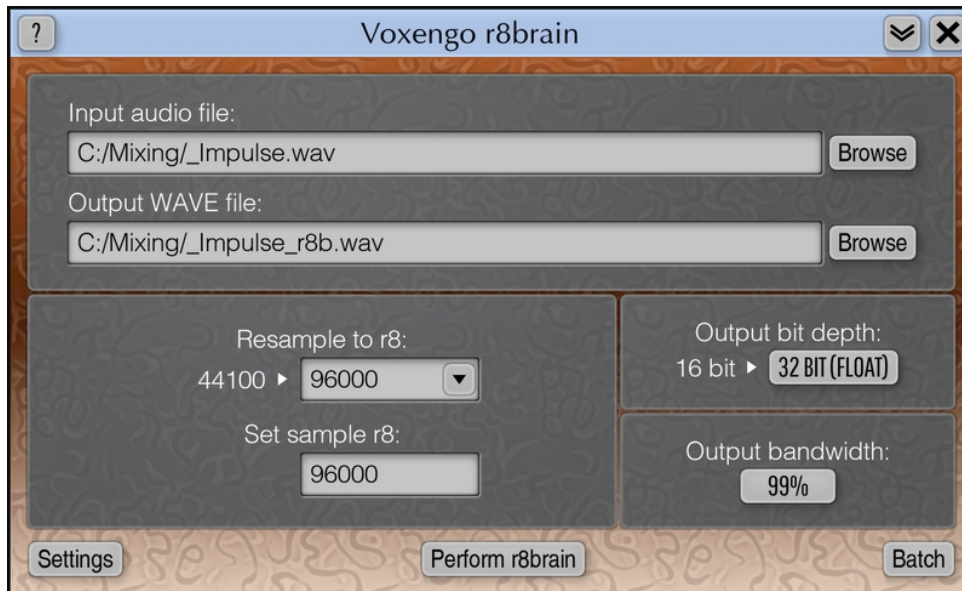

Voxengo r8brain User Guide



Version 2.10

<https://www.voxengo.com/product/r8brain/>

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Introduction

r8brain is an audio sample rate converter application. It was designed to allow both amateur and professional users to perform high-quality sample rate conversion of uncompressed PCM WAV files. For example, if your project WAV file was recorded or saved as 48000 (or 96000) sample rate file, and you need it to bring to MP3 or CD format, then you will likely need to convert it to 44100 sample rate first. This utility application will help you to perform such conversion using its high-quality algorithm. Virtually any source and destination sample rates are supported by this converter, thus permitting non-conventional uses.

r8brain also supports batch processing which is essential if you need to transfer all your project's audio data from one sample rate to another.

Features

- Very fast conversion
- Multi-threaded batch conversion
- Mono, stereo and multi-channel support
- Virtually any input and output sample rate
- Full WAV file support
- RF64 file support
- Broadcast wave extension support
- Markers and loops support
- Wave64, AIFF support (on input only)
- Batch processing
- 64-bit floating point processing
- All sample rates support

Compatibility

This application is compatible with 64-bit Windows (Vista, 7, 8, 10 and later versions) computers (2.5 GHz dual-core or faster processor with at least 4 GB of system RAM required).

User Interface Elements

Note: All Voxengo applications and plug-ins feature a highly consistent user interface. Most interface elements (buttons, labels) located on the user interface are the same in all Voxengo applications and plug-ins. For an in-depth description of these and other standard features and user interface elements please refer to the “Voxengo Primary User Guide”.

Main Window

The main window allows you to select a single input file to process together with several options.

The "Resample to r8" (resample to rate) performs the actual resampling. r8brain displays file's sample rate when you choose it. Note that the resampling process by its nature cannot increase “quality” of the file being processed. Moreover, when changing the sample rate down it reduces its quality. For example, increasing the sample rate of a file from 44100 to 96000 will not make the file to have a “better” quality. However, depending on the sound card you are using and its hardware implementation, sound files at 96000 may sound subjectively better. But this does not mean resampling is adding a quality to the sound file. Note that when the “Resample” rate is equal to input file's, only the bit depth conversion will take place.

The "Set sample r8" (set sample rate) forces a specific sample rate in the output file. This option does not perform resampling, but it allows you to resample the file to the desired sample rate, but then force the file to have a different sample rate.

The “Output bit depth” selector allows you to change the bit depth of the input file. r8brain performs industry standard TPDF dithering when either 8, 16 or 24 bit depth was chosen. Dithering is not applied to floating-point output files.

The “Output bandwidth” selector specifies the “quality” of resampling which means the amount of spectral information to preserve during sample rate conversion process. 99% is the highest quality setting.

Batch Processing

Batch processing allows you to resample several wav files in the specified folder to the specified output bit depth and sample rate. r8brain batch uses multi-threaded processing.

The “Delete original files” option replaces existing files after conversion. Use with care! A good practice when using this option is to create a copy of a folder you'd like to process and perform batch processing over the copied folder.

The “Scan subfolders” option performs sample rate conversion of files in all subfolders of the specified folder. All supported filetypes present in the folders will be converted to the “wav” files: this includes “.wav”, “.w64”, “.aif” and “.aiff” files.

The “Result file prefix” string is appended to the resulting files that were batch-resampled. For example, if the input file name is “Cymbals.wav” and the prefix is equal to “A”, the resulting filename will be “ACymbal.wav”.

The “Result file suffix” string appends a suffix to the names of the result files. For example, if input filename is equal to “Drumbus.wav” and the suffix is equal to “_converted”, the resulting filename will be “Drumbus_converted.wav”. It is not necessary to specify the “.wav” extension in the suffix.

Credits

DSP sample rate conversion algorithm was created by Aleksey Vaneev.

Graphics user interface code by Vladimir Stolytko. Graphics elements by Vladimir Stolytko.

This application is implemented in multi-platform C++ code form and uses “zlib” compression library (written by Jean-loup Gailly and Mark Adler), “LZ4” compression library by Yann Collet, Intel IPP and run-time library by Intel Corporation (used under the corresponding licenses granted by these parties).

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