

SoundHack 0.60

©1992 Tom Erbe
Center for Contemporary Music
Mills College
Oakland, CA 94613
tom@mills.edu

I am writing this program to perform various soundfile manipulations on the Macintosh that have previously not been available. At the present, I have implemented soundfile type conversion, soundfile convolution, the phase vocoder, a binaural filter and an amplitude analysis and gain change module. In the near future I hope to add a spectral dynamics processor, and Larry Polansky's mutation synthesis. SoundHack can now read and write the following formats: Sound Designer II, Audio IFF, IRCAM, DSP Designer and NeXT .snd (or Sun .au). It can read (but not write) raw data files. It can read and write 8-bit μ Law, 8-bit linear, 32-bit floating point and 16-bit linear data encoding.

This program only works on Mac II and above machines! That is: you need a 68020 + 68881, a 68030 + 68882 or a 68040.

Thanks to Dr. Durand Begault of NASA-Ames for letting me use his binaural filter coefficients, Dan Ellis of MIT Media Lab for helping me with the Csound analysis feature and also to the Center for Contemporary Music for sponsoring my activities.

Please write me if you have any problems or suggestions!

Tom Erbe

Menus:

File

Open... Opens soundfiles with the following filetypes: Sd2f, IRCM, AIFF and DSPs.

Open Any... Opens any file as a soundfile. Useful for opening soundfiles that were created on other machines and have no filetype, or for opening headerless soundfiles.

Close Obvious.

Save A Copy... Saves a copy of the opened soundfile in any soundfile format.

Quit Obvious.

Edit The usual stuff, not too useful in this program though.

Hack Most of the things in this menu involve a lot of calculations, and take time! SoundHack pretty much takes over your Mac to do these, so if you think your Mac has frozen up, it probably hasn't. The Phase Vocoder is the worst of these processes in terms of taking over your Mac. So have patience.

Binaural Filter... This process allows you process a monaural soundfile (creating a stereo file) so that the result puts the mono signal at a position around your head. Either enter the desired position in the "Angle" dialog box (in degrees) or click the appropriate radio button. This processing module has filter data for 12 positions around the head, but if you enter an angle between two positions you will get a filter which is the mix of the two filters around it. Check the "Normalize" box if you want the output to be normalized after computation.

Convolution... This process takes two soundfiles, one is the open file and the other is used as a filter (impulse response). It then multiplies the spectrum of the two files together, producing a new soundfile. The effect is similar to a vocoder, but with many more filter bands. This process is very slow and takes a lot of memory, but the results can be very beautiful. If you want to use large impulse responses, you will have to quit SoundHack and reset the application memory size. The "Filter Gain" button should usually be set to "High", but if the impulse and input have similar spectrum, it should be set to "Med" or "Low" to avoid a clipped output. "Moving" is a new feature that moves through the impulse file, taking a new impulse after every block of processing. It is set so that the impulse file reaches the end at the same time that the input file reaches the end. Check the "Normalize" box if you want the output to be normalized after computation.

Gain Change... The equivalent functions on Sound Designer II, Audiomedica and Alchemy are faster, so in most cases you will want to use those. However, my module will give you an RMS value, and will allow two different gain factors (for left and right), plus will work on floating point and μ law files. Click on "Analyze" and the peak, peak position (in samples) and rms values will be figured. The gain factors will be set to normalize both channels independently. "Change Gain" will create a new file adjusted by the gain factors previously set. If you are dealing with a monaural file, the channel 1 information is applicable.

Header Change... Allows you to change the sample rate, number or channels, and data format of the file open. If you open a headerless file, you should use this dialog to set things properly before saving a copy!

Phase Vocoder This process allows you to change pitch without changing length and to change length without changing pitch. It does this through phase interpolation. The setting for the number of "Bands" allows you to set the number of filter-oscillator pairs used (actually, an FFT-IFFT pair). If you set this to be a larger, you will get better frequency resolution, if you set it smaller, you will get better time resolution. The "Overlap" setting adjusts the size of the window (relative to the number of filter bands) for analysis and synthesis and thus, the sharpness of the filter. A large setting (4x) will give the sharpest filter. The sharper filter will differentiate better between frequencies which are between bands, but a sharper filter will respond to amplitude changes slower. Click the "Time" button if you want time scaling, "Pitch" for pitch scaling. Type the scale factor in the "Scale" box. Click on the word "Scale" if you want to specify time scaling by the length desired, or pitch scaling by equal tempered semitones. Click the "Analyze Only" box to produce Csound compatible pvoc analysis files.

Control

Show Output This will show your sound whenever SoundHack writes sound to your output soundfile (except during file copying and normalization).

Pause Process This allows you to pause any long processing job in case you need to use your Mac, but you don't want to start all over with your processing. If you are running a convolution, it sometimes takes a while to pause (up to 3 minutes...).

Continue Process This will pick up your processing where you left off.

Stop Process This will kill your process and close the output soundfile.

Bug fixes and revisions:

- .60 Added the phase vocoder, phase vocoder csound analysis and show output. Fixed problem with normalization after moving convolution.
- .59 Added moving convolution.
- .58 Fixed problems with 8-bit AIFF files.
- .57 AIFF files used to read everything from SSND chunk to EOF, now only SSND chunk is read. The Binaural processor often destroyed AIFF headers, I think I have this fixed. DSP Designer files are now written as well as read (though I have no way of checking, please DSP Designer users, give me feedback).
- .56 Filter files are now closed properly after convolution.
- .55 Normalization feature disabled Binaural filtering, now fixed. Filter sensitivity added to Convolve.
- .54 Normalize after processing feature added. Dialogs adjusted for small monitors.