Voxengo r8brain PRO User Guide
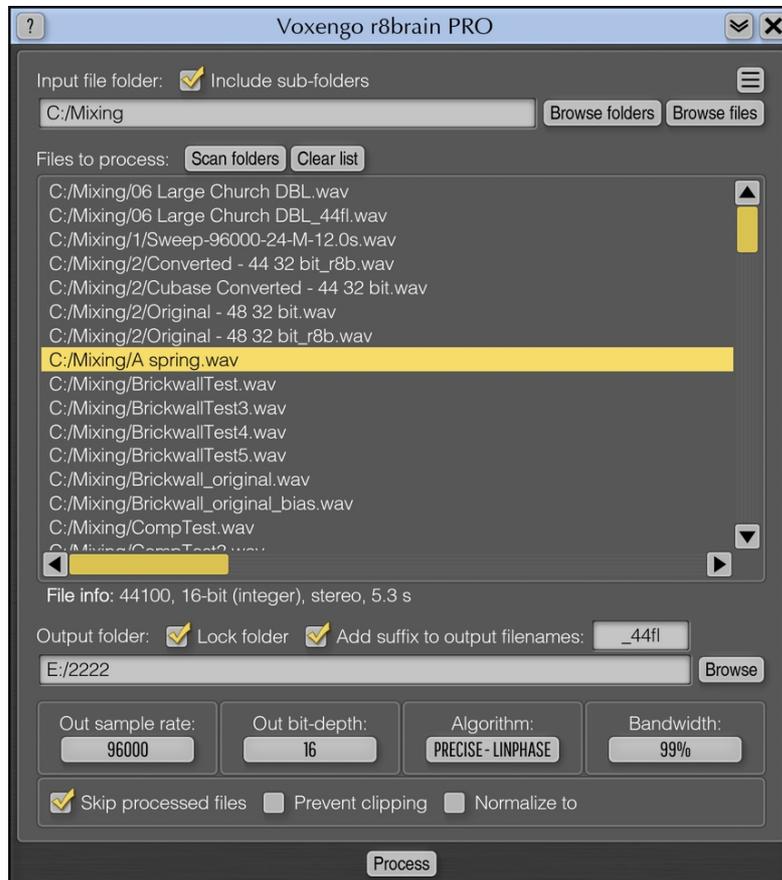
Version 2.12

https://www.voxengo.com/product/r8brainpro/

Contents

Introduction

r8brain PRO is a professional sample rate converter tool software designed to deliver an unprecedented sample rate conversion (SRC) quality. Unlike other existing SRC algorithms available on the market, r8brain PRO implements sample rate conversion processing in its full potential: interpolation and decimation steps without exploiting any kind of simplifications; the signal is resampled in a multi-step manner using a series of intermediate sample rates based on whole-numbered up/down factors. This makes conversion perfect – both in signal-to-noise and timing precision aspects. Such whole number-factored SRC can be considered a golden standard in sample rate conversion since it is not subject to jitter nor timing errors.

Like many existing SRC algorithms, r8brain PRO offers you a linear-phase conversion mode. But more importantly, you also have an option of using the minimum-phase conversion mode, which finally brings SRC with true analog qualities to affordable digital audio workstations: in this mode, r8brain PRO works like an ideal digital-to-analog converter followed by an analog-to-digital converter, to resample the audio. This eliminates pre-ringing associated with linear-phase designs while introducing a minimal amount of phase coloration.

r8brain PRO can read mono, stereo, and multi-channel files in WAV, RF64, Wave64, and AIFF file formats, creating 16-, 24-, 32-, and 64-bit mono, stereo, and multi-channel WAV (RF64) files in fixed- and floating-point formats. EBU BWF (broadcasting) extensions, extensible wave format, markers, sample loops, and textual data residing inside a file are also supported. For the sake of convenience, r8brain PRO allows you to perform multi-threaded batch conversions that utilize the full potential of your computer.

r8brain PRO's bit-depth conversion uses an industry-standard "flat" TPDF dithering. We have decided not to implement noise-shaping dithering because audio production software available on the market usually offers the user a noise-shaping dithering of some kind already (you can also use our Elephant plug-in for this feature). We also based our decision on the fact that the sample rate conversion process often adjusts peak profile of the original program material, thus, in many cases, making a subsequent peak-limiting a necessity. To prevent output audio from clipping we have implemented a level normalization feature.

Features

- Mono, stereo, and multi-channel file support
- Linear-phase and minimum-phase modes
- Selectable output signal bandwidth
- Automatic peak normalization
- Full WAV file support
- RF64 input and output support
- Broadcast wave extension support
- Markers and sample loops support
- Wave64, AIFF support (on input only)
- Multi-threaded batch conversion
- 64-bit floating point processing
- Standard sample rates support

Compatibility

This application is compatible with 64-bit Windows (Vista, 7, 8, 10 and later versions, if not announced otherwise) and macOS (10.11 and later versions, if not announced otherwise, 64-bit Intel and Apple Silicon processor-based) computers (2.5 GHz dual-core or faster processor with at least 4 GB of system RAM required). A separate binary distribution file is available for each target computer platform.

User Interface Elements

Note: All Voxengo plug-ins feature a highly consistent user interface. Most interface elements (buttons, labels) located on the top of the user interface are the same in all Voxengo 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 "Input file folder" is the name of the folder containing the original (input) WAV, Wave64 or AIFF files. To select a folder, use the "Browse folders" button, or use the "Browse files" to select one or more specific files from a folder. The "Scan folders" button can be used to find all supported files in the specified folder if you it entered manually. The "Include sub-folders" option performs the file scan in all sub-folders of the specified folder.

The "Files to process" list box lists all the files which should be processed when the "Process" button is pressed. The right-mouse-click (or the "X" button) can be used to remove entries from the list. The left-mouse-double-click can be used to open the currently selected file using the default associated application. You can drag&drop files into this list from the operating system's file explorer/finder. Press the "Clear list" button to remove all items from the list.

The "Output folder" specifies the folder where resulting files should be written. This field is filled automatically after each new input file folder selection. Note that r8brain PRO preserves the original folder structure during conversion, and the resulting files will reside in sub-folders with the original names.

The "Lock folder" switch locks the output folder so that folder scanning, browsing for folders and files, drag & drop, do not change it. The output folder can still be changed either manually or via the "Browse" function.

The "Add suffix to output filenames" switch allows you to append any specified suffix to the output filenames. This allows you to easily identify resampled copies of your files. Please note that disabling this option may overwrite the original files if the output folder is equal to the input folder. Disable this option with care!

The "Out sample rate" selects which sampling rate resulting files should have. 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 better. But this does not mean resampling is increasing a quality of the sound file.

The "Out bit depth" selects which bit-depth resulting files should have. r8brain PRO performs the industry standard TPDF dithering when non-float bit depth was chosen.

The "Algorithm" selects algorithm type to use during sample rate conversion. The "Precise – LinPhase" performs linear-phase resampling using precise algorithm. The "Precise – MinPhase" performs minimum-phase resampling using precise algorithm. The "Fast – LinPhase" performs linear-phase resampling using the fast algorithm also found in the free "r8brain" converter. Note that all algorithms produce excellent results, but the "Precise" algorithms produce an ultimate-quality conversion.

The "Bandwidth" selects the "quality" of resampling which means the amount of spectral information to preserve during sample rate conversion process. 99% is the highest quality setting.

The "Skip processed files" switch enables skipping (ignoring) of previously processed/resampled files even if they are present in the list. r8brain PRO writes a standard "ISFT" ("Software") comment field to the output file to distinguish between processed and original files.

The "Prevent clipping" switch enables automatic normalization if the signal clipping was detected during conversion.

The "Normalize to" dBFS option can be used to normalize the resulting file to the specified dB (full scale) peak value. Peak value should be specified in decibels, with decimal part separated with the point (e.g. -0.3).

The "Process" button starts batch-processing of the files listed in the file list. Since batch processing is multi-threaded, it may utilize all available cores of the computer if there are many files in the list. Any file that could not be processed will be listed again after the batch processing finishes. This allows you to check/edit or remove any such file from the list, and continue the batch processing later.

The "Settings" button above the "Browse files" button opens the settings window where you can fine-tune visual aspects of application's interface. Authorization/registration details can be also entered in this window.

What Is Whole-Number-Factored Resampling?

The multi-step whole-number-factored resampling decomposes the whole resampling process into independent resampling steps that target varying intermediate sampling rates. Each step performs up-sampling or down-sampling by a whole number (integer) factor. Each pair of up and down steps uses its own specific low-pass filter.

For example, consider 44100 to 96000 Hz resampling steps:

1. Input = 44100

2. 8 up = 352800

3. 1/7 down = 50400

4. 8 up = 403200

5. 1/7 down = 57600

6. 5 up = 288000

7. 1/3 down = 96000

8. Output = 96000

Credits

DSP sample rate conversion algorithm, user interface layout by Aleksey Vaneev.

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

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, "base64" code by Jouni Malinen, FFT algorithm by Takuya Ooura, Intel IPP and run-time library by Intel Corporation (used under the corresponding licenses granted by these parties).

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