

Wave Export in ArtemiS

ArtemiS offers the possibility to export sound files stored in HDF or DAT format to WAV format. This function allows the user to make recordings available to customers or coworkers who are unable to play back HDF files. Furthermore, WAV files can be embedded, for example, in a PowerPoint® file and played during a presentation. This ensures a much more meaningful experience of, for example, improvements in sound quality during a presentation than is possible by showing diagrams alone.

The Wave Export function of ArtemiS provides various options for configuring the export for different applications. This Application Note explains the individual settings.

Properties Dialog of the Wave Export Function

After adding a Wave Export item to the Results Pool, the Properties dialog of this item can be opened by clicking on it with the right mouse button. In this dialog, the desired settings for the WAV export can be configured. The user interface is shown in figure 1.

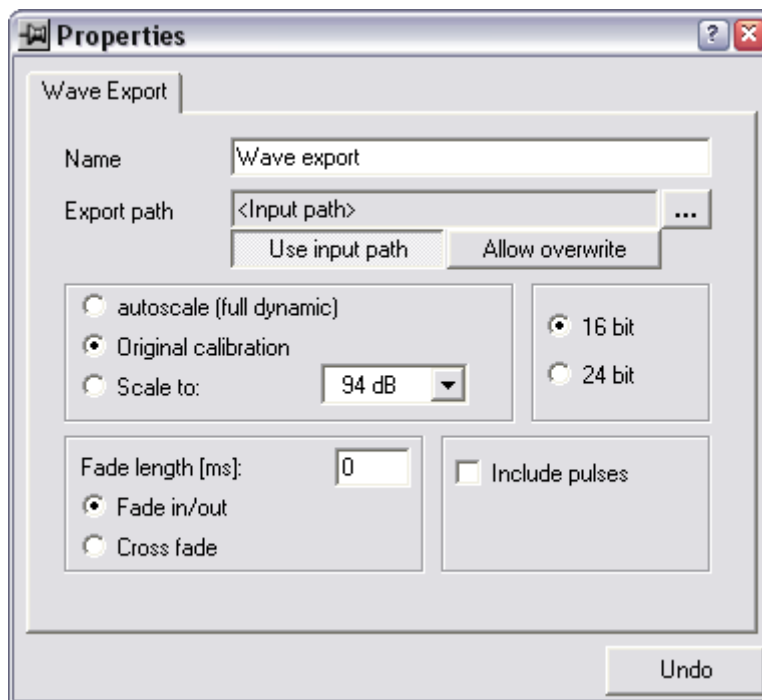


Figure 1: Wave Export Properties dialog

In the top area of the Properties dialog, you can specify the name of the pool item. Below, you can enter the path where the WAV file is to be saved. If the “Use input path” button is active, the WAV file is saved to the same folder as the original file, and the Export path field reads “<Input path>”. Clicking on the button with the three dots “[...]” disables this function and opens a file selection dialog, where the desired path can be selected. The “Allow overwrite” button allows

existing files of the same name to be overwritten, which is useful if you export the same file repeatedly.

Dynamic Range

In the dynamic range area, three options are available: “Autoscale”, “Original calibration” and “Scale to”, see figure 2).

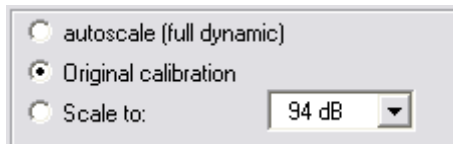


Figure 2: Dynamic range area of the Wave Export Properties dialog

This selection determines the dynamic range of the created WAV file depending on the original dynamic range. This allows files with different dynamic ranges to be adapted to each other. Furthermore, a suitable selection can optimize the signal-to-noise ratio of the WAV file.¹

With the “Autoscale” setting, the WAV file is scaled so that the highest signal level occurring in the original file uses the maximum available dynamic range in the WAV file.

The “Original calibration” setting retains the relative signal level of the original file for the WAV file.

The third setting allows a new dynamic range to be specified for the WAV file. The values of the original file are then rescaled so that they fit the new dynamic range of the WAV file.

The following table schematically shows the effect of the different settings.

Setting	Signal level of original file	Signal level of WAV file
Autoscale		
Original calibration		
Scale to:	 Original dynamic range: 94 dB	 User-defined (new) dynamic range: 84 dB

Table 1: Effect of dynamic range settings

¹ The high numeric resolution of a HDF file with floating point numbers ensures a high resistance to overmodulation. To convert the signal into a WAV file, the data need to be converted to an integer format (whole numbers). In order to achieve the best possible signal-to-noise ratio, the available range of integer numbers must be utilized as fully as possible.

These different settings are useful for different applications. For example, an application may require that data with different dynamic ranges are to be converted into WAV files that should have similar volume levels when listened to. For example, if two original files have dynamic ranges of 94 dB and 104 dB, respectively, the “Original calibration” setting would cause the volume of the WAV file generated from the 104 dB original to be too low compared to the 94 dB file. This can be avoided by selecting the “Scale to” option with a level of “104 dB”. That way, both files are scaled to the same dynamic range, resulting in a consistent level of the two WAV files relative to each other. Note that in order to prevent overmodulation, the highest of the original dynamic ranges must be selected. However, if the original signals are badly adjusted so they do not utilize their entire dynamic range, it is also possible to scale to a lower range in order to improve the dynamic range of the WAV file. On the other hand, the amplification should not be too high, as the background noise of the signal might be increased significantly.

If a playback system from HEAD acoustics, such as a PEQ IV or a PEQ V, is used for playing the WAV files, the device can be adjusted to the dynamic range of the original file or to the newly selected dynamic range, thus ensuring accurate playback levels. For exported artificial head recordings, the appropriate equalization can also be configured on the PEQ, allowing an aurally accurate playback. A file with a dynamic range of 94 dB exported with the “Original calibration” setting can thus be played back with the correct level by a PEQ, provided that the device is set to 94 dB and a sound card with an AES interface is used.

Resolution

The resolution of the WAV file to be generated can be set to either 16 bit or 24 bit (see figure 3).

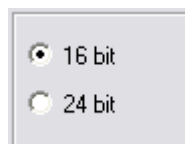


Figure 3: Resolution area of the Wave Export Properties dialog

Of course, a 24-bit file has a better resolution than a 16-bit file; however, the following should be considered when choosing the resolution:

A resolution of 24 bit significantly increases the memory requirements compared to a 16-bit file. If the original file has a resolution of 16 bit only, the resolution is not improved by choosing the 24 bit setting – this only leads to a bigger file size.

Not all software that is suitable for playing back WAV files is capable of processing 24-bit files. Taking this into account, users can create a WAV file with the desired resolution in ArtemiS.

Fading in and out

To prevent crackling noise at the beginning or end of WAV files during playback due to sudden level jumps, ArtemiS offers the possibility to fade the signals in and out. Two options are provided: “Fade in/out” and “Cross fade” (see figure 4).

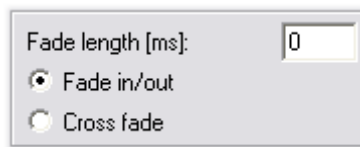


Figure 4: Fading area of the Wave Export Properties dialog

The “Fade in/out” function fades a single file, which is to be played only once at a time, in and out, where the fading period can be specified in milliseconds. The “Cross fade” function alters the file so that no interruption occurs when playing the file repeatedly in an endless loop. This is achieved by moving the last part of the file to the beginning and merging it with the fade-in section, resulting in a smooth transition between the end and the beginning of the file. Since the “Cross fade” function overlays the beginning and the end of the file for the specified period of time, the total length of the resulting file is reduced by that period. In the case of the “Fade in/out” function, the specified period of time refers to both the beginning and the end of the file. The total length of the file is not reduced. To deactivate the fading function, just set the period to “0”.

Whether fading is required and how long the fading period should be cannot be stated generally. It widely depends on the signal length and the signal characteristics. A long signal can be faded in over a longer period of time without causing an unnatural impression, whereas this is not possible with a short signal. If the signal has been recorded so that its level is low at the beginning and increases afterwards, additional fading may not be required at all. For a recording that has been cut afterwards in ArtemiS, so it starts suddenly with a high level, it is highly recommended to choose a sufficient fading period. For a very short, stationary signal, it is advisable to play it in an endless loop and therefore to choose the “Cross fade” function. The continuous, uninterrupted playback allows the listener to tune in to the sound, whereas a repeated playback of short, non-stationary signals is quickly perceived as unpleasant by listeners.

Exporting Pulse Information

With the option “Include pulses”, the Wave Export function can be configured so that any pulse information encoded in the signal is also transferred into the resulting WAV file. When exchanging WAV files with included pulse data, it is advisable to use a test file to check whether the target software is capable of reading the WAV files correctly, so both the audio channels and the pulse channels are imported properly.

The ArtemiS Wave Export writes pulse information from the first channel of the original file to the first channel of the WAV file etc. If some channels are disabled during the export, this can cause pulse data to be shifted. For example, if the first channel of the original file is disabled, but contains pulse information, its pulse data is written to the first channel of the exported WAV file. That way, only those channels whose audio data are actually required by the user must be exported. As soon as the “Include pulses” option is checked, all available pulse information is exported, regardless whether the corresponding audio channel is exported or not.

General Notes on the Wave Export

Exporting Multi-channel Files (> 2 Channels)

ArtemiS is capable of converting files with more than two channels to the WAV format. Before exporting large amounts of data, it is advisable to check whether the target software supports multi-channel WAV files.

Resampling in ArtemiS

Most sound cards and programs for the playback of WAV files can only play files with standard sampling rates (e.g. 44.1 kHz). Files with other sampling rates are converted to one of the standard sampling rates prior to playback. Since some playback programs and sound card drivers are unable to perform such a conversion in high quality, it is recommended to first resample original files with non-standard sampling rates in ArtemiS. ArtemiS provides algorithms ensuring a high precision of the resampling process. To use this function, select the "Resampling" item in the Filter Pool. In the Properties dialog of this element, the option "Use Audio Sampling Rates" must be checked. With this function, ArtemiS performs a conversion from the original sampling rate to the nearest standard sampling rate.

Oversampling of Pulse Information

Pulse information recorded with oversampling may not be stored correctly in an exported WAV file, as the WAV format does not support oversampled pulse information. To improve the accuracy of pulse information in the WAV file, first a file export with the "16 bit" setting should be performed in ArtemiS. If this newly created file is then used for the WAV export, the deviations between the pulse information of the WAV file and the original file are reduced significantly.

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