Sonic Studio Process

User Manual

www.sonicstudio.com

Sonic Studio Process User Manual Table Of Contents

1.0 .... Sonic Studio Process Overview

To increase your productivity and to free up soundBlade and Amarra from tasks that are time consuming and suited for unattended execution, Sonic Studio developed the Sonic Studio Process application. Sonic Studio Process is a standalone application that is used for ultrafidelity sample rate conversion and DeCrackle batch processing.

Note that the Sonic Studio Process app that's included with Amarra has a limited feature set. Not all parameters listed in this document are available.

DeCrackle is not available for Amarra users; it requires our Manual DeClick option to be installed on both your iLok and Mac.

The Sonic Studio Process consists of two windows, each represented by its own tab in the top right corner. The first window is 'Parameters' and is accessed via the "Params" button. In this section, sound files can be added and selected for processing, processes can be assigned to each file, an output folder for processed files can be set and parameters can be defined for each process.

The second window of the Sonic Studio Process is the 'Queue Manager' and is accessed from the "QUE MGR" button. Once sound files have been assigned a process, processes can be managed via the Queue Manager.

1.1 The Parameters tab

By default, the Parameters Tab is selected when Sonic Studio Process opens. The Parameters Window largely consists of 2 areas, the “Soundfiles” area and the “Processes” area.

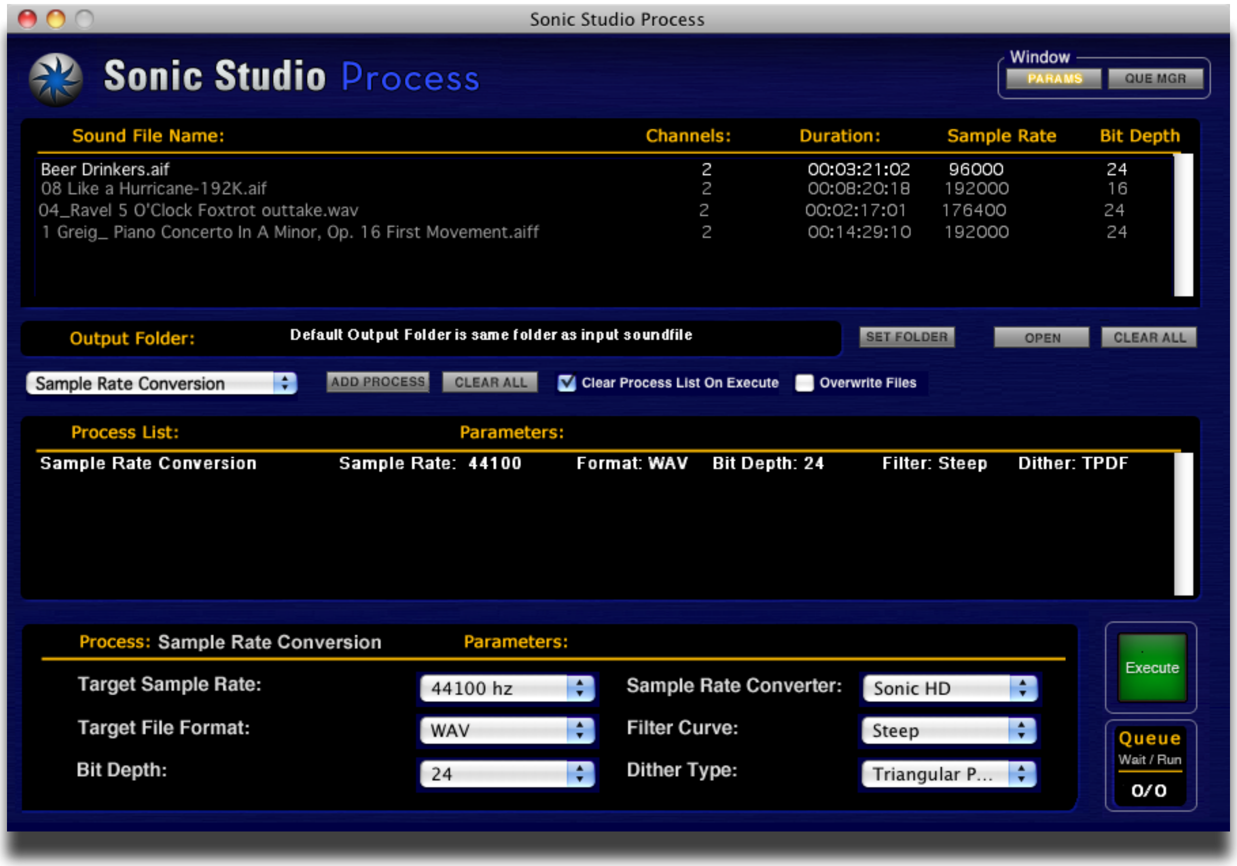


Figure 01: Sonic Studio Process showing the Params Window

1.1.1 Soundfiles Area

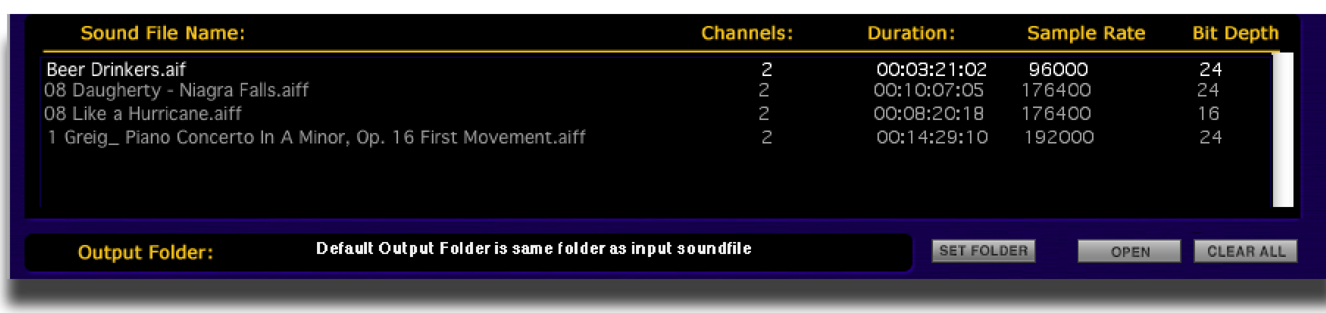


Figure 02: Soundfile area of the Parameters Window

In the Soundfiles area, sound files can be selected and added by:

Clicking the ‘Open’ button

or

Dragging and dropping one or more files from the Finder into this area.

HINT: With either the Open button or drag and drop, multiple sound files can be added to the list by selecting all the files you need to work on.

Note that, only uncompressed files - WAV, AIFF or CAF - are supported at this time. Convert MP3, FLAC, AAC, ALAC or other compressed files to AIFF or WAV before adding to Sonic Studio Process.

Once a sound file is added, it is placed in the Sound File Name list. This list displays the file name, number of channels, duration, sample rate, word length or bit depth of each file.

In the 'Output Folder' field, the target folder for newly created sound files is specified. By default, the Output Folder is the same as the input file.

To select a different Output Folder, click on the 'Set Folder' button. A standard Mac OS browser opens, allowing you to navigate, create and/or choose the desired destination folder.

To remove sound files from the list, click to select the list entry and delete it by hitting the backspace or delete key. To clear the list of all sound files, click the Clear All button to the lower right of the Soundfiles list.

1.1.2 Processes Area Sample Rate Conversion

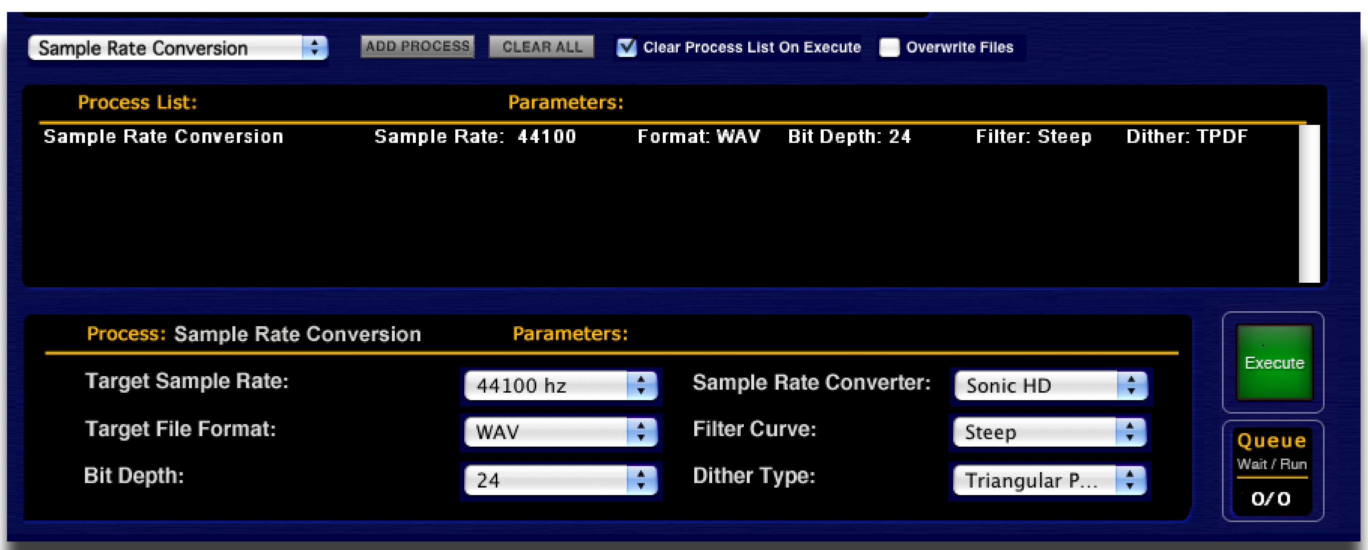


Figure 03: Process area of the Parameters Window

Once all sound files have been added to the Soundfiles area, one or more processes can be applied to each file here in the Processes area. An "Add a Process" menu allows for selection of the desired process(es).

Note that, 'Sample Rate Conversion' is the default process for Sonic Studio Process app.

If you are a soundBlade user and have purchased Sonic Studio's Manual DeClick option, 'DeCrackle' will appear in the Process pull-down menu.

DeCrackle is not available for Amarra users.

When a process is selected, in the space below the Processes area, details and available parameters are shown for the selected Process. This enables adjustment or selection of further options, depending upon the Process selected.

Processes can be deleted from the processes list by clicking on the process and then hitting the backspace or delete key. To clear all processes from the Processes list, click the "Clear All" button to the right of the Processes list.

Once all necessary processes have been selected and placed in the right processing order, they can be automatically performed upon the selected sound files by clicking the "Execute" button. The sound files and the processes to be applied upon them are then "queued" and processing commences. To indicate this action, the "Queue" indicator below the Execute button shows the number of sound files and processes queued as Wait/Run.

Once the processes are being executed, all sound files and processes remain in their respective lists. This allows for a quick method of applying different processes to the same file(s) or vice versa. To erase both lists and start new lists for sound files and processes, simply click the "Clear Files" or "Clear All" buttons next to the lists and the Parameters is cleared from its current list of sound files and processes, respectively. To remove all items in the lists automatically, enable the Clear List on Execute option.

Note that, upon initiating a process, the Sonic Studio Process will ask to confirm the destination folder.

Note that the order in which sound files are processed is fixed and dependant upon the order in which sound files and processes are listed in their respective areas. The primary processing order is defined, top to bottom, by the order in which the sound files are listed in the Soundfiles area. The secondary processing order is given by the order of processes in the Processes list. So, in practice, if more than one sound file is listed and multiple processes are listed as well, first the initial sound file in the Soundfiles list is processed by all processes listed in the Processes list in the order in which they are listed. Be aware that each process is performed upon the resulting file of the previous process. When all listed processes are performed upon the first listed sound file, the following sound file is processed with the same processing in the same order. This is repeated until all sound files are processed with all listed processes.

The end result of this is one resulting sound file per process per sound file.

The following processes are available in Sonic Studio Process:

1.1.2.1 Sample Rate Conversion

The double precision sample rate algorithms in Sonic Studio Process SRC mode provide conversions between each of the commonly-used sample rates, and produce 24-bit output in all cases (which should be dithered to the appropriate release resolution if other than 24 bits).The algorithms are designed with careful attention to filter parameters and computation precision in order to preserve sound quality to the highest extent possible.

Because the multi-phase filters needed for the non-integer conversions (e.g. 96 kHz:44.1kHz) require very large numbers of coefficients and customizing of dsp resources, the filters cannot be designed "on-the-fly." In order to offer choice in the sound of the filters, three versions of each filter are offered- Steep, Gentle and Gentlest; the characteristics each version are described in this section.

The Sample Rate Conversion (SRC) process offers:

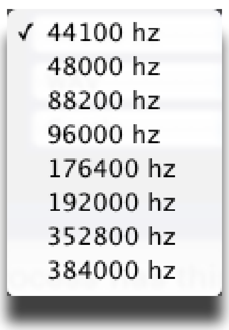
<table>
<tr>
<td>Eight choices of Target Sample Rates. Each preserves the pitch and duration of the original file.


Figure 04: Sample Rate Conversion pull-down menu showing available target sample rates
The Default value is 44100 Hz.</td>
<td>Two Sample Rate Converter types - Sonic HD and Classic:


Figure 05: Sample Rate Conversion type pull-down menu
The Default value is Sonic HD
See the LEARN Section on the next page</td>
</tr>
</table>

<table>
<tr>
<td>Three Target File Formats:


Figure 06: Target Format pull-down menu showing available target file types
The Default value is WAV</td>
<td>Three Filter Curves:


Figure 07: Filter Type pull-down menu
The Default value is Steep
See Section 1.1.2.2 Selecting a Filter Curve below</td>
</tr>
</table>

Two Bit Depths (Word Length):

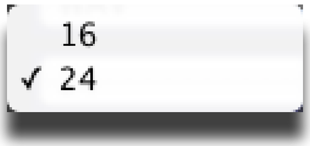


Figure 08: Bit Depth pull-down menu showing available target bit depths
The Default value is 24 bits

Three Dither types:

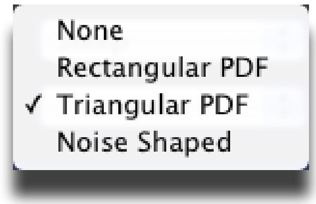


Figure 09: Dither type pull-down menu
The Default value is Triangular PDF
See Section 1.1.2.3 Selecting a Dither Type below

LEARN: 'Classic' SRC is the long-established Sonic "monolithic" SRC - monolithic meaning single stage. The monolithic SRC calculates its coefficient set at runtime when the converter starts up. It's perfect for general SRC.

'Sonic HD' (High Definition) SRC is our new multi-stage algorithm. The multi-stage SRC is based on pre-calculated coefficients for each stage. It's great for general purpose SRC and is recommend for non-integer SRC: 96 kHz -> 44.1 kHz or 48 kHz -> 176.4 kHz for example.

See Chapter 2, 'Sample Rate Conversion Technical Details', below, for more information.

Note: Regardless of the input bit-depth, the output soundfile will be a 24-bit file. This preserves the full dynamic range of your soundfile throughout the conversion process. You can use Sonic Studio Process 'Bit Depth' and 'Dither Type' parameters to change the bit depth if necessary.

1.1.2.2 Selecting a Filter Curve

The heart of the sample rate conversion process is the low pass filter, which is designed to reject aliases (down conversion) or images (up conversion) — both of which are detrimental to sound quality. The filter's response above cutoff must be very steep if it is to be effective at rejecting these artifacts.

Very steep filters can introduce audible distortions, however. Any filter design for sample rate conversion therefore involves trade-offs among a number of factors: aliasing/imaging, inband/stopband ripple, reduced gain at very high frequencies, and time smear. Each of these factors has effects that can vary with the sonic characteristics of the material: sometimes it may be preferrable, for example, to allow a small amount of aliasing in order to preserve other aspects of signal integrity.

To provide the most flexibility in tailoring the conversion process to different types of program material, Sonic Studio Process' SRC offers three different filter choices:

Steep is a Nyquist filter optimized for maximum alias/image rejection

Gentle and Gentlest options trade a limited amount of aliasing/imaging for reductions in other distortions.

HINT: while the Steep Filter is designed for most SRC, your choice of a filter curve for a given sample rate conversion operation should be based on careful audition.

Note: The digital filter that Sonic Studio Process SRC uses exhibits a small amount of amplitude ripple near the cutoff frequency (this is known as the "Gibbs Phenomenon" and is a characteristic of all FIR low-pass filters). Left uncompensated, this ripple could cause source material with high frequencies recorded at high levels to become clipped.

For this reason, SRC reduces the gain at the input of the conversion by 0.1 dB to avoid clipping. Repeatedly converting the same file therefore will gradually reduce its overall level.

1.1.2.3 Selecting a Dither Type

Dither is an intentionally applied form of noise used to randomize quantization error, preventing audible artifacts from being induced when calculating sample rate or bit depth changes. No single dither is a magic bullet, so three different variations of dither are available in Sonic Studio Process.

Sonic Studio Process Sample Rate Conversion performs sample rate conversion with ultra-high precision, and the final step in the process of generating an output file is to round the output to produce a 24-bit file. To preserve fine sonic details such as reverberation tails, Sonic Studio Process SRC affords the option of applying one of three unique dithers to the output file. To assure the highest sound quality, Sonic Studio recommends that you apply dither whenever you use SRC or change bit depth.

Sonic Studio Process Dither Types:

Rectangular PDF (Rectangular Probability Density Function): In this random, flat dither, the noise signal has equal energy at each frequency (think white noise). Rectangular PDF is the most audible type of dither. With rectangular dither the increase in noise varies with the signal. It is recommended for use on LOUD overly compressed music such as rock or pop as it give a certain 'edge' to the sound.

Triangular PDF (Triangular Probability Density Function) [Default]: Shaped triangular dither cuts the noise level at low frequencies while boosting it a high frequencies. This results in the dither noise being less audible. This shaping or weighting reduces the audibility of the noise to a degree. With triangular dither, the added noise level is constant and slightly higher. It is recommended for use on all types of music.

Noise Shaped: This dither type alters the spectral shape of the error that is introduced by dithering such that the noise power is at a lower level in frequency bands at which noise is perceived to be more undesirable and at a correspondingly higher level in bands where it is perceived to be less undesirable. Since this dither is 'packed' at the higher end of the hearing spectrum, this shaping or weighting reduces the audibility of the noise to a degree. This dither is recommended for general use.

Note: In general, triangular PDF dither is preferable.

Note That with either type of dither, the added noise is very small and each dither option acts differently with different types of program material. You should experiment with each dither type to achieve the results that you desire.

Note: Regardless of the input bit-depth, the output soundfile will be a 24-bit file. This preserves the full dynamic range of your soundfile throughout the conversion process. You can use Sonic Studio Process 'Bit Depth' and 'Dither Type' parameters to change the bit depth if necessary.

1.1.3 Processes Area DeCrackle

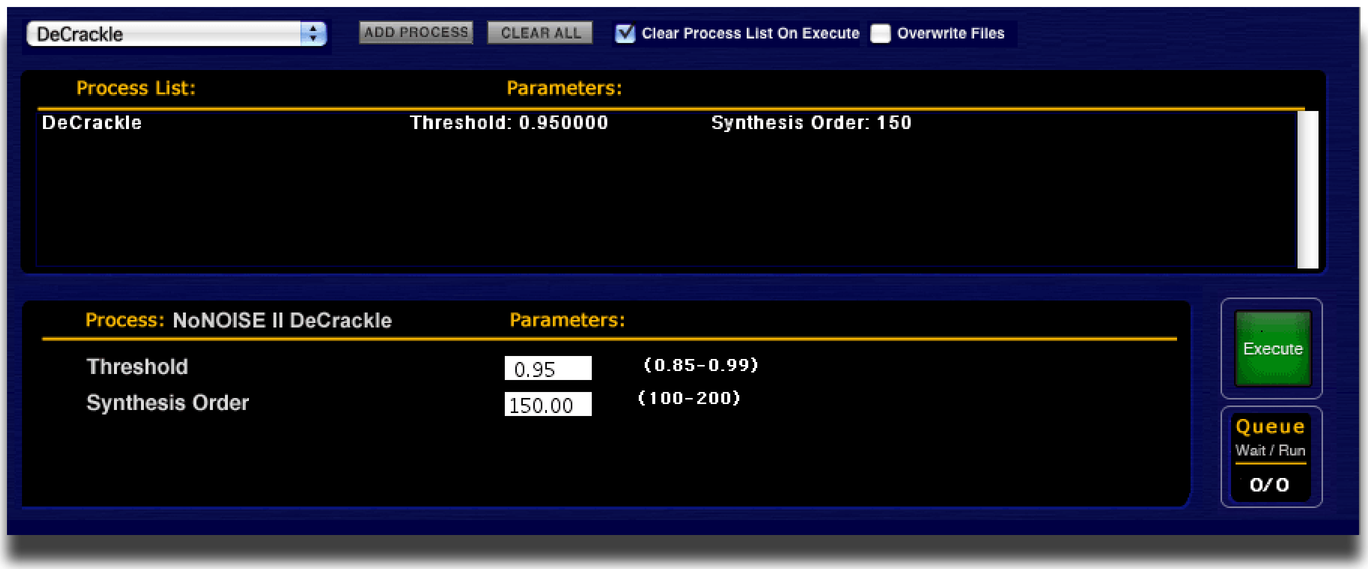


Figure 10: DeCrackle Parameters area of the Parameters Window

A common type of impulse noise is crackle, in which small impulses crowd against one another, producing a nearly continuous noise, like bacon frying in a pan. This type of artifact requires a different processing approach.

The Decrackle process in Sonic Studio Process is effective for correcting this type of dense, impulsive noise by performing a type of sliding interpolation that isolates good audio between impulses and uses the good audio as the basis for resynthesis.

Note that DeCrackle is not available for Amarra users as it requires our Manual DeClick option to be installed on both your iLok and Mac.

DeCrackle is targeted at our professional soundBlade users.

1.1.3.1 Threshold (Click Fraction)

This number affects the detection part of the Decrackle algorithm, in which audio samples are separated into good and bad categories. The synthesis process then replaces the samples that are deemed bad with synthesized material that matches the surrounding sound.

The Threshold is the percentage, or fraction of the samples that will be left in the good category. The range of the parameter is from 0.85 to 0.99. The higher the Threshold, the less aggressive the decrackling.

For heavier decrackling, make the Threshold smaller, as low as 0.85. Any lower is not recommended, as there will be too little of the original signal to provide a basis for resynthesis. Lowering the Threshold by only 0.05 is enough to make a noticeable difference.

HINT: If the Threshold is set too high then there will be some amount of crackle remaining in the program. If the Threshold is set too low you may begin to decrackle good audio, which could result in low frequency artifacts (audible thumps).

1.1.3.2 Synthesis Order

The Synthesis Order determines the precision of the Decrackler's resynthesis. In general, the larger this number, the cleaner and more artifact-free the output. The default value of 150 is suitable for the majority of source materials.

If low frequency thunks occur in the processed output, try raising the value of the Synthesis Order to 175 or even 200. However, raising the Synthesis Order markedly increases the amount of time required to process the soundfile. Synthesis Order should generally be left at 150 unless the process is producing unacceptable numbers of artifacts.

Paradoxically, processing artifacts tend to occur more with clean recordings that have high signal-to-noise ratios. For (relatively) recent material, such as tape recordings from the early 1950s, it is common to set the Synthesis Order to 128. For 78s from the 1930s, however, it is common to set the Synthesis Order to 125.

1.2 The Queue Manager tab

To monitor and manage queued processes, the Queue Manager tab offers status information and limited options to influence the queued jobs and tasks being performed.

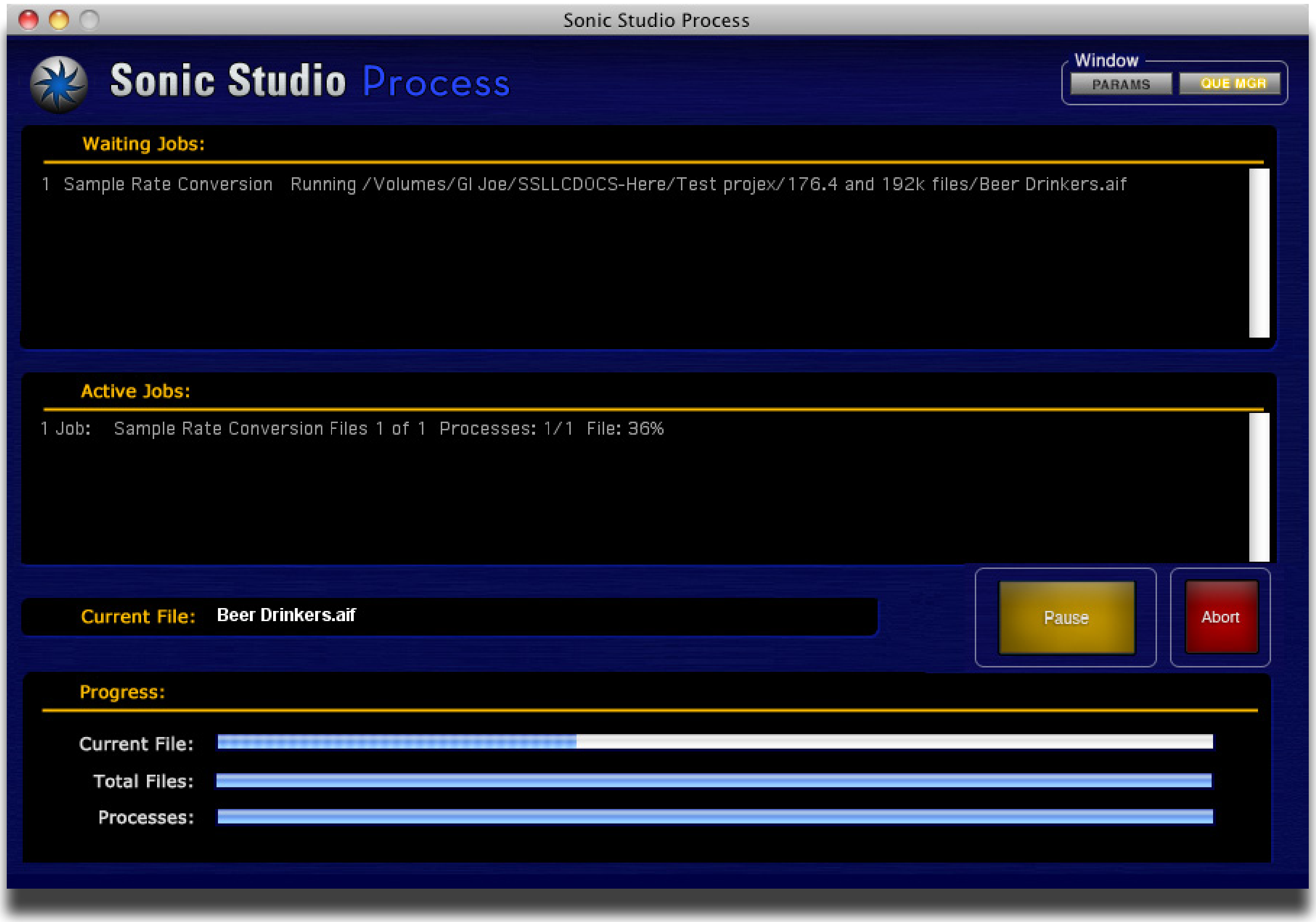


Figure 7.2: The Queue Manager tab

In the top half of the window, all processing jobs in the Queue are listed.

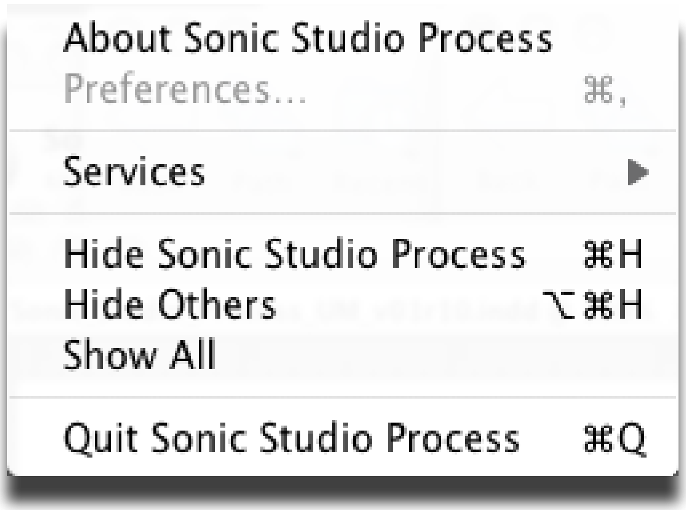
In the bottom half, the active jobs are listed. The progress bars below this area show which file is being processes and which process of the total number of processes to be applied is currently being applied.

To suspend currently running processes, click the "Pause/Resume" button. The current process is halted and queued processes are held in the queue. To resume processing, click the Pause/Resume button again.

To abort the current process, click the Abort button and the current process will be aborted, initiating the next process in the Queue.

1.3 Menus

1.3.1 Sonic Studio Process Menu



1.3.1.1 About Sonic Studio Process

Opens a dialog box describing the version and build numbers of your Sonic Studio Process application. Clicking on the dialog box closes it.

1.3.1.2 Preferences..

Disabled for Sonic Studio Process.

1.3.1.3 Hide Sonic Studio Process

Use the Hide Sonic Studio Process menu item to hide Sonic Studio Process and all of its open windows, allowing you access to other programs running in Mac OS X. Clicking on the Sonic Studio Process icon in your Dock returns Sonic Studio Process to view.

1.3.1.4 Hide Others

Use the Hide Others menu item to hide all visible applications except Sonic Studio Process allowing you to focus on Sonic Studio Process alone. Clicking on any icon in the doc will return that application to view.

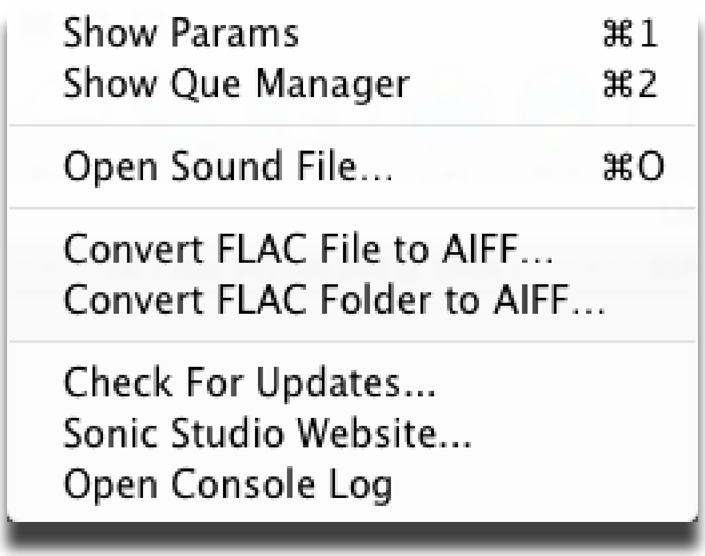
1.3.1.5 Show All

The Show All menu item un-hides all running programs in Mac OS X.

1.3.1.6 Quit Sonic Studio Process

Use the Quit Sonic Studio Process menu item to quit Sonic Studio Process and close all open windows.

1.3.2 File Menu



1.3.2.1 Show Params

Opens the Sonic Studio Process Parameters window.

1.3.2.1 Show Que Manager

Opens the Sonic Studio Process Que Manager window.

1.3.2.1 Open Sound File...

This command opens a Mac OS file browser, allowing you to select any sound file recognized by Sonic Studio Process. This includes AIFF, WAV, AIFC 32 bit floating point files and BWF files along with SD2 or Sound Designer II files with region definitions. Sonic Studio Process is also able to open audio files by dragging and dropping the files into the Parameters window.

1.3.2.2 Convert FLAC File to AIFF

Navigate to FLAC files and convert them to AIFF format.

The shift and command keys allow you to select more than one file.

1.3.2.6 Convert FLAC Folder to AIFF

Navigate to a folder containing FLAC files and convert the folder’s contents to AIFF.

Specify the source folder and a destination folder for the AIFF files.

1.3.2.7 Check For Updates

Check for updates to Sonic Studio Process.

1.3.2.8 Sonic Studio Website...

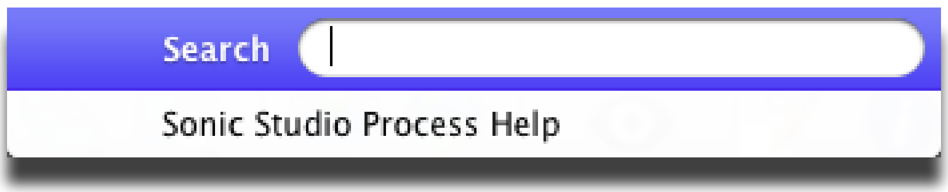
Opens www.sonicstudio.com into a new web browser window.

1.3.2.9 Open Console Log

Allows Sonic Studio Process to log information on its operation to the Console Log. Use the Console Application to view this log (in Utilites Folder).

1.3.3 Help Menu

Allows access to Sonic Studio Process Help information.



2.0................. Sample Rate Conversion Technical Details

All rate conversions rely on a relatively simple process, in which the source data is filtered with a low pass filter (lpf) with bandwidth determined nominally by the Nyquist frequency of the lower of the two frequencies (source frequency, destination frequency).

For noninteger conversions, the filter is oversampled, usually at a high rate, to produce a set of coefficients which are then used to filter the data, but only at the output rate determined by the destination frequency. The upconversion and downconversion factors are defined by the equation:

$$T = (M/L)T'$$

where the integers M and L provide the smallest whole integers relating the new (T') and old (T) sample periods.The oversampling, or upconversion, rate is M/T.

The ratios 96:44.1 and 44.1:96 require high oversampling (M=147 and M=320, respectively) and hence substantially more complicated implementation than the simple decimation/interpolation ratios 88.2:44.1 and 44.1:88.2 (M=1 and M=2, respectively). However, the conversions are based on the same low pass filters, and as long as the same filter design approach is used and high precision is maintained in the computations, the sonic performance should be comparable.

LEARN:

The coefficient technique for Classic monolithic SRC is called windowed sinc, meaning that, during calculation, filters are used to separate one band of frequencies from another.

The coefficient technique for Sonic HD multi-stage SRC is optimized Mean-Squared Error. Mean Squared Error is a way to quantify the difference between values implied by an estimator and the true values of the quantity being estimated. In this case, multi-stage means that the re-sampling can be done in incremental stages. For example, a conversion from 44.1 to 48 is broken down into two stages of 8/7 and 20/21, so that 147 samples in yield 160 samples out (multiply by 3 to get 441 in -> 480 out). There are also stages for 2:1 and 1:2 conversion, for doubling and halving the rate.

2.1 Sonic Issues To Consider

2.1.1 Data Precision

Filter coefficients in Sonic Studio Process algorithms are calculated using advanced arithmetic, converted integers, and highly-efficient DSP techniques. The audio data (internal word length 24 bits) is convolved with the filter coefficients using a double precision multiply algorithm. The output is rounded (not dithered) to 24-bit integer under the assumption that the roundoff error spectrum is below audibility and that the output will be dithered to the final bit depth. This implementation preserves 24-bit precision in the output data, even with very long filters.

2.1.2 Low pass Filter Parameters

On mathematical grounds, low pass filters requiring steep rolloff in order to attenuate aliases or images just outside the audio band will be fairly long. Depending on the design, a strictly Nyquist filter for 96 kHz:44.1 kHz downconversion having flat inband gain to 20 kHz and stopband attenuation at 22.05 kHz of -120 to -145 dB, has about 220-300 coefficients (or equivalently a length of 2.3 ms - 3.1 ms in time).

Apart from ease of implementation, the advantage of simple decimation ratios, if these are the only ratios of concern, is that optimal filter design approaches such as the Remez discrete Chebyschev approximation can be used to compute the filter.

Other things being equal, the longer the filter, the greater the potential for smearing of the natural time relationships in the audio source. The filter can be shortened by relaxing the steepness of the rolloff, but at the expense either of reducing gain at the upper edge of the audio band or by allowing aliasing.

From a sonic standpoint, aliasing, reduced gain at 20 kHz, and increased inband/stopband ripple all create audible distortions which, at significant levels, may be greater factors than the less objectionable effects of filter length. On the other hand, small amounts of certain distortions may be tolerated with various audio materials in order to minimize the effects of filter length.

In many cases, the shorter filters offer clear benefits and should be judged by audition. The rationales for the levels and choices are discussed in a preprint published by the 105th AES Convention, San Francisco, Sept, 1998 ("Multiphase Filters for Sample Rate Conversion of High Resolution Audio").

2.2.3 Classes of Rate Conversion

Sonic Studio Process performs two classes of upward or downward rate conversion: integer ratio and fractional ratio.

2.2.3.1 Integer Ratio Conversion

In integer ratio conversion, the source and output sample rates bear a simple, integer relationship to one another (for example, 44.1 kHz to 88.2 kHz is 1:2). Integer ratio conversions are the most straightforward.

Integer ratio down conversion is achieved by omitting every other sample, in a process known as decimation, and is illustrated in Figure 1. Here, curve A shows the original analog waveform with its higher-sample-rate digital representation superimposed. Curve B is the down conversion result.

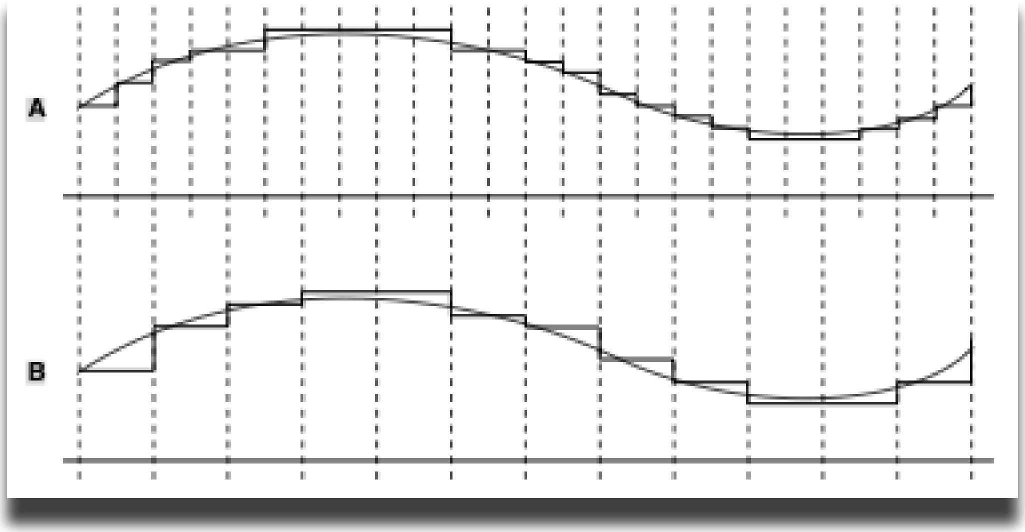


Figure 2-01: Integer Ratio Down Conversion

Because decimation causes aliasing (folding of frequencies above the Nyquist frequency back into the audible band), down conversion requires a steep lowpass filter which cuts off at the Nyquist frequency of the new, lower sample rate. (The Nyquist frequency is one-half the sample rate; it determines the usable bandwidth of the system.) For computational efficiency, the decimator and lowpass filter are normally combined in a single processing step.

For upward sample rate conversion, the converter needs to create new sample values that are evenly spaced between the source samples. This process is called interpolation, and is illustrated in Figure 2-03.

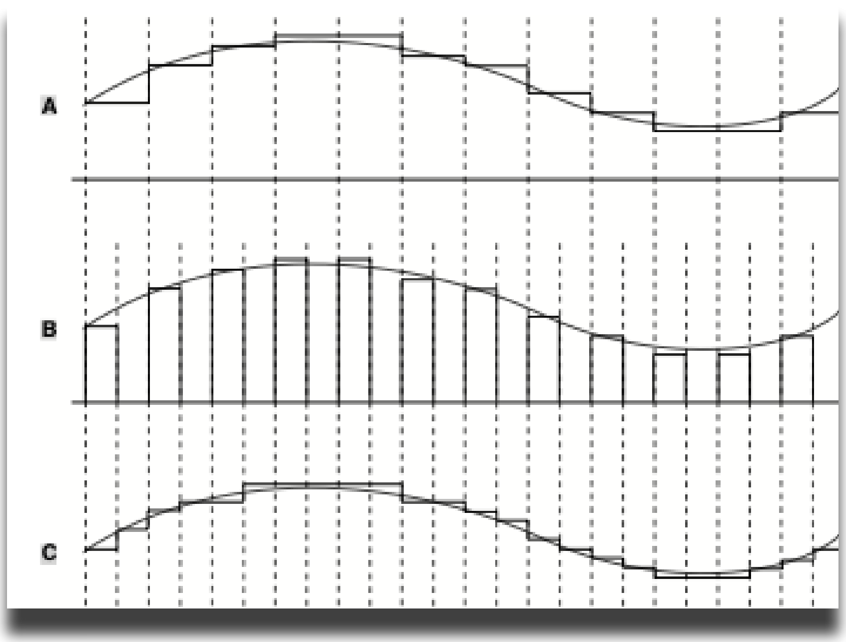


Figure 2-03: Integer Ratio Up Conversion
You can think of up conversion as occurring in two steps:

- First, zero-value samples are added between input samples at the new, higher sample rate (Figure 2-03 Curve B).

• Then, the signal is filtered using a steep lowpass which cuts off at the Nyquist frequency of the old, lower sample rate. (Figure 2-03 Curve C).

The up conversion filter is called an interpolator. Besides providing correct values for the new samples, it also prevents images of the input spectrum from appearing in the new, wider bandwidth.

2.2.3.2 Fractional Ratio Conversion

Fractional rate conversion takes advantage of the fact that, where the source and output sample rates are not related by a simple integer, there is nonetheless a higher clock rate to which both will have an integer relationship. This is illustrated in Figure 2-03.

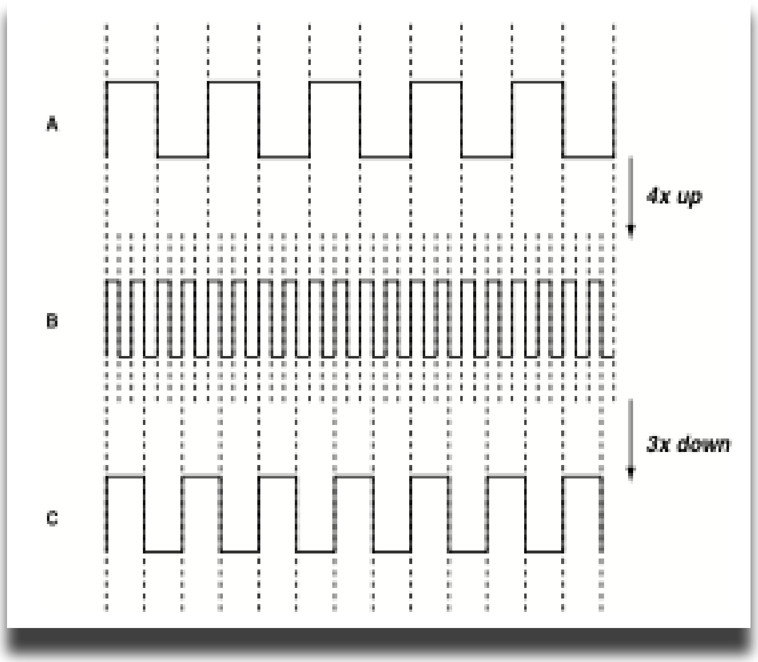


Figure 2-03: Fractional Ratio Conversion

Here, the ratio between clock rate A and intermediate rate B is 1:4. In turn, the ratio between clock rate B and rate C is 3:1. Oversampling at clock rate B therefore enables conversion between signals sampled at rate A and those sampled at rate C.

In practice, fractional-rate conversion is performed by an FIR lowpass filter which cuts off at the Nyquist frequency of the lower of the two sample rates (source or output) and is oversampled at some high frequency to which both sample rates have an integer relationship. The Nyquist filter prevents aliases (down conversion) or images (up conversion).

Sonic Studio Process User Manual Index