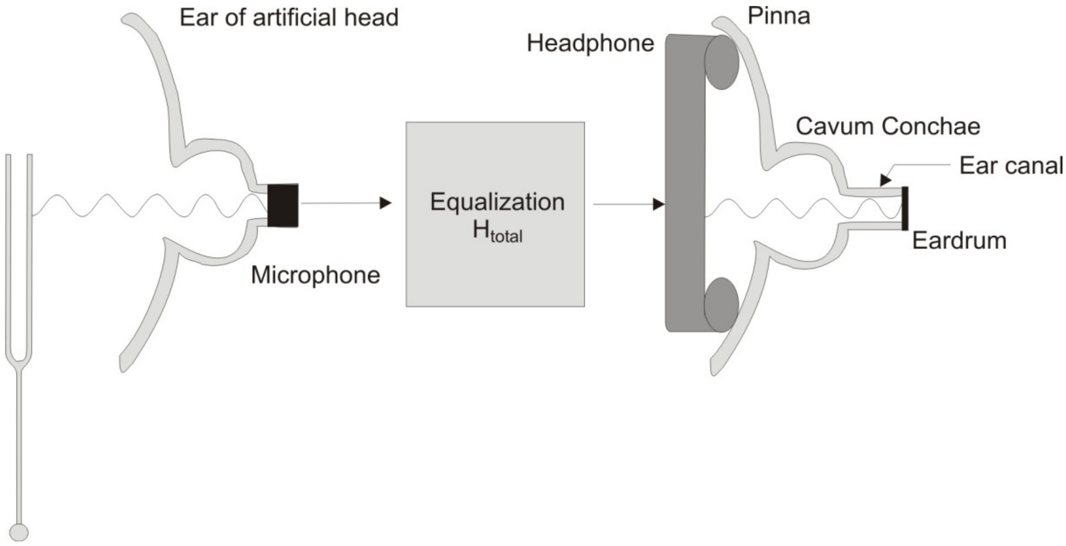


Figure 5: Equalization of an artificial head recording for authentic playback

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 (H_{record} and $H_{playback}$), 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.

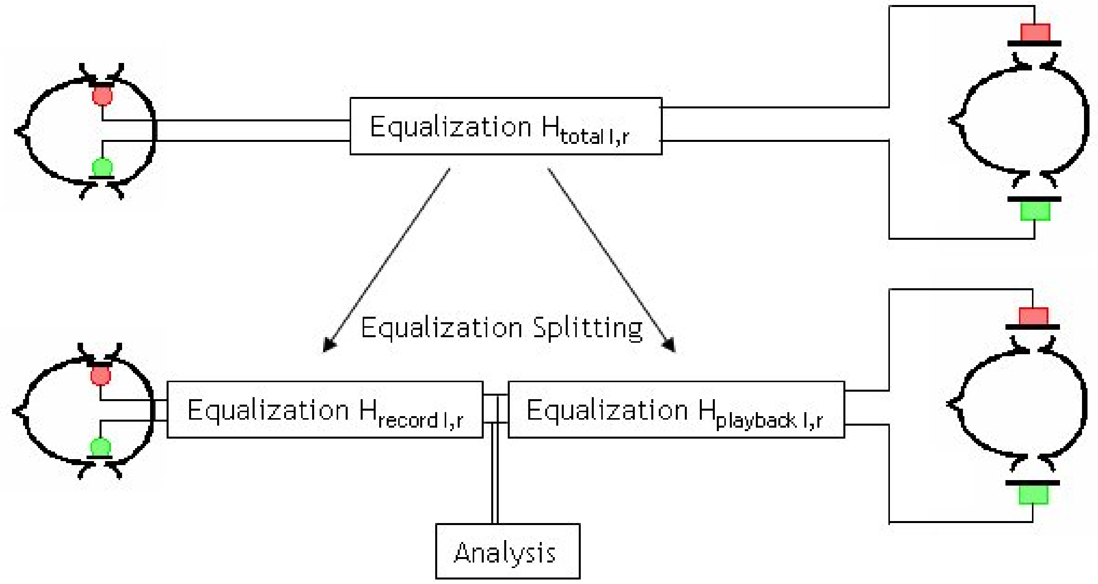


Figure 6: Splitting of the equalization for analysis purposes

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 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_{record} combined with the playback equalization $H_{playback}$ results in the correct total equalization H_{total}, there are different types of playback equalizations corresponding to the different recording equalizations. For an aurally accurate playback, the playback equalization $H_{playback}$ must compensate for the H_{record} equalization and supplement it so that the total equalization H_{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 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.

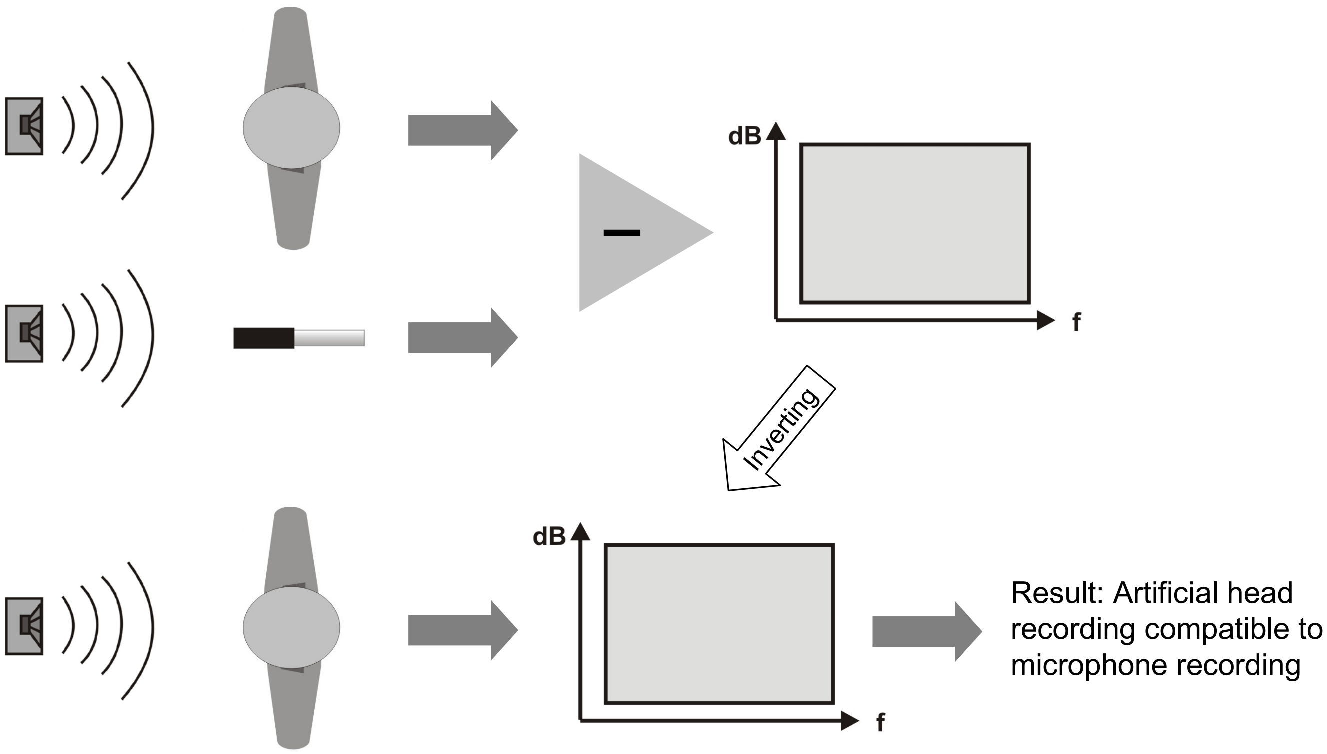


Figure 7: Measurement for determining the free field equalization (simplified)

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.
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.

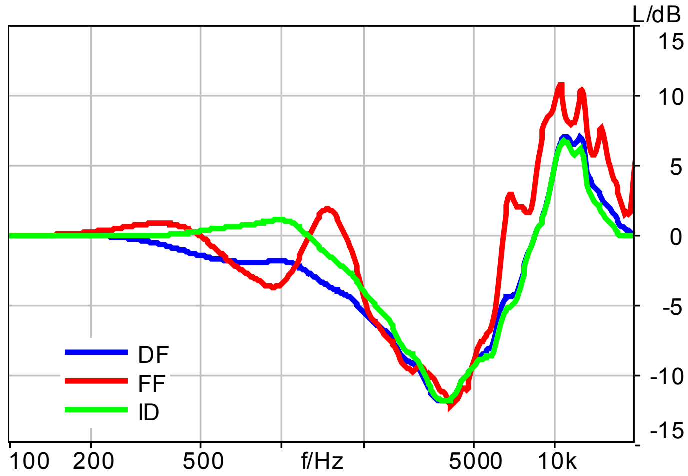


Figure 8: Frequency curves of the different equalization functions

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 best represents the real-life impression. Furthermore, such a comparison shows which of the two channels should get special attention.

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 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 ID equalization is available for binaural head microphone recordings, i.e. 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. This headset is a binaural recording and playback unit that can be connected to the SQuadriga frontend from HEAD acoustics (figure 10).

Figure 10: SQuadriga with BHS

BHS 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 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. The equalization of the BHS recording is performed automatically by the SQuadriga frontend, so the BHS recording is immediately available with the correct equalization and ready to be analyzed just like an artificial head recording.

Playback of Binaural Recordings

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 $H_{playback}$ must be chosen so that together with the recording equalization H_{record}, it results in the total equalization H_{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 provides a Programmable Equalizer (PEQ, figure 11). With this equalizer, the correct signal level and the correct equalization for the playback can be adjusted. Furthermore, this equalizer contains filters that allow possible variations of the headphone transfer characteristics to be compensated for. These filters are calibrated individually for the specific headphone units shipped with the PEQ. But in addition, standard filters for other headphone types are provides as well.

Figure 11: PEQ V with electrodynamic headphone HD IV.1

Selecting the playback equalization

The programmable equalizer is programmed with all the required filters for the playback equalization $H_{playback}$, so the binaural recording can be equalized and played back correctly in order to achieve an acoustic impression comparable to the original sound field. The playback equalization can be set manually or automatically on the PEQ. The PEQ IV requires HMS data[1] for an automatic adjustment. The successor model PEQ V no longer needs HMS data, as it can be remotely controlled via the USB interface.
When recordings are played within the ArtemiS analysis software, it is possible to pass HMS data to the PEQ. The required setting can be made in a Properties dialog that can be opened via the "Settings" command in the "Options" menu (see figure 12). As soon as the option "Generate HMS/RPM pulse" is enabled, ArtemiS generates HMS data for the playback with the PEQ. If information about the equalization used is present in the recording file, this information is used and forwarded. If this information is not available, HMS data is generated using the equalization type specified in the line labeled "Default Equalization".

[1] HMS data contain information about the measurement range and the equalization type used for a recording.

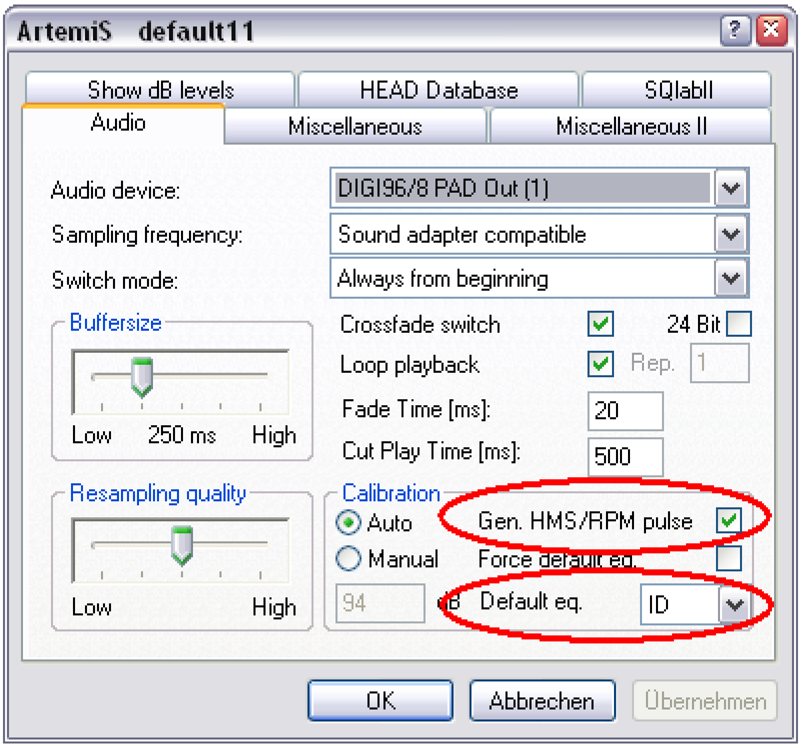


Figure 12: Properties dialog for enabling the use of HMS data

Selecting the playback level for artificial head recordings

Besides the selection of the correct equalization setting, the selection of the correct playback level is essential for an accurate, calibrated playback. For artificial head recordings, the measurement range is predefined in steps of 10 dB (84 dB, 94 dB etc.). This measurement range corresponds to the signal level that needs to be set on the PEQ for a correct playback. The measurement range information is stored in the file and can be forwarded to the PEQ to make it adapt itself automatically. For the PEQ IV, it is also required to enable the option “Generate HMS/RPM pulse” as described above. That way, HMS data are generated that the PEQ IV requires for the automatic adaptation. The PEQ V model does not require HMS data any more, because as long as it is in the “Auto” mode, it is configured correctly via the USB interface even if the “Generate HMS/RPM pulse” option is disabled.
Just like the equalization, it is also possible to adjust and change the signal level manually on the PEQ IV and PEQ V. In this case, it is important to note that an improper setting will cause the recording to be played at a different level than it was recorded with. For a correct playback of artificial head recordings, the PEQ must be set to the level corresponding to the original measurement range.

Selecting the playback level for BHM recordings

The playback of a BHM recording is also normally done via the ArtemiS analysis software, a PEQ and headphones. Since only the ID equalization is available for BHM recordings, the PEQ must always be set to ID for the playback. The level setting on the PEQ must match the measurement range chosen for the recording. Depending on the recording configuration, a BHM recording can contain HMS data. If this is the case, the measurement range is one of the known 10 dB steps, so the PEQ must be set to match the correct level. This can be done automatically

or manually. When using a PEQ IV, the automatic adjustment requires the option “Generate HMS / RPM pulse” to be active.
If the option “Generate HMS / RPM pulse” is not activated for the playback in ArtemiS, the PEQ IV must me set to a level value that is as close as possible to the measurement range of the recording.
If the two channels of the BHM measurement have different measurement ranges, the PEQ must be set to a level matching the higher measurement range. The channel with the lower measurement range is automatically converted by ArtemiS, so this channel is also played back with the correct level.

Playback with SQuadriga and the BHS Headset

Since the BHS headset is a combined recording and playback unit, not only the recording, but also the playback takes place directly via the BHS. The combination SQuadriga / BHS allows not only BHS recordings, but also other binaural recordings to be played back, for example artificial head recordings. Once the function “Auto” in the Monitoring menu is selected, the equalization and level settings for the playback are adjusted automatically by SQuadriga. Furthermore, it is also possible to select the settings for the equalization and the level manually. As with the PEQ, playback via SQuadriga requires that the same equalization is used for the playback that was used for the recording. The playback level must equal the measurement range of the recording.

Playback with a fixed level range

Figure 12 shows a Properties dialog of ArtemiS showing the adjustment function for the playback level to the left of the option “Generate HMS / RPM pulse”. Two options are available: “Auto” or “Manual”. All procedures described above for configuring the PEQ for the playback of binaural recordings refer to the “Auto” setting. The following paragraph describes how to use the “Manual” setting.

Selecting the “Manual” option activates the area below it, where a fixed playback level can be set. Each file played back with ArtemiS is now converted to the level specified here. The playback level should be chosen so that it corresponds to the 10 dB steps (84 dB, 94 dB etc.) predefined in the HMS data. For a playback with the correct level, the PEQ must be set to the same value. If the PEQ is set to “Auto”, the level values are set automatically (for the PEQ IV, this requires the option “Generate HMS / RPM pulse” to be active). Settings can be made manually as well. This function allows ArtemiS to play back files that are using a different measurement range and do not contain HMS data. In that case, a level range is used for the playback to which the PEQ must be set and will be used for the playback of all files. Note that the level range should be large enough to avoid distortions caused by clipping.

Appendix: 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 13 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.

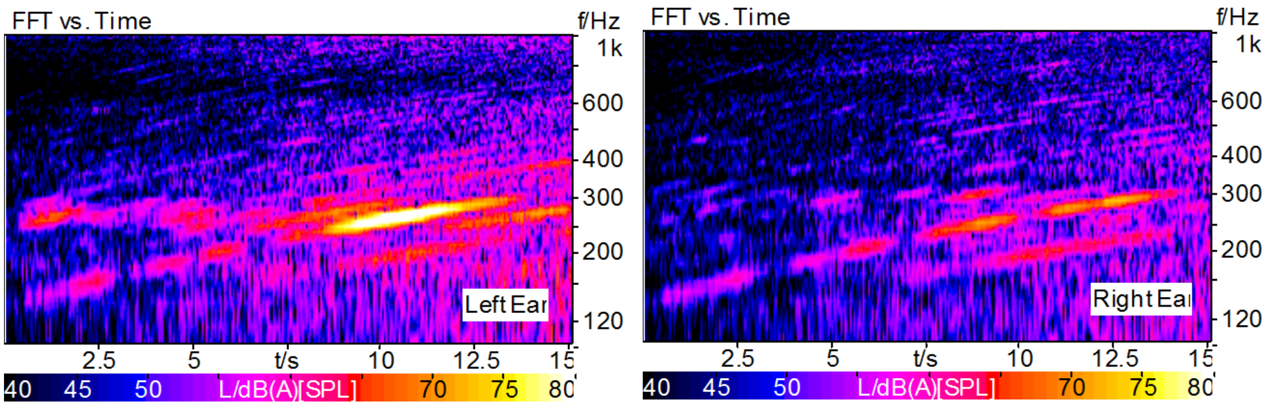


Figure 13: FFT vs. time analysis of a vehicle interior noise

Figure 14 shows the result of a Specific Prominence Ratio analysis of the vehicle interior noise shown in figure 13. 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.

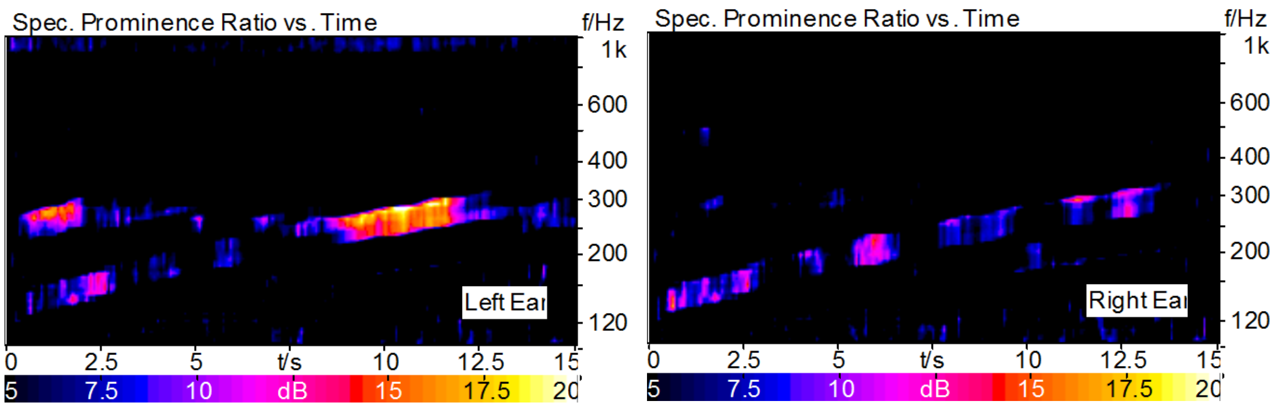


Figure 14: Spec. Prominence Ratio analysis of the vehicle interior noise from fig. 13

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 15 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.

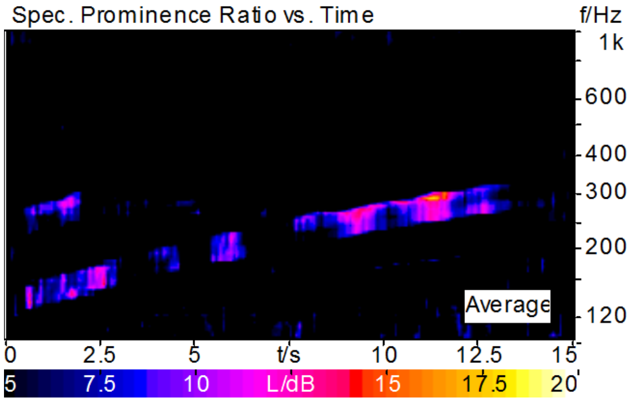
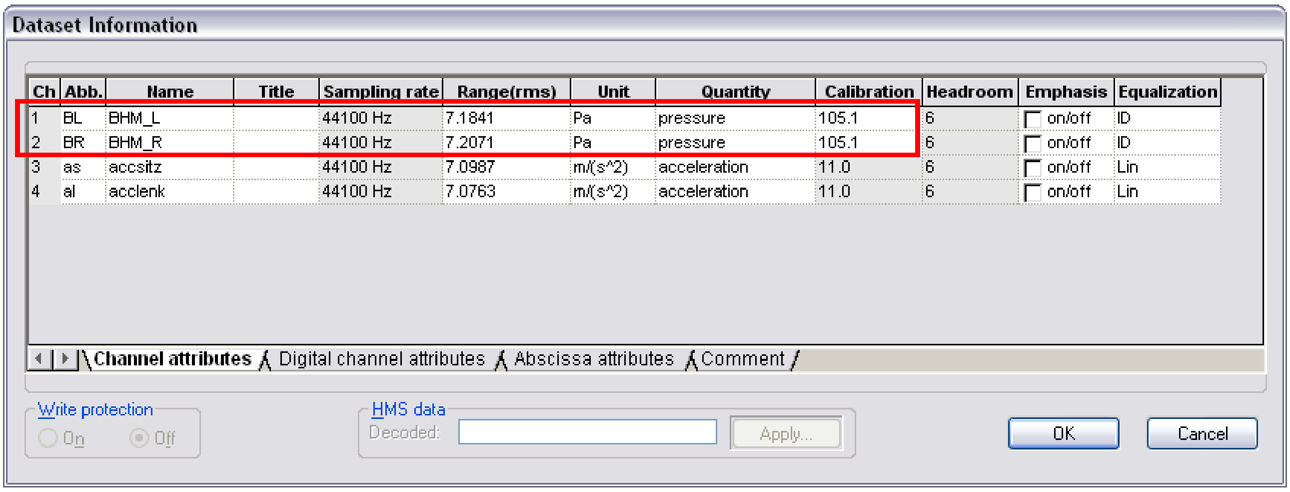


Figure 15: Averaged values of the Spec. Prominence Ratio analysis from fig. 14

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

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 16). [2]



Ch	Abb.	Name	Title	Sampling rate	Range(rms)	Unit	Quantity	Calibration	Headroom	Emphasis	Equalization
1	BL	BHM_L		44100 Hz	7.1841	Pa	pressure	105.1	6	on/off	ID
2	BR	BHM_R		44100 Hz	7.2071	Pa	pressure	105.1	6	on/off	ID
3	as	accsitz		44100 Hz	7.0987	m/(s^2)	acceleration	11.0	6	on/off	Lin
4	al	acclenk		44100 Hz	7.0763	m/(s^2)	acceleration	11.0	6	on/off	Lin

Figure 16: Dataset information of the BHM recording

[2] 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.

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 level range (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 questions to the author? Please contact us at imke.hauswirth@head-acoustics.de.
We are looking forward to your feedback!