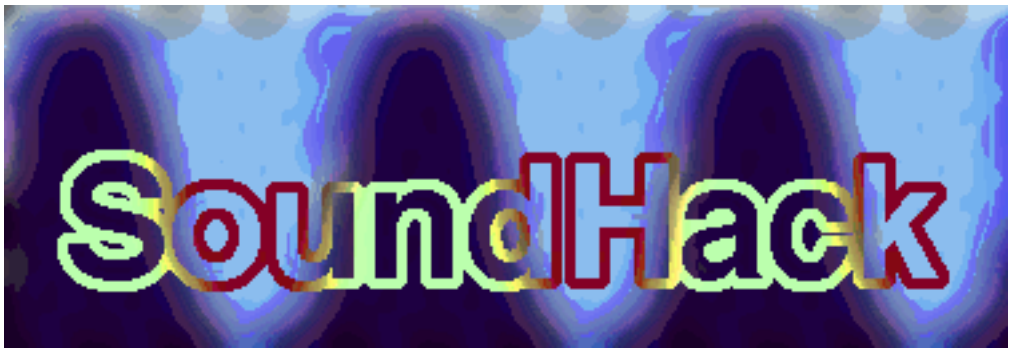


user's manual

version 0.888

Tom Erbe

School of Music
CalArts

INTRODUCTION

SoundHack is a soundfile processing program for the Macintosh. It performs many utility and esoteric sound processing functions available nowhere else. These functions make SoundHack invaluable to computer musicians, sound effects designers, multimedia artists, webmasters and anyone else who enjoys working with sound.

Sound Processing

- Time stretching or pitch shifting with the phase vocoder or varispeed.
- Spatialize with the binaural filter.
- Cross-synthesis between two soundfiles with soundfile convolution, ring modulation or spectral mutation.
- Noise reduction, spectral expansion or compression with the spectral dynamics processor.
- Separate transient and steady-state components with the spectral extractor.
- Use your own spectral algorithm with spectral analysis/resynthesis and the spectral assistant example code.
- Save sonograms of your soundfile in QuickTime™ movies or convert QuickTime™ movies into sound by sonographic analysis.

Utility Functions

- Play almost any type of soundfile (including AU, AIFF and WAVE).
- Record any size soundfile from the Macintosh sound input.
- Import soundfiles from audio CDs.
- Convert between different types of soundfiles with optional gain scaling and sample rate conversion.
- Change values in the soundfile header (sample rate, number of channels, loop points and marker info).
- Read and write Sound Designer II, Audio IFC, Audio IFF, BICSF (IRCAM), DSP Designer, QuickTime/AIFF, Microsoft WAVE (RIFF), NeXT .snd, Sun .au, ULaw, IMA4, TEXT and headerless (raw) soundfiles.

The rest of this document is a small tutorial (in progress) then a menu by menu description of SoundHack. Please write me if you have any problems or suggestions!

Tom Erbe

CONTENTS

SYSTEM REQUIREMENTS

SoundHack requires System 7.0 to operate, Sound Manager 3.0 to playback soundfiles and Quicktime 2.0 to import CD audio tracks and read and write sonograms. It comes in 3 flavors to accommodate different hardware platforms.

SoundHack NF runs on all Macintoshes, albeit very slowly. This is the only version that will work on 68LC040 based Macintoshes.

SoundHack FPU runs on 680x0 Macintoshes with hardware floating point. The following 3 CPU/FPU configurations will work: 68020+68881, 68030+68882 and 68040. It will also run on FPUless Macintoshes (except 68LC040 machines) with John Neil's excellent **SoftwareFPU** floating point emulator. **SoftwareFPU** is available on most Macintosh software archives or from John Neil & Associates (johnneil@netcom.com).

SoundHack PPC runs on Power Macintoshes. It is from 10 to 40 times faster than SoundHackNF.

SHAREWARE INFO

SoundHack was once shareware/musicware, but no more. You may freely do whatever you want with it. If you still want to send me some of your art, music or favorite postcard my address is:

```
Tom Erbe
608 Carla Way
La Jolla, CA 92093
```

If you do not have internet access, you can buy it for $65 from my distributor.

```
Frog Peak Music
PO Box 1052
Lebanon, NH 03766
603-448-8837
Email: frogpeak@sover.net
URL: http://www.sover.net/~frogpeak
```

ON THE INTERNET

The current shareware version of SoundHack (SoundHack, SoundHackNF and documentation) is available at www.soundhack.com

There is a SoundHack mailing list. To join, send email to *listproc@music.calarts.edu* and include the text "*subscribe freesound Your Full Name*" in the message body.

There is also a MP3 station for SoundHack users. Go to *http://shoko.calarts.edu/~tre/freesound.pl* for information on listening or submitting work.

RELEASE 0.880: New Features

* The **QT Coder,** a sound to QuickTime Movie to sound converter. Put your sounds through After Effects, synthesize a graphic score. Turn your home movie into a rhythm track.
* The **varispeed** function now maps correctly to the output file time.
* **Playback** enhancements: **looping playback** with control-space and a playback after processing option is added to the "Preferences" dialog.
* **Loop & Marker** features: WAVE loop and markers are now read and written, the loop and marker dialog now uses sample numbers instead of time.
* The fast motorola PPC math library is built into this version.
* New read and write types:

SDII	24 & 32 bit linear
Sun/Next	aLaw
AIFF-C	aLaw, uLaw, 32-bit floating point, MACE3, MACE6 and IMA4

* New playback types:
  aLaw and 32-bit floating point
* **DrawFunction** panel enhancements: a random function, a mean value can be set and the function can be normalized.
* Increased sample rate conversion quality
* An even stranger auto naming function.
* A new splash screen with flashy messages.

Bug Fixes

+ AIFF uses the common chunk order
+ AIFF uses correct blocking and chunk boundaries
+ AIFF, AIFF-C and WAVE no longer overrun the end of the sound chunk (no end of file click)
+ Large Kaiser windows don't crash program.
+ Playback returns to beginning after reaching end.
+ Very short soundfiles playback OK.
+ Lack of QuickTime doesn't crash PPC version.
+ Better memory conservation.
+ QuickTime files now contain proper Moov resource.

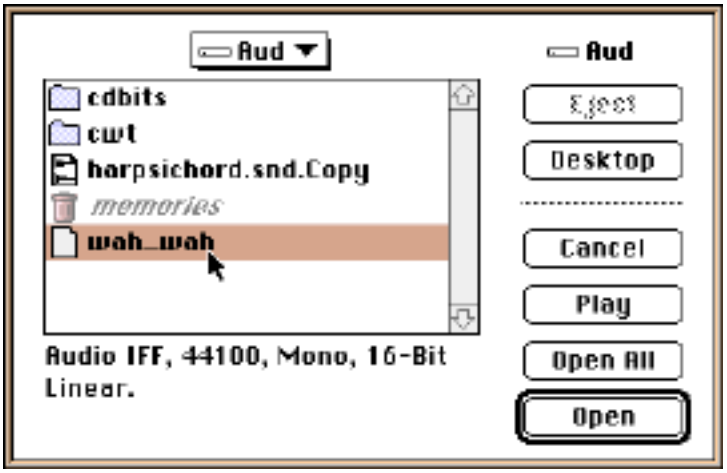
TUTORIAL: 7 points of departure

In this section I will take you step by step through some common sound processing operations. For all of these you will need a source soundfile to process and enough disk space to hold the result.

1. Spatialization

SoundHack provides spatialization of monaural soundfiles with its binaural filter. This filter, known as the head-related transfer function (HRTF) emulates the filtering action of your head and outer ear for any position around your

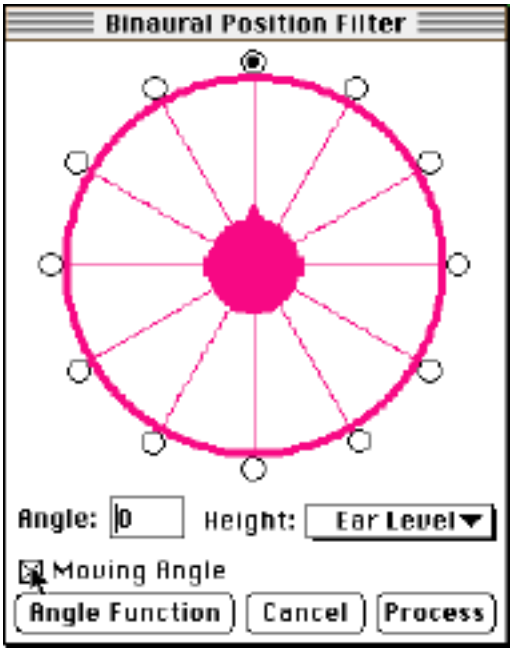
head. The following example will spin a sound twice around your head.



Type **command - O** to open the soundfile to be processed. This should be a monaural file with a sample rate of 44100 as the filters are tuned for this sample rate. The open dialog box will show both the sample rate and number of channels for the selected file.

Note that some sounds spatialize better than others. One of the main cues for spatialization is the inter-aural time difference or ITD. This is the delay from the moment the sound is heard at the near ear to the time the sound is heard at the far ear. Because the ITD is so important, sounds with little temporal variation will not spatialize well. For example, a snare drum is much easier to locate than a constant flute tone. In addition to the ITD, keep in mind that most of the spectral differences between the near and far ear are above 1500 Hz, as frequencies below this point will diffract around your head. Sounds with little high frequency energy will therefore spatialize poorly.

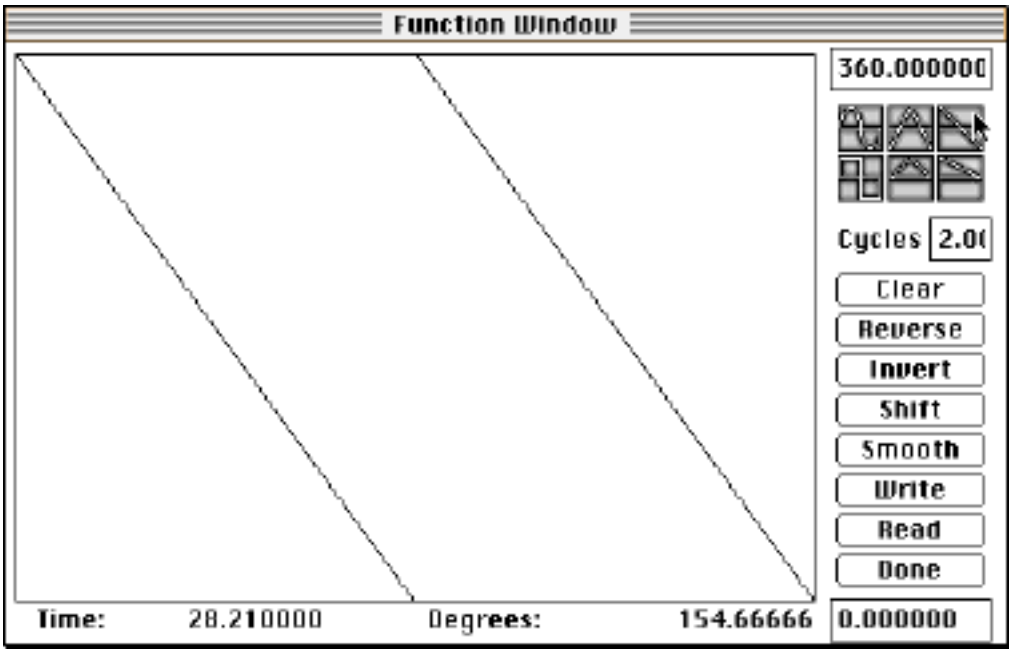
Once the soundfile is open, type **command - B** to bring up the binaural filter dialog. Here you can set the various parameters for a binaural spatialization. For instance, you could set a specific azimuth angle by typing in the **Angle:** box or by



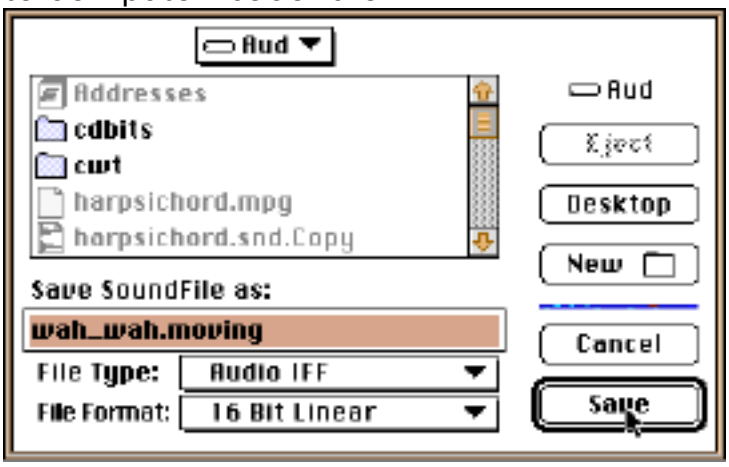
clicking on one of the radio buttons. You can set the height to be above, below or at ear level (though the detection of height is usually dependent on head movement, so it is not well simulated with a static filter).

To spin the sound around your head, click the box called **Moving Angle.** When you do this, the **Angle Function** button will appear. Click on this button to bring up the **Draw Function Window**.

This window allows you to draw a curve to control the angle of the sound as it moves around your head. You will see two legends on the bottom of this window: **Time:**, which indicates the current time in the input soundfile, and **Degrees:**, which is the azimuth (0 degrees is straight ahead). You will now use one of the presets to smoothly spin a sound twice around your head. First, type "2.0" in the **Cycles** box,

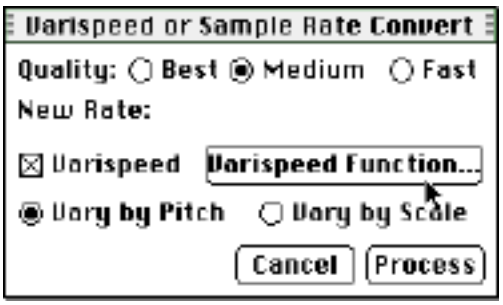


then click on the ramp function icon (see the mouse arrow in the picture to the left). This creates two cycles of a ramp function, which will control the azimuth angle during processing. Now click the **Done** button in this dialog (the **Draw Function Window** will disappear), and then click the **Process** button in the binaural function dialog. The "Save Soundfile as: " dialog will now appear, which has options to set the soundfile type and format. You will probably just want to set the soundfile type to **Audio IFF** and **16 Bit Linear**, as this is the format most commonly used by sound editors. After you click **Save**, SoundHack will start processing. All you need to do now is wait. On a 33MHz 68040, the compute time to real time ratio is 55 to 1 (that is, it takes 55 seconds to compute 1 second of



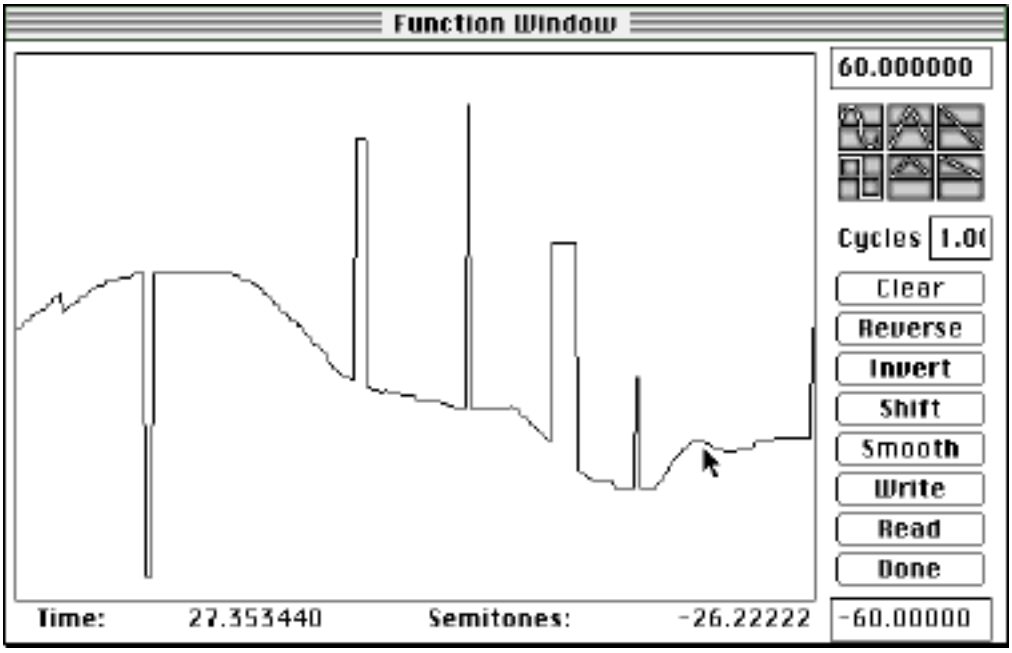
sound). While processing, the status/playback window (shown below) will show the current positions in the input and output soundfiles as well as a progress bar. At any time, you can pause the processing by typing **command - ,** and listen to the output soundfile by pressing the spacebar. After it is finished playing, another **command - ,** will start the processing again.

2. Time distortion nr. 1



One of the main features of SoundHack is the ability to distort the timebase and/or pitch of the soundfile. In this example, we will use a simple varispeed (variable sample rate conversion) to modulate both the pitch and timebase.

Open a soundfile as before (**command - O**).



Bring up the **Varispeed** dialog by typing **command - V**. To enable varispeed (rather than simple sample rate conversion), click on the **Varispeed** box. Now, bring up the **Draw Function Window** by clicking on the **Varispeed Function...** button (see mouse arrow in picture above). When the **Draw Function Window** appears, draw a varispeed function by click/dragging the mouse in the function drawing area (see mouse arrow at left). This varispeed, unlike a tape varispeed, is capable of a 12 octave range. That is equivalent to a tape machine that has speeds from 0.5 to 1,920 inches per second, so take advantage of this broad range. Click **Done** in the **Draw Function Window** and **Process** in the **Varispeed** dialog to start processing. On a 33MHz 68040, with the varispeed quality set to **Medium**, the average process time to real time ratio is 43 to 1. Slowing the soundfile takes more processing time, as more sound is created.

3. Pitch shifting

SoundHack provides pitch shifting without time scaling with a technique known as the phase vocoder. In this technique, the sound to be shifted is sent into a bank of band-pass filters which are evenly spaced from 0 Hz to half the sample rate. SoundHack measures the amplitude and phase for each frequency at the output of this filter bank,. These amplitudes, phases and frequencies are used to control a bank of oscillators. Pitch shifting simply involves multiplying each frequency by a factor.

In this example, you will shift a soundfile up one octave. You should open a soundfile as you did in the previous example (**command - O**). Pitch shifting is not limited by sample rate or number of channels; any type of soundfile will work. However, this effect sounds best with sounds that are not harmonically dense. If more than one partial appears in any band of the filter bank, inter modulation distortion will result in that band.

Once you have opened the soundfile to be processed, type **command - P** and the **Phase Vocoder** dialog will appear (see box at left). Here there are many options; most are related to particulars of the analysis filter bank, and most can remain at their default setting. The phase vocoder can be set for a one octave pitch shift with just a few steps. First, click on the **Pitch Scale** button. Second, click on the **Scaling:/Semitone Shift:** popup menu and select **Semitone Shift:**. Third, enter "12.0" in the box to the right of the **Semitone Shift:** popup menu. At this point you could click on **Process** and start processing, but there is another parameter you might want to adjust first. **Bands:** sets the number of bands in the filter bank as well as the number of oscillators for resynthesis. This will allow you to work with sounds of varying degrees of harmonic density. You might think that a large number of bands is always preferable, but more bands require tighter filters which in turn causes more phase distortion. So it is a trade-off. I have found 512, 1024 and 2048 to be the most useful settings. You should also try the **Scaling Function** option. This will bring up the **Draw Function Window** and allow you to have variable pitch shifting with a 12 octave range.

Pitch shifting is one of the slowest processes in SoundHack. With a 33MHz 68040, default settings, and a 44100 sample rate, monaural file as the source, the process time to real time ratio is 290 to 1.

4. Time distortion nr. 2

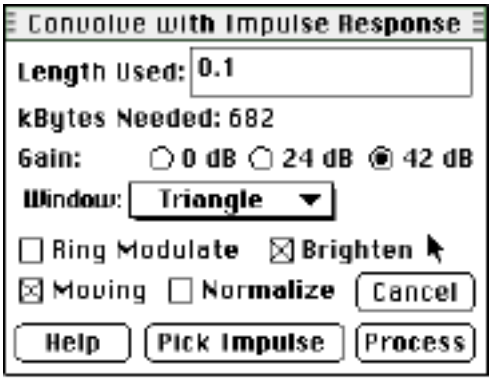
Just as the phase vocoder technique allows pitch shifting without time scaling, it also allows time scaling without pitch shifting. In this example, you will stretch a soundfile to twice its length.

Open a soundfile (**command - O**). As in the previous example (Pitch shifting), the results will be better for sounds which are not harmonically dense. Type **command - P** to bring up the **Phase Vocoder** dialog. This time, click the **Time Scale** button, and select the **Scaling:** popup menu. Type "2.0" in the box to the right of the **Scaling:** popup menu, and click **Process** to start processing. For an interesting special effect, set **Threshold Under Max. (dB):** to "-6" before processing. This will allow only a few loud partials to be resynthesized. Phase vocoder time scaling is relatively fast. With a 33MHz 68040, the process time to real time ratio is 69 to 1.

5. Cross-synthesis nr. 1

Cross-synthesis is the combining of two sounds to create a new sound. This is done by analyzing and extracting significant characteristics from the two source soundfiles, then combining these characteristics in the synthesis of the new soundfile. SoundHack has two functions which perform cross-synthesis, convolution and mutation. In this example you will try convolution.

Convolution takes two sounds, analyses them for spectral content, then multiplies the two spectra to create a new sound. This emphasizes frequencies which are held in common and greatly reduces frequencies which are not. For an interesting convolution it is good to start with sounds that have spectra which are neither too similar nor dissimilar.

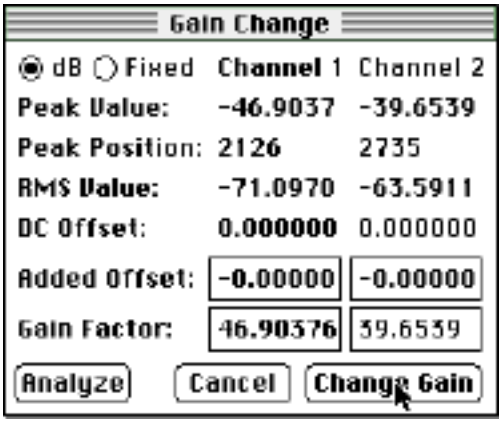


Open one of the two source soundfiles by typing **command - O**. Type **command - C** to bring up the **Convolve** dialog. Open the second of the source soundfiles by clicking on the **Pick Impulse** button. Click **Moving** to have convolution move through the second soundfile, otherwise this process will convolve against a fixed segment of the second soundfile. Click **Brighten** to avoid over-attenuation of high frequencies. Select **Triangle** in the **Window:** popup menu to smooth the convolution. If you prefer an unsmoothed convolution, select **Rectangle**.

Now set the **Length Used:** number. This is used to designate how much sound gets processed in each block. You can treat this number like the decay time of a reverb, because convolution has a sound which is like a very complex,

tuned reverb. A length of 0.1 seconds (100 milliseconds) is a good place to start. For higher settings of **Length Used:**, the memory requirement increases exponentially.

Now that everything is set, click **Process**. Because the gain in convolution is unpredictable, save the new soundfile in a format with a large dynamic range. When the **Save Soundfile as:** dialog appears, select a **File Type:** of **NeXT (.snd)** and a **File Format:** of **32 Bit Floating Point**. Click **Save** and wait.



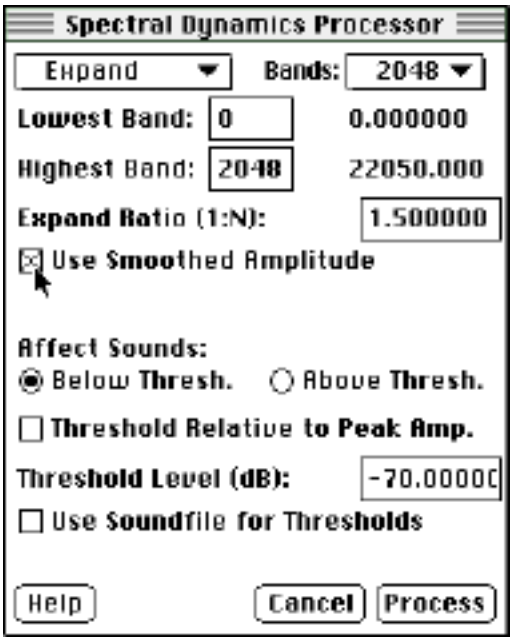
Once your convolution is done, you will need to convert the new soundfile from a floating point format to something usable (usually 16 bit linear). You will also need to adjust the optimum gain for this conversion. To do this, select the new soundfile, then type **command - G**. The **Gain Change** dialog will appear. Click **Analyze** to find the peak amplitude, then click **Change Gain** to create a new, normalized soundfile. In the **Save Soundfile as:** dialog, set the **File Format:** to **16 Bit Linear**.

Convolution is the fastest of SoundHack's functions. For a moving convolution, with a 33MHz 68040, and 44100, monaural soundfiles, the process time to real time ratio is 14 to 1.

6. Noise reduction

As you have seen in the previous examples, much of SoundHack's processing involves the spectral analysis of an input soundfile and subsequent resynthesis of an output soundfile. One very practical use of spectral analysis/resynthesis is noise reduction, where noise components are identified and reduced before resynthesis. A very simple scheme for identifying noise components is to assume that any spectral component with an amplitude below a certain threshold is noise. This works well when the noise is fairly constant, and not too loud (tape hiss, for instance).

Open the soundfile as usual (**command - O)**. Type **command - D** to bring up the **Spectral Dynamics** dialog. This is similar to a typical analog multiband dynamics processor except that here there are up to 4096 separate bands.

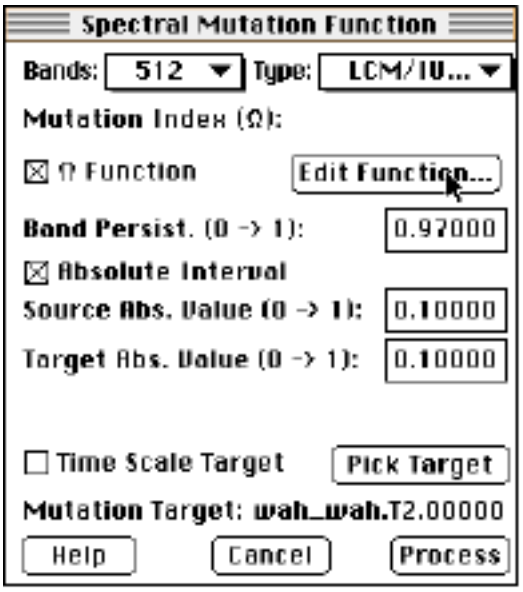


For noise reduction, set **Bands:** to **2048**. Set **Type:** to **Expand**. You could also use **Gate** for noise reduction, but expansion gives a gentler amplitude reduction and sounds more natural. The **Lowest Band:** and **Highest Band:** should be set to 0 and 2048 unless you want to limit the noise reduction to a particular frequency area (as you would in 60 Hz hum removal). Set the **Expand Ratio** anywhere from 1.5 to 3.0. A higher ratio will be more effective in reducing noise, but may cause artifacts. Set the **Threshold Level** anywhere from -50 to -90 dB, slightly above the hiss level. If you are too close to the hiss level, the dynamics processor will turn on and off too much and this will cause weird modulation noises. It is best to set the **Threshold Level** about 9 dB above the hiss level. Click the **Use Smoothed Amplitude** box. This feature is essential for good hiss removal as it will anticipate abrupt sounds by using an averaged amplitude for threshold detection.

Now click **Process** and let it go. After SoundHack creates a few seconds of sound, it is probably a good idea to pause processing and play the output soundfile. The parameters for noise reduction usually take a lot of experimentation.

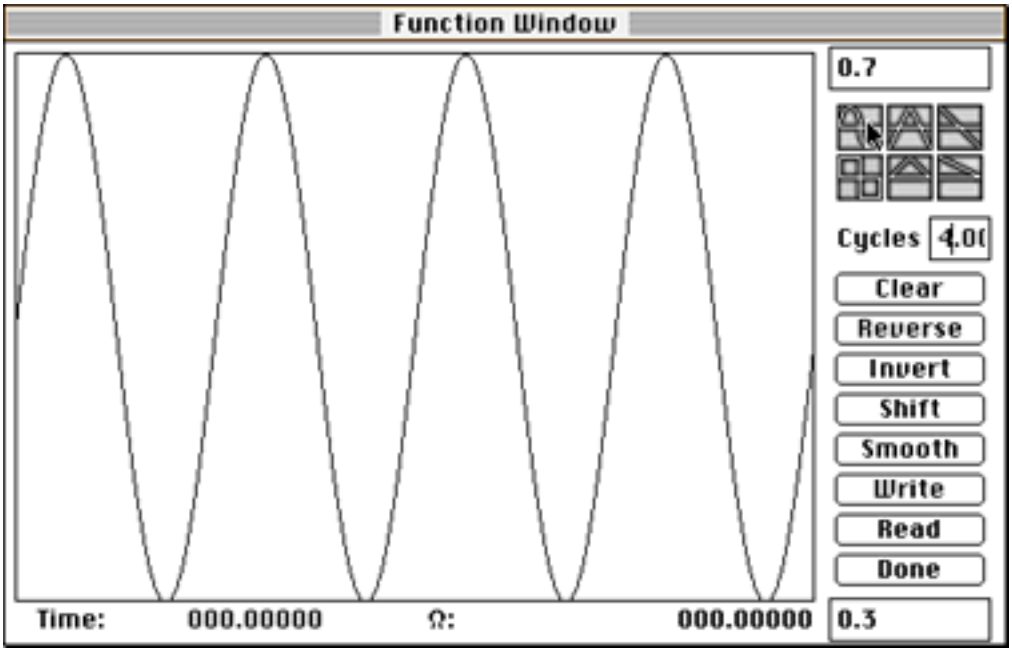
With a 33MHz 68040, and 44100, monaural soundfiles, the process time to real time ratio for spectral dynamics is 70 to 1.

7. Cross-synthesis nr. 2



In this final example, you will use the second form of cross-synthesis in SoundHack, spectral mutation. Mutation measures the spectral change over time in two soundfiles (called the source and target) and resynthesizes a new soundfile (the mutant) through various strategies of combination.

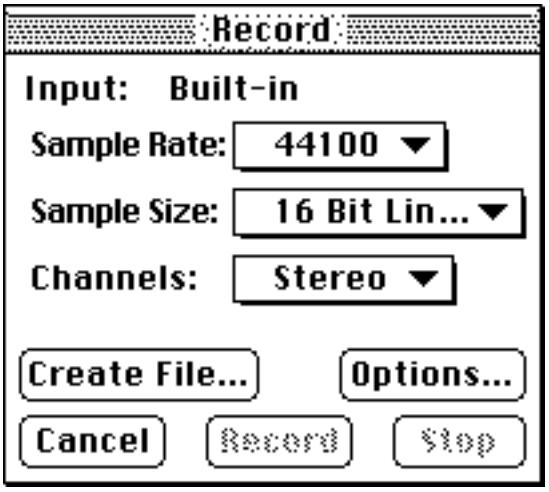
For mutation, you should choose soundfiles that have similar spectral characteristics. When the soundfiles are too different, the mutant seems to just fluctuate between the source and target sounds, without producing many interesting merged sounds. Open one of the soundfiles (the source) with **command-O**. Type **command - M** to bring up the **Spectral Mutation Function** dialog and open the second soundfile (the target) by clicking on the **Pick Target** button. Set the mutation **Type:** to **LCM/IUIM**. The differences between various mutation types are discussed later in this manual (page 21). Click the Ω **Function** box so that you can vary the mutation index. For the LCM/IUIM type, an index of 0.0 produces a mutant of all source and an index of 1.0 produces a mutant of all target. Now click **Edit Function...** to bring up the **Draw Function Window**.



To keep the mutation index near the center of its range, set the function limits to 0.7 and 0.3. Set the number of cycles to 4.0 and click the sine wave preset icon (see mouse arrow in picture). This will produce a mutant which oscillates slowly from almost the source to almost the target. Click **Done** in this window and **Process** in the **Spectral Mutation Function** dialog. Mutations are very hard to predict, so a lot of experimentation is required before you will get satisfying results. The main factor in the success or failure of a mutation is the choice of source and target soundfiles.

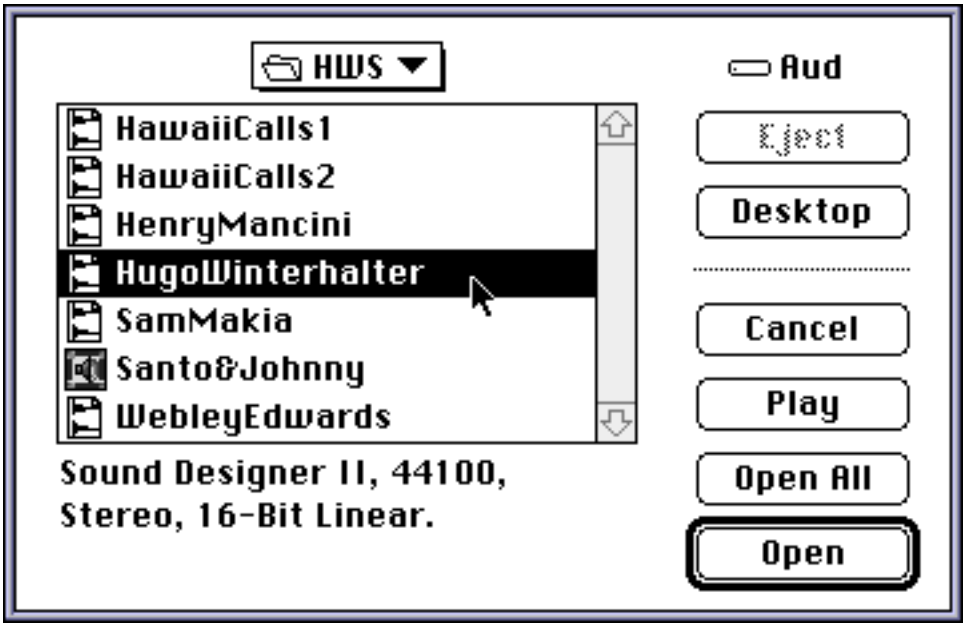
SOUNDHACK REFERENCE
FILE MENU

(Command - N) Record New...



If you have an input device, this will allow you to record an AIFF soundfile. You should first set your desired sample rate, sample size and number of channels, and then click **Create File...** to create the soundfile. Once you have initialized the soundfile the popup menus will no longer be accessible. **Options...** allows you to change settings peculiar to your input device. You will then be able to start recording by clicking **Record**. **Stop** will stop the recording, open the soundfile and add it to the SoundFile menu. Some slower hard disks may cause glitches while recording at the highest rate.

(Command - O) Open...



Clicking **Open All** will open all of the soundfiles in the current folder. Clicking

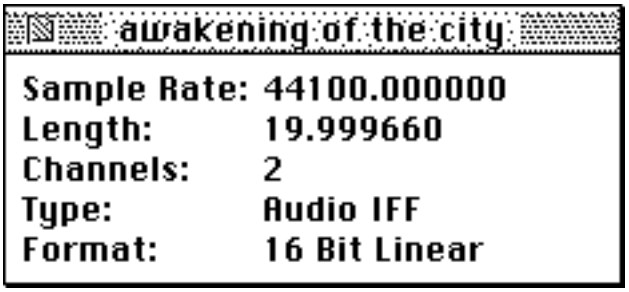
Play will play the soundfile.

Soundfiles with the following file types will appear in this dialog box:

```
AIFF        Audio Interchange File Format
AIFC        Audio Interchange File Format - Compressed
DSPs        DSP Designer
IRCM        BICSF/IRCAM
MSND        MacMix
NxTs        NeXT .snd/Sun .au
PICT        PICT
SCRN
CSCR        PICT Resource
MooV        QuickTime Movie
RIFF
.WAV        Microsoft WAVE
Sd2f        Sound Designer II
            Audiomedia
```

If the file doesn't have a Macintosh 4 character file type, it will still appear in the open file dialog box provided it has one of the following name extensions:

```
.aifc .aiff .au .irc .sf .snd .wav .WAV
```



Once the soundfile is open, it is added to the SoundFile menu and the soundfile information dialog box appears. This dialog box gives the name, sample rate, length in seconds, number of channels, type and numeric format of the soundfile. You are no longer restricted (as you were in earlier versions of SoundHack) to having only one soundfile open at a time. The file with the front-most dialog box (which I will refer to as the "selected soundfile") is used as the input soundfile to the processes under the Hack menu.

Files can be opened with the **open document Apple Event** ('ODOC') in SoundHack version 0.860 and later.

(Command - A) Open Any...

This does the same as above, but makes no attempt to read the soundfile header. This is useful when opening headerless, damaged and text soundfiles. Text soundfiles should be formatted so that each line is a fixed point sample. Here is an example of how the text should look:

```
-0.054688
```

```
-0.015625
-0.007812
0.015625
0.000000
-0.117188
```

Soundfiles opened with **Open Any...** must be saved to another format before being processed. You should set the proper sample rate and data format with the **Header Change...** dialog before making this conversion.

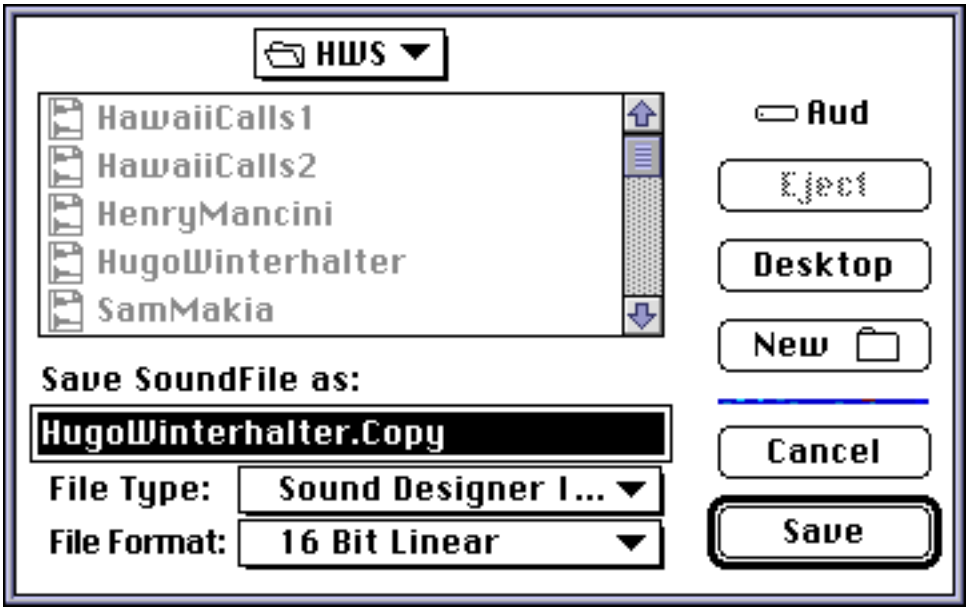
(Command - W) Close

Closes the selected soundfile.

Close & Edit

This is available if you have selected a soundfile editor in **Preferences....** It closes the selected soundfile, sets its file type to be the same as the selected editor and sends an AppleEvent to the Finder to open the document. Occasionally this does not work (the file just reopens in SoundHack), because the Finder does not always catch that you have changed the file type.

(Command - S) Save A Copy...



Saves a copy of the selected soundfile in any supported soundfile format. This is the main command for those of you who are just copying from one soundfile format to another.

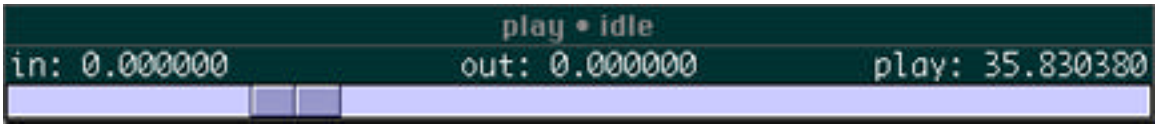
Split Into Mono Files...

This will divide a stereo or quad soundfile into multiple monaural soundfiles. This

will be useful for those users who create quad soundfiles with Csound or Cmix and need to get the files into a more portable format. This may also be useful to those who are using Pro-Tools, Deck or Sonic Solutions (which currently imports only monaural AIFF files). Soundfile splitting does not work with 24-bit soundfiles.

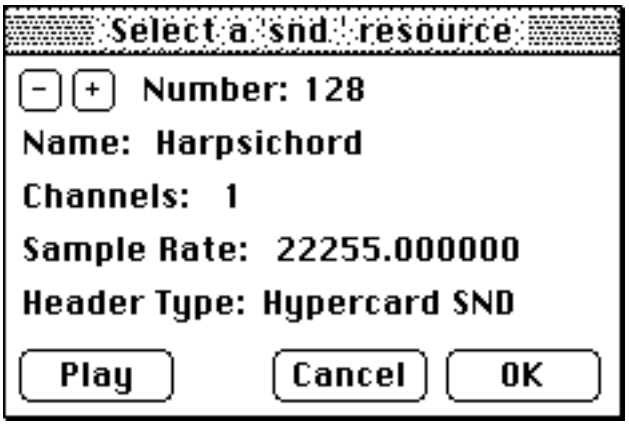
(Space) Play File

This will play the selected soundfile if it is 16-bit linear, 8-bit linear, unsigned, µlaw encoded, aLaw encoded, 32-bit floating point or ADPCM. You cannot play soundfiles while processing sound. However, playback will be enabled if you pause the process (**Command - ,**). You can start playback from any point in the file by using the slider in the process window (shown below). Pressing the **spacebar** alternately starts and stops playback. Pressing the **return** key stops playback and returns the slider to the beginning of the soundfile. **Control - Space** engages looping playback. SoundManager 3.0 or an AV machine is required for sound playback to work properly.



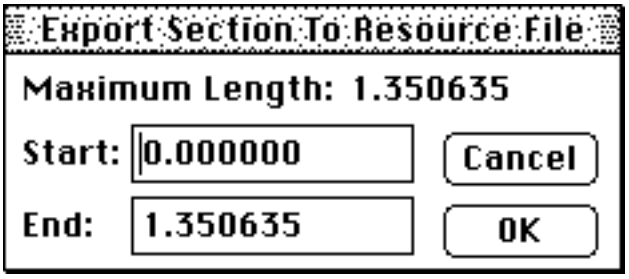
File playback can be triggered through the print document Apple Event ('PDOC') in SoundHack version 0.863 and later.

(Command - I) Import SND resource...



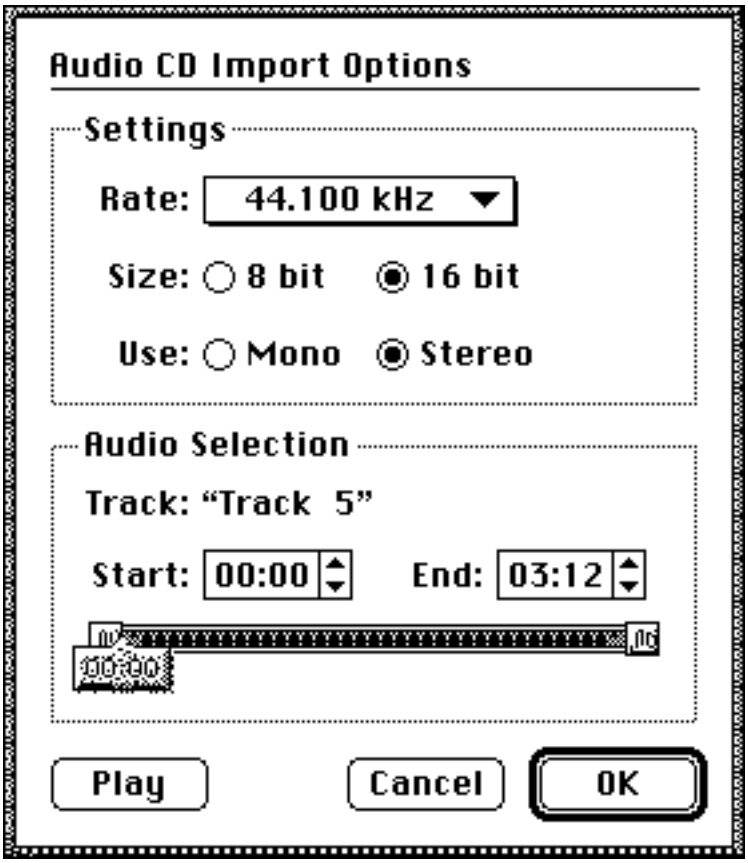
This will allow you to convert an Apple sound resource ('snd ') to a soundfile. This does not yet work with compressed 'snd ' resources or 16-bit resources.

(Command - E) Export SND resource...



This will allow you to write part of the selected soundfile into an Apple sound resource (also known as a double-click able soundfile). The length of the sound resource exported is limited to the amount of memory allocated to SoundHack. This will only make 8-bit
' snd' resources.

(Command - =) Import CD Track...



This allows you to import a CD audio track and save it to an AIFF soundfile. After selecting the track (make sure you know the one you want before hand!), the Quicktime **Audio CD Import Options** dialog box will appear. From this box you can set the sample rate, sample size, number of channels and the portion of sound to import for the AIFF soundfile. You can also preview the sound (though with only 8-bit quality). Importing sounds takes a long time, so be prepared to wait. Quicktime 2.0 and a CD-ROM drive is required for this feature. Importing may only work reliably on an Apple CDROM drive, though some have reported success with the HDT CDROM Tool kit and third party drives.

(Command - Q) Quit

This command allows you to do something else.

EDIT MENU

The required Macintosh functions. They are not too useful in this program though (as SoundHack is not an editor).

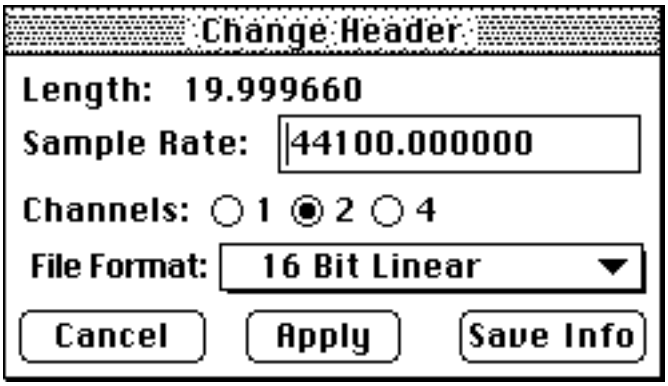
HACK MENU

This is where all the soundfile processing is. Most of the things in this menu

involve a lot of calculations, and take time! SoundHack takes over your Mac to do these, so if it appears that your Mac has frozen up, it probably hasn't. Spectral Mutation is the slowest of these functions, so have patience.

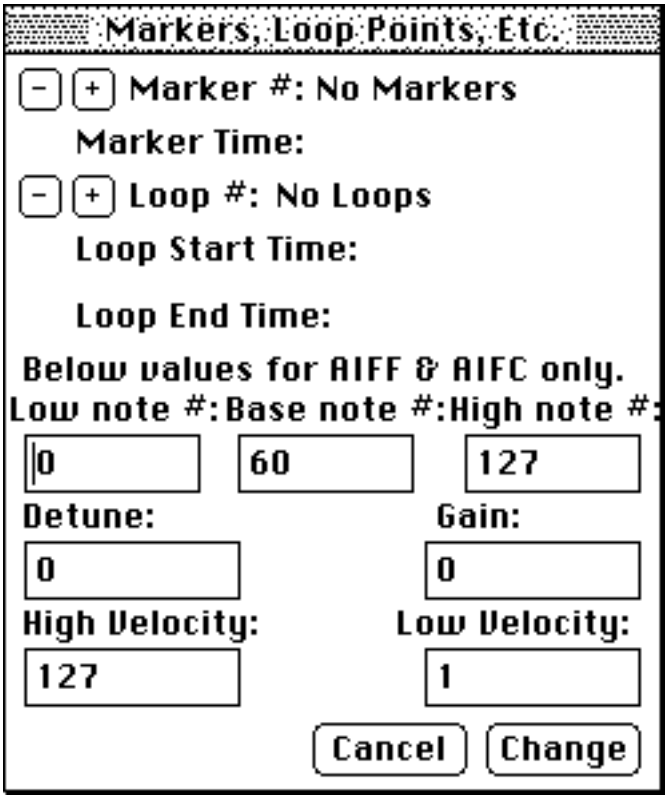
In most of these functions, sound is processed in blocks (a group of samples). After each block is processed you will see the soundfile time window update. You will be able to pause or stop a process between blocks, but not while a block is being processed. In the functions which process sound spectrally, each block is analyzed for amplitude and phase at a number of frequency bands. The bands are spaced evenly between 0Hz (DC) and the Nyquist frequency (half the sample rate). For example, if you choose 1024 bands and your sample rate is 44100, you will have a new frequency band every 21.53 Hz (44100 / (1024 x 2)).

(Command - H) Header Change...



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

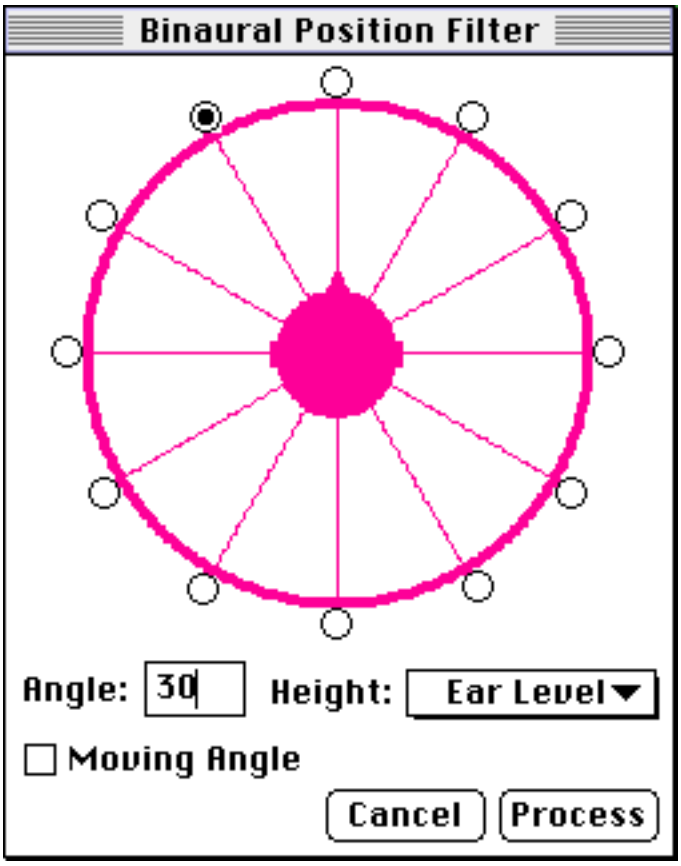
(Command - L) Loops & Markers...



This menu allows you to change loop pointers and marker locations for soundfiles in AIFF, AIFC, WAVE and Sound Designer II formats. It also allows you to change AIFF/AIFC specific information.

(Command - B) Binaural Filter...

This process allows you process a monaural soundfile (creating a stereo file). The result is a signal at a simulated position around the head. This is done by using a

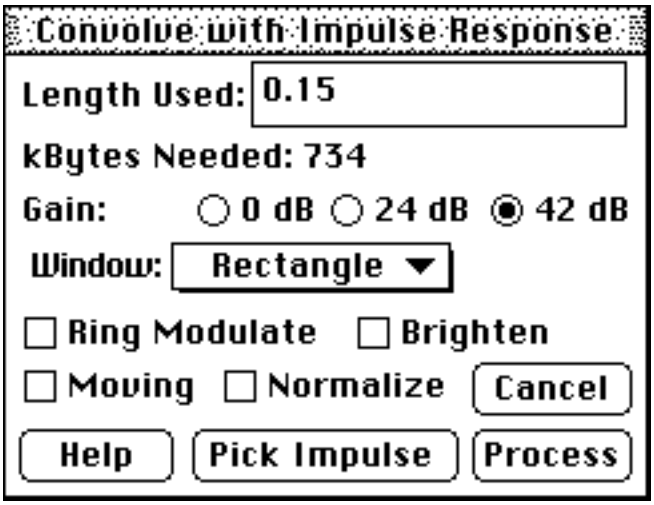


HRTF (head related transfer function) as a filter, with a separate function for each position around the head.

To use the binaural filter, enter the desired position in the **Angle** text box (in degrees) or click the appropriate radio button. This processing module has filter data for 12 positions. If you enter an angle between 2 positions you will get a filter which is the mix of the 2 filters around it. The **Moving Angle** box will allow you to do moving spatialization.

These HRTFs were obtained by Durand R. Begault, using methods described in "3-D Sound for Virtual Reality and Multimedia" (Academic Press: ISBN 0-12-084735-3).

(Command - C) Convolution...



This process takes 2 soundfiles: an input (the selected soundfile) and an impulse response file. It multiplies the spectra of the 2 files together, producing a new soundfile. The effect is a type of cross-synthesis, in which common frequencies are reinforced. In this implementation of convolution the sound is processed block by block, with each block as large as the impulse response. The **Length Used** window allows you to designate how much of the impulse response file to use. The **kilobytes Needed** number is an estimate of the application memory size that needs to be set for processing. If you want to use large impulse responses, that is, convolutions which cause this number to go over 1200 (the default size), you will have to quit SoundHack and reset the application memory size.

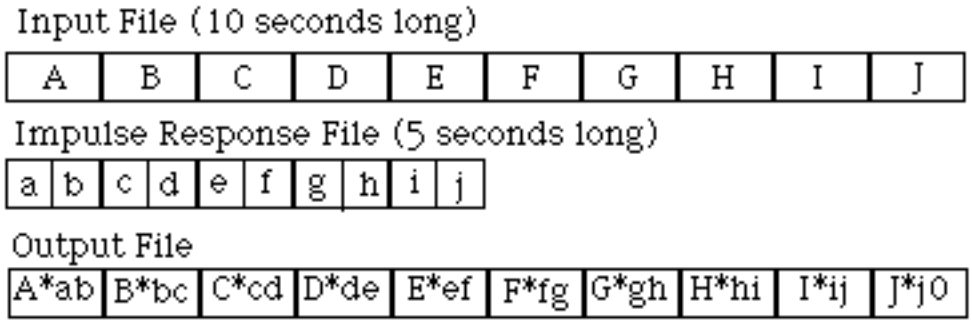
SoundHack attempts to automatically scale the amplitude of filter gain, but this value is impossible to predict. The **Gain** buttons gives you additional control over this. Set it to **42 dB** for most cases, but if the input and impulse response have similar spectra, extreme resonance will occur and the button should be set to **24 dB** or **0 dB**. However, the best way to deal with the uncertain gain problem in convolution is to avoid it altogether. Save the output in NeXT/Sun or BICSF floating point format, which has enormous dynamic range, then use the **Gain...** module to normalize it back to an integer format.

Checking the **Ring Modulate** box allows for ring modulation (or convolution in frequency) between 2 soundfiles. The **Brighten** box applies a simple +6dB per octave high-pass filter to the impulse. This is useful as most natural sounds have a roll-off from 6-12 dB per octave. Convolve two of these sounds and your result will have a roll-off from 12-24 dB per octave, much too dull. **Brighten** is a simple (probably too simple) fix for this.

Checking the **Window** popup will cause SoundHack to apply the selected envelope onto the impulse before convolution, resulting in a smoother convolution. A smoothing window is desirable when performing a moving impulse response convolution (described below), since the impulse response will be changing for each block of samples processed. This is also true for moving ring modulation. The triangular window is probably the best for smoothing, a rectangular window is the same as no window at all. Check the **Normalize** box if you want the output to be normalized after

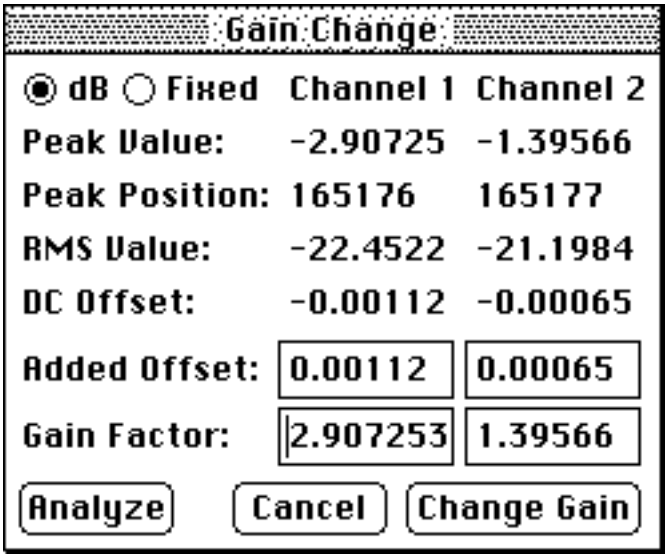
computation.

The **Moving** box allows you to perform a moving impulse response convolution. In this process a window moves through the impulse response file, selecting a new impulse response after every block of processing. The window size is set in the **Length Used** field. This window moves through the impulse response file at a rate which insures that the ends of the impulse response file and the input file are reached at the same time. For example, if your input file is 10 seconds long and your impulse response file is 5 seconds long and you have set **Length Used** to 1.0 for 1 second impulse response windows, the process would look something like this:



The first 1.0 second frame of the input file (A) will be convolved with the first 1.0 second frame of the impulse response file (ab). Then the window on the input file is moved 1.0 second forward to B, but the window on the impulse response file is moved only 0.5 seconds to bc. This is so both files will finish at the same time. (Actually, the impulse response file reaches the end first and the last impulse response is zero-padded). In the other case, when the impulse response file is longer than the soundfile, sections of the impulse response file will be skipped over. It is a good idea to save the output in a floating point format if using the moving impulse. Since the impulse is continually changing, the scaling is unpredictable.

(Command - G) Gain Change...



The equivalent functions in Sound Designer II and Alchemy are faster, so in most cases you will want to use those. However, SoundHack will give you an RMS value for the file, and will allow a different gain factor for each channel. It will also work on floating point, ADPCM, and µLaw files. Finally, it will allow you to correct for DC offset in your file.

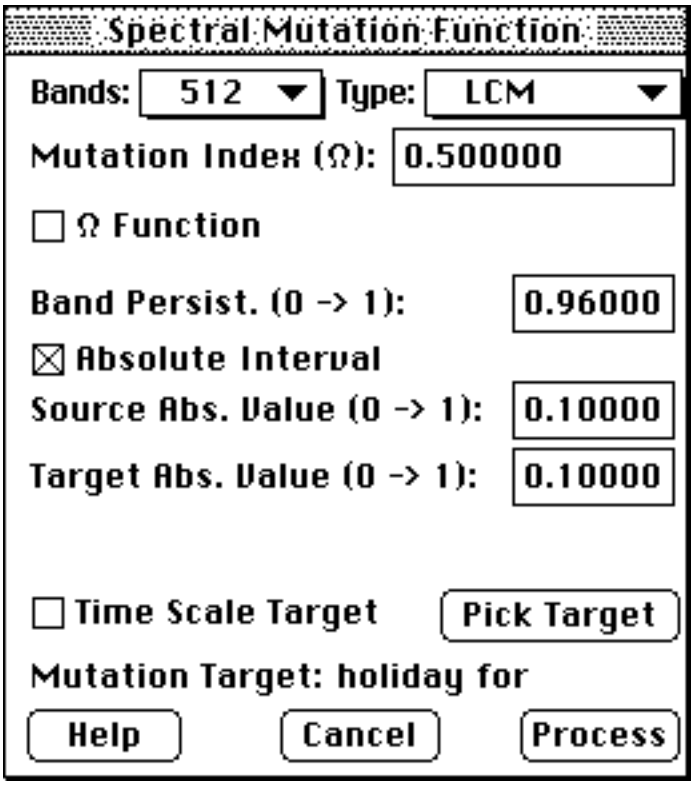
Click on **Analyze** and the peak amplitude, peak position (in samples), RMS values
and DC offset will be calculated. The gain factors will be set to normalize both channels independently and the additional offsets (a number between 1.0 and -1.0) will be set to correct the file. **Change Gain** will create a new file adjusted by the gain factors set. If you are dealing with a monaural file, only the channel 1 information is applicable. **Analyze** will tie up your machine when it is doing its stuff, so please be patient. I will fix this in a future version.

(Command - M) Mutation...

(This section of the manual is written by Larry Polansky.)
The seven different spectral mutation functions (*USIM, ISIM, IUIM, UUIM, LCM, LCM/IUIM, LCM/UUIM*) produce different types of timbral "cross-fades." Each mutation takes 2 soundfiles: a *source* and a *target* , and returns a third soundfile, called the *mutant* . The mutation functions operate on the phase/amplitude pair of each frequency band of the source and target spectra. The output of the functions is a phase/amplitude pair for each frequency band in the mutant soundfile. Each phase/amplitude pair in the mutant is some "combination" of the phase/amplitude pairs of the source and target, for the corresponding frequency band. The mutations work on the sign (Contour) or the magnitude of an interval, or both. They change completely a selected number of bands from the source to the target (Irregular) or partially change all frames (Uniform).

Type

The **Type** box allows you to select between seven quite different mutation functions: *USIM* (Uniform Signed Interval Mutation), *ISIM*(Irregular Signed), *IUIM* (Irregular Unsigned), *UUIM* (Uniform Unsigned), *LCM* (Linear Contour Mutation), and the concatenations *LCM/IUIM* and *LCM/UUIM*. (For specific definitions of the functions, see below. For more information see the two articles cited in the bibliography).



Try starting with the simplest ones, the *USIM* (a simple spectral crossfade) and the *ISIM* (a spectral replacement). These two mutations, unlike the *UUIM* and *LCM*, actually arrive at the source or target, depending on which direction you mutate (that is, you will actually hear the source or target with Ω = 0.0 or Ω = 1.0, respectively). The *IUIM*, *UUIM* and the *LCM* will, with Ω = 1.0, give an image of the target. These "incomplete mutations" mutate either the sign or the magnitude, but not both of the intervals between the amplitudes of successive spectral bands.

The concatenations (*LCM/IUIM*, *LCM/UUIM*) are "pipes": they apply the second mutation to the output of the first. The concatenations are complementary: for each frame the *LCM* mutates sign while the *IUIM* and *UUIM* mutate magnitude. Different frequency bins are used for each "stage" of the concatenation, so the mutation trajectory can be rather unpredictable.

Mutation Index (Ω)

Each of the mutation functions uses an *index*, called Ω (omega), or an Ω-function. This determines the amount of spectral mix, from 0 to 1, between the source and target resulting in the mutant. Ω = 0 results in all source file, Ω = 1 all target. Ω may vary over the course of the mutation. A constant index will result in a sound which is a spectral mix of the source and target. More dynamic sounds are produced with an index function, which changes Ω over time.

Absolute Interval

There are two methods used by the mutation functions to compute intervals between frequency bands: *Absolute* (the default) and *Relative*. You may check or uncheck the **Absolute Interval** box to get these two methods. If Absolute intervals are used, you may specify an absolute amplitude value

between 0.0 and 1.0 for the source and target (Source Abs. Value, Target Abs. Value), from which all intervals will be taken. The choice of values can produce interesting effects, often "centering" the frequencies in which the mutation happens, making the mutations themselves less extreme. If two different values are used, amplitudes will be "transposed" from the source to the target. The use of Absolute intervals rather than Relative will be most noticeable for the *LCM*, *IUIM* and *UUIM* , as well as the concatenated mutations. Low values (around .1 - .2) are a good place to start (note that .1 means 1/10th of the total amplitude of the soundfile's spectra).

If Absolute is unchecked (Relative intervals), each mutation function uses amplitude intervals between successive frames of the spectra, multiplied by Ω, to create the corresponding frame of the mutant sound. Relative interval mutations will tend to "drift," often in extreme ways, and the *ISIM* and concatenated mutations may never "arrive." However, some very interesting sonic results may be produced in this way.

Delta Emphasis

If the mutation uses Relative Intervals (**Absolute Interval** box unchecked), you can set a value for *Delta Emphasis* (DE). DE allows control over the degree to which successive mutation intervals are emphasized in the resulting mutant. DE values range from -1.0 to 1.0, with the default at 0.0 (no emphasis or de-emphasis). For positive DE values, the current frame's intervalic characteristics will be emphasized more than the previous mutant frames. For negative values, the current frame will be "damped," emphasizing the previous information. One way to think about this is as a way of "slowing down" the mutation: a negative DE value will keep the more chaotic mutations from getting "out of control." A negative DE value will function as a low-pass filter, averaging the previous spectral frames into the current output. Positive DE values will accentuate the often high-frequency activity of the mutations. *Delta Emphasis* can be useful in tailoring the relative interval mutations, especially the "incomplete" ones.

Band Persist (Irregular Mutations Only)

Band Persist pertains only to irregular mutations, the *LCM* , *ISIM* , *IUIM* and the concatenations (and not to uniform mutations, the *USIM* and *UUIM*). It will only appear on the screen when irregular mutation is selected. High values for *Band Persist* (towards 1.0) will produce more "stable" mutations. Low values (towards 0.0) will introduce a kind of frequency pumping at the frame rate. Try changing the number of bands for unusual results.

In irregular mutations, not every frequency band is mutated for each FFT frame. Ω determines the percentage of bands that are mutated for a given frame. If a band is mutated, it completely assumes the particular characteristic (interval sign or magnitude) of the target interval, and retains either the sign or the magnitude of the source interval. For example, the *LCM* takes the sign of the target interval, and "pastes" it onto the magnitude of the source. However, it only does that for (Ω * #-of-bands). The selection of which bands to mutate in irregular mutations is done stochastically, but setting *Band Persist* high will ensure that once a band is mutated, it will keep being mutated as long as possible. That is why a high value for *Band Persist* will stabilize these highly unusual mutations, making them a bit more "well-behaved." A good

experiment is to try an irregular mutation (*LCM* , *IUIM* , *ISIM* , the concatenations) with a fixed Ω, and two different values of *Band Persist*, one high and one low.

Time Scale Target

The form of the mutation functions used in SoundHack require that each soundfile (source, target, mutant) be of equal length. The default technique is to truncate the longer of the two files, producing a mutant which is the length of the shorter file. If **Time Scale Target** is checked, the target soundfile will be time-stretched or -compressed to be of the same length as the source. Other techniques are of course possible (including windowing and zero-padding), and we encourage other software developers to investigate these.

Theoretical Descriptions of the Mutation Functions

The mutation functions can be classified as follows:

	Uniform	Sign	Magnitude
USIM	yes	yes	yes
ISIM	no	yes	yes
UUIM	yes	no	yes
IUIM	no	no	yes
LCM	no	yes	no

Note that the opposite of "irregular" is "uniform." Uniform mutations do something to every frequency band. How much they do depends on Ω. Irregular mutations change a given frequency band completely from source to target (sign, magnitude or both), but the number of frequency bands they operate on depends on Ω. Note that there is no *UCM* given here (it wouldn't make sense).

The *USIM* and the *ISIM* are the simplest mutations, a spectral cross-fade and spectral band-replacement, respectively The *UUIM* and the *IUIM* are two different ways of "pasting" the magnitude differences of the target spectra onto the sign of the source, resulting in a mutant which, when completely mutated, is still some combination of the source and target. The *LCM*, perhaps the most unusual sounding and difficult mutation to control, does the opposite, pasting the signs of the target onto the magnitudes of the source.

The functions are defined below. *S* and *T* are the source and target soundfiles. *Si* , *Ti* are the amplitudes for a given frequency band of the FFT for the *ith* frame of the sound. *Sj* , *Tj* are either the amplitudes of the same band in the previous frame (Relative Interval) or some absolute amplitude (Absolute Interval). *Mi* is the new amplitude of the given frequency band of the current frame of the output sound, *Mj* is the amplitude for that band in the previous frame of the output sound (Relative Interval), or some absolute amplitude value between 0.0 and 1.0 (Absolute Interval). *Tint* , *Sint* and *Mint* are the signed magnitude intervals between the amplitude of the current frame for a given band, and the amplitude of that band in the previous frame (Relative) or to some fixed amplitude (Absolute). Each of the equations below applies to one frequency band of the source, target and mutant soundfile spectra. In other words, for all of these functions, *Si* , *Ti* , and *Mi* run from 0 to the

number of bands in the FFT.

|Si - Sj| = Smag - and - *(|Si - Sj|)/(Si - Sj) = Ssgn*

• Uniform Signed Interval Magnitude *(USIM)*
*Mi = Mj + (Sint) + Ω * (Tint - Sint)*
- where *Sint* and *Tint* are *(Ssgn * Smag)*
and
*(Tsgn * Tmag)* respectively

• Uniform Unsigned Interval Magnitude *(UUIM)*
*Mi = Mj + Ssgn * (Smag + Ω * |Tmag - Smag|)*

• Linear Contour Mutation *(LCM)*
*Mi = Mj + Tsgn * Smag* (for mutated intervals)
*Mi = Mj + Ssgn * Smag* (general form for non-mutated intervals)

• Irregular Unsigned Interval Magnitude *(IUIM)*
*Mi = Mj + Ssgn * Tmag*
(for mutated intervals; non-mutated intervals same as *LCM* above)

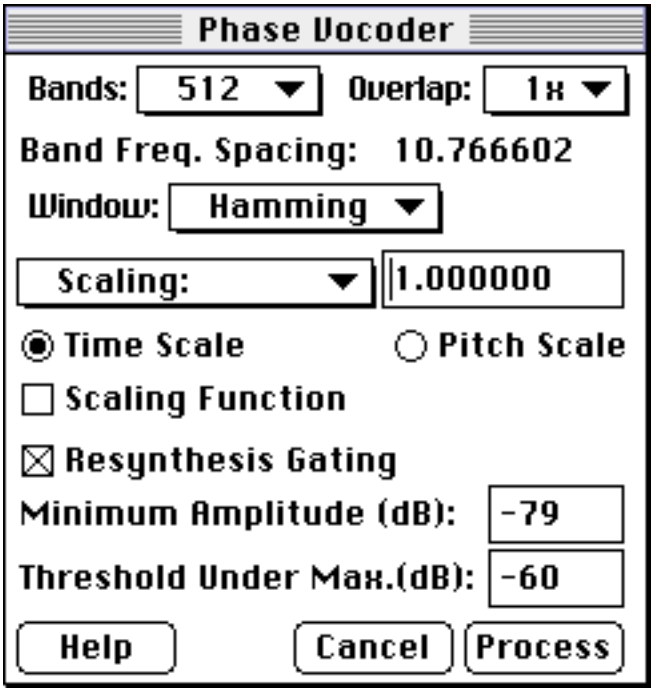
• Irregular Signed Interval Magnitude *(ISIM)*
*Mi = Mj + Tsgn * Tmag*
(for mutated intervals; non-mutated intervals same as *LCM* above)

Notes:
1) In Absolute Interval mutations, *Mj,* the absolute amplitude to which the new interval is added, is interpolated between the absolute values for source and target, according to the value for Ω.
2) The Irregular mutations are in two forms: one for the case when that particular frequency band is chosen for mutation, one for when it is not.

More Information and Acknowledgments

For more on morphological mutations, including their use in other contexts, please contact Larry Polansky at Dartmouth College, Bregman Electro-Acoustic Music Studio, Music Dept., Hanover, NH 03755, email: larry.polansky@dartmouth.edu. Various students at Dartmouth, including Chris Langmead, Eric Smith, Steve Berkley and Martin McKinney have provided invaluable assistance over the past few years in helping me to formulate both the spectral mutation ideas and accompanying software techniques. Ken Overton, Sergei Kossenko, and Chris Langmead all contributed valuable suggestions to this documentation.

(Command - P) Phase Vocoder



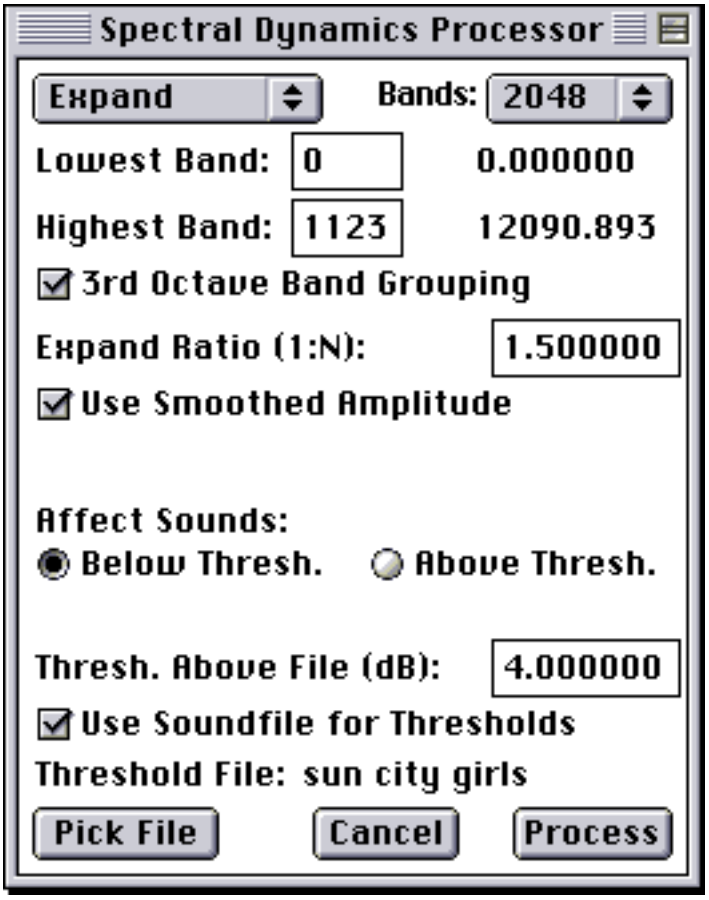
This process allows one to change pitch without changing the length of the soundfile or to change length without changing pitch. It does this by extracting amplitude and phase information for from 8 to 4096 frequency bands with a bank of filters. If time stretching is desired these phase and amplitude envelopes are lengthened (or shortened, for time compression), and then given to a bank of oscillators with corresponding frequencies to the filters. For pitch shifting, the envelopes are untouched, and given to a bank of oscillators with frequencies related by the pitch ratio.

To use the phase vocoder, set the number of **Bands** to the number of filter-oscillator pairs one would like to use. A large number of bands will give one better frequency resolution, a small number of bands will give one better time resolution. The **Window** menu allows one to chose different pre-FFT windows for different filtering characteristics. Only the **Hamming**, **von Hann** and **Kaiser** will give good results (the others are there only because I wanted to use a single menu throughout the program for all window selection). The **Overlap** setting adjusts the size of the filter 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. A sharper filter will differentiate better between frequencies which are between bands, but responds to amplitude changes slower. Click the **Time Scale** button for time scaling, **Pitch Scale** for pitch scaling. Type the scale factor in the **Scale** box. Click on the word **Scale** (a popup menu) to specify time scaling by the length desired, or pitch scaling by equal tempered semitones. If one wants the time expansion factor or the pitch transposition factor to change during processing, click the **Scaling Function** box, and the **Draw Function...** button. This will bring up the **Draw Function Window**, which is described later in this document.

Resynthesis Gating performs a simple spectral gate which lets only some of the spectral data through. If a band is below the **Minimum Amplitude** it is not let through. **Threshold Under Max.** cuts off all bands which are lower than the threshold below the peak band in a given block of samples. So if the peak

band has an amplitude of -7 dB and **Threshold Under Max**. is set to -40 dB, all bands below -47dB will be cut off.

(Command - D) Spectral Dynamics...



This process performs standard dynamics processing (gating, ducking, expansion, compression) on each spectral band individually. It has individual threshold detection for each band, so that one band could have the dynamics process active, while another is inactive. The process can be limited to affect only a specific frequency range. One can select whether to affect sounds which are above the threshold or sounds which are below the threshold. The threshold level can be set to one value for all bands, or it can be set to a different value for each band by reading in and analyzing a soundfile. This soundfile's amplitude spectrum is used for the thresholds for each band. This is especially useful if there is a sound that one wants to emphasize or de-emphasize (hiss or hum).

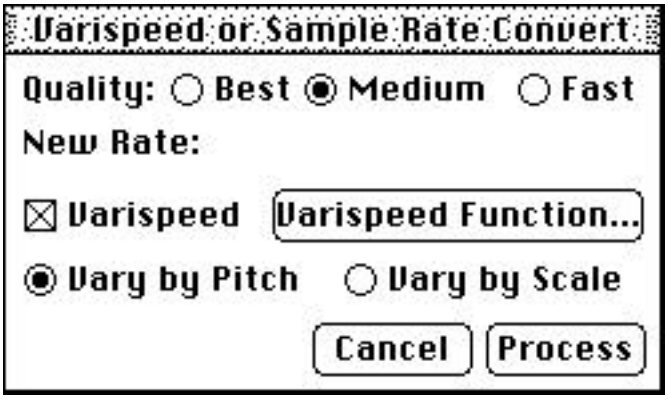
Most controls are self-explanatory. The first popup menu allows you to select the type of process to use; gating/ducking, expansion or compression. The second popup sets the number of filter bands to separate the sound into. 512 is a good compromise for the number of bands at a 44100 sample rate as each band is about 43 Hz apart and the filters used have a (512*2)/44100 or .023 second delay. In other words, a pretty good frequency resolution (provided no partials are closer than 43 Hz) and not too much time smearing.

The **Highest Band** and **Lowest Band** boxes allow one to limit the frequency

range affected. **3rd Octave Band Grouping** causes the bands to be grouped with a 3rd octave spacing with a threshold trigger for each group (instead of each band). This octave grouping may give a more natural sounding dynamics process. The next box is either labelled **Gain/Reduction, Expand Ratio** or **Compress Ratio**. It allows you to set the amount of gain or reduction for the bands which are past the threshold when gating. For compression and expansion it allows you to set the gain ratio. When affecting sounds below the threshold, the compressor and expander hold the highest level steady and affect lower levels (also known as "downward" expansion or compression) . When the process is set to affect sounds above the threshold, the compressor and expander hold the threshold level steady and compress/expand up from there.

Use Smoothed Amplitude to avoid abrupt gating. Instead of comparing the input soundfile directly to the threshold, a windowed average of the input is compared to the threshold. This setting allows one to reduce the "martian voices" effect (a common problem in spectral dynamics processing) and is particularly nice for hiss and hum removal. **Attack/Decay Time** allows one to set the speed that each triggered band opens or closes. The default is the minimum time for the number of bands used. If this value is set too slow, one loses transients, too fast and you start modulating the soundfile (whistling sounds). **Threshold Level** is where you set the threshold level! If you check **Threshold Relative to Peak Amp.,** the threshold is now set relative to the peak amplitude for each block of sound processed. For example, if you are using the spectral gate, and the loudest frequency band for the current block of sound has an amplitude of -12 dB and the threshold is set at -40 dB, the gate will be active for sounds below -52 dB.

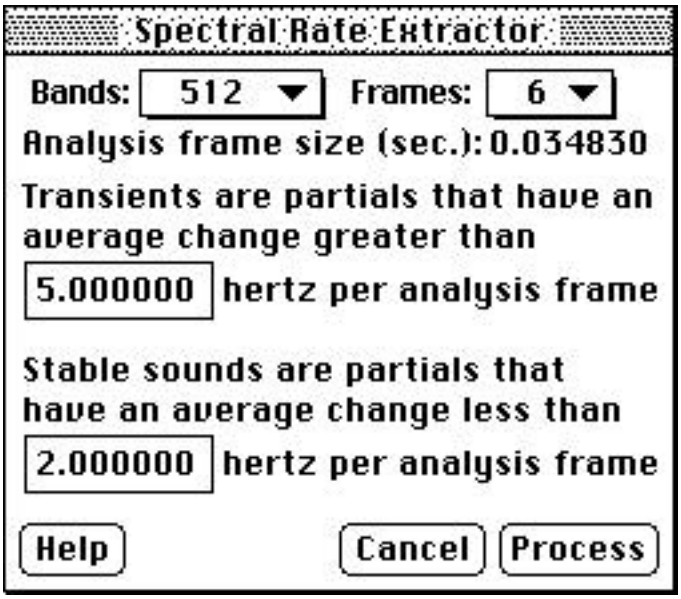
(Command - V) Varispeed...



This function is a high quality sample rate convertor as well as a variable sample rate conversion utility (varispeed). The **Varispeed** box enables this feature. The **Varispeed Function...** button will bring up the **Draw Function Window**, giving one control over a 10 octave varispeed. The **Quality** buttons give one control over the size of smoothing filter used, and the resultant quality of interpolation/decimation. The **Vary by Scale** and **Vary by Pitch** buttons allow one to draw a curve for either pitch or scaling factor.

(Command - X) Spectral Extraction...

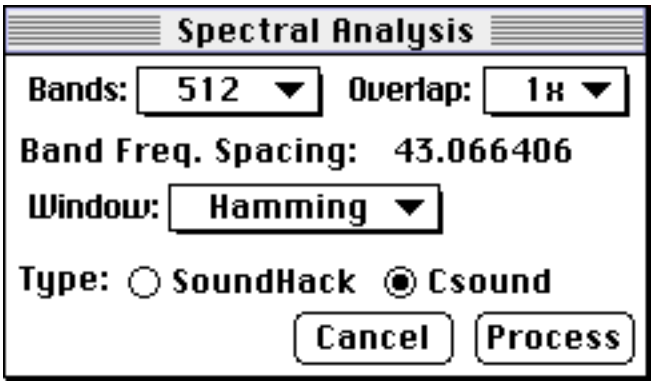
This function attempts to separate the stable (pitched) and transient (unpitched) parts of a sound. It does this by measuring the speed of frequency deviation. If the deviation is too quick, it is marked unpitched information and output to the transient soundfile. If it is too stable, it is marked pitched information and output to the stable soundfile.



You can control the separation with this dialog box. Set the **Bands** to a high number if the sound being processed is harmonically dense, otherwise keep it around 512. Setting the number of **Frames** allows you to set the size of the analysis frame (in multiples of FFT frames). Set this higher if you are having difficulty separating the pitched material, lower if you are having difficulty separating the transient material. The two frequency values specify the amount of change allowed during each analysis frame. In this example, if the harmonic deviates by more than 5 hertz in 0.035 seconds it is put into the transient soundfile, if the harmonic deviates by less than 2 hertz in 0.035 seconds, it is put in the stable soundfile.

This function draws heavily from the work of Zack Settel and Cort Lippe on the ISPW workstation using Max-DSP. Thank you Zack and Cort for sharing a great idea.

(Command - -) Spectral Analysis/Resynthesis...



This function will create a Csound or SoundHack format spectral data file from a soundfile (analysis) or create a soundfile from either format spectral data file (resynthesis). With this, you can create files to be used with Csound's pvoc unit generator or for you to use with your own programs to process the spectral data directly. The format of the spectral data file (a limited version of the Csound pvanal format) is a header, followed by multiple frames of spectral data.

Here is a C structure describing the header:

```
typedef struct
{
  long  magic;          // 517730 for Csound files,
                        // 'Erbe' for SoundHack files
  long  headBsize;      // byte offset from start to data
                        // (usually sizeof(SpectHeader))
  long  dataBsize;      // number of bytes of data not including
                        // the header
  long  dataFormat;     // (short) format specifier
                        // always 36 for floating point
  float samplingRate;
  long  channels;
  long  frameSize;      // number of points in FFT
                        // (number of bands * 2)
  long  frameIncr;      // number of new samples each frame
                        // (frames overlap)
  long  frameBsize;     // bytes in each file frame
                        // frameBsize = sizeof(float)
                        // * (frameSize >> 1 + 1) << 1;
  long  frameFormat;    // this is either 3 for SoundHack
                        // files (amplitude & phase) or 7 for
                        // Csound files (amplitude & frequency)
  float minFreq;        // 0.0
  float maxFreq;        // maxFreq = samplingRate/2.0;
  long  freqFormat;     // flag for log/lin frequency
                        // (always 1 for linear)
  char  info[4];        // extendible byte area
} SpectHeader;
```

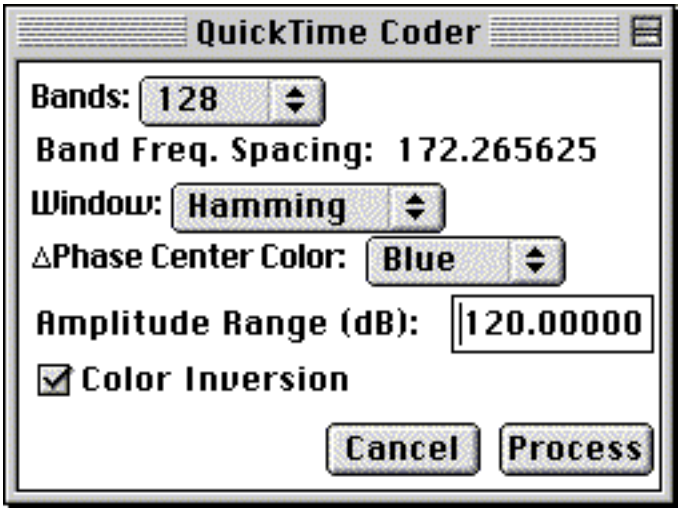
The subsequent frames of spectral data are organized as follows:

```
// frameFormat == 3, amplitude and phase pairs, SoundHack file
typedef struct
{
    float    amplitude;    // from 0.0 to 1.0
    float    phase;        // from 0.0 to (2.0 * pi)
}   band;

band    spectralFrame[(frameSize >> 1) + 1];

// frameFormat == 7, amplitude and frequency pairs, Csound file
typedef struct
{
    float    amplitude;    // from 0.0 to 1.0
    float    frequency;    // from 0.0 to samplingRate/2.0
}   band;

band    spectralFrame[(frameSize >> 1) + 1];
```

If the spectral file is stereo, the frames are interleaved, first left then right.

Included with SoundHack is the source code for a simple spectral data processor (Spectral Assistant) which should illustrate how to read and write this format.

(Command - U) QT Coder



This function will convert your soundfile into a QuickTime™ movie and will convert the QuickTime™ movie back into the same sound. You can also turn any QuickTime™ movie into a soundfile.

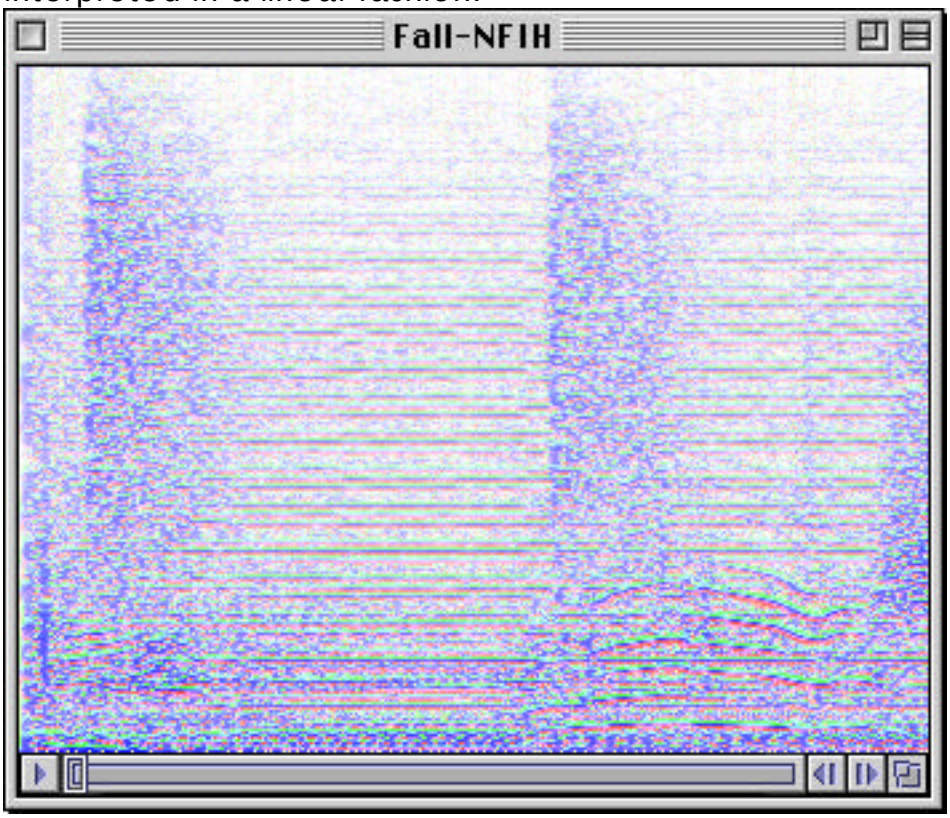
The QuickTime™ movie will contain a series of sonograms representing your sound. The sonograms have linear frequency on the vertical axis and time on the horizontal axis. Color saturation is used to represent ampiltude and hue is used to represent delta phase.

There are very few settings in this function. The number of bands one chooses will affect the size of the movie frame. 256 bands will require a 343x257 frame (a 4:3 aspect ratio). Since the QT Coder uses 32-bit uncompressed images, it consumes a lot of memory, and the memory need increases exponentially as the number of bands increase. For example: a 512 band, monaural QT Coding will require about 3 megabytes allocated to SoundHack.

Window: selects the FFT windowing function. Kaiser is best for band separation, Hamming is best for smooth band transition. The choice of window is probably not critical to most uses of this function. **ΔPhase Center Color:**selects which color will represent a Δ phase change of zero degrees. This color usually becomes the predominant color in the sonogram. **Amplitude Range (dB):** determines which sounds are encoded in the QuickTime™ movie. For a high fidelty encoding, 120 dB seems sufficient. **Color Inversion** inverts the RGB color values.

After the QT Coder creates the QuickTime™ movie, you will be able to open it in any QuickTime™ application and view, edit or modify the sonogram. After doing this, you will be able to open the modified movie in SoundHack and convert it back into sound using the QT Coder again.

One could also use the QT Coder to convert any QuickTime™ movie into a soundfile. The resultant sound will tend to be repetitive, it will also tend to be biased toward the high frequencies since the frequency in the frame is interpreted in a linear fashion.



This function is rather experimental and I am not quite sure what it will be good for yet.

For future versions I am working on an exponential frequency encoding scheme as well as color schemes which follow visual perception models.

(name that tune?)

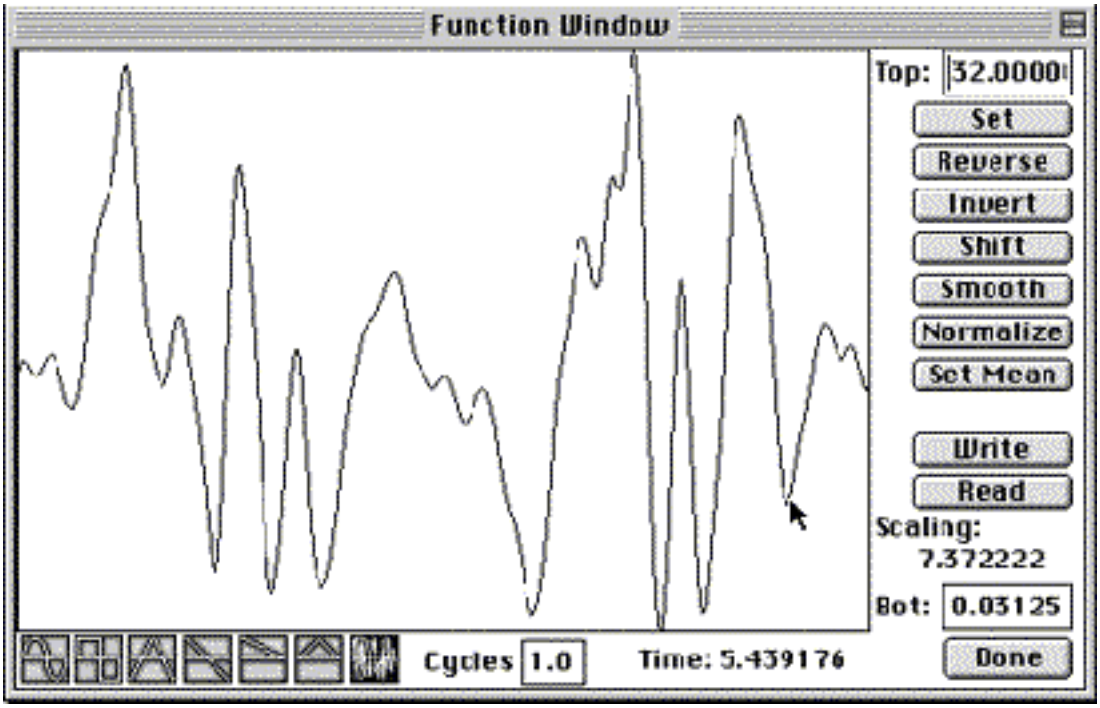
(Command - ;) Normalize

This does a simple, no-questions asked, normalization of the selected soundfile.

Draw Function Window

This window allows for the creation of control functions. It actually creates a function of 400 entries (one per pixel) which is then interpolated on during processing. The function controls either binaural position, mutation index, degree of time stretching, amount of pitch shifting, or degree of varispeed depending on which process you are using.

To create a control function, you can simply draw in the function window (your mouse position is continually tracked and translated into relevant units) or click on one of the preset waveform buttons. The waveform buttons will fill the function with as many cycles as are set in the **Cycles**: box. If you click a waveform after there is something already in the function window, the waveform will modulate the function, rather than replace it. The **Set** button will allow you to set all of the points to the same value. The **Reverse** button will time-reverse the function. The **Invert** button will flip it. It **Shift** button will rotate the function by up to 399 positions. The **Smooth** button will average adjacent entries in the function. This smoothing wraps around the beginning and end of the function, so watch out. The **Normalize** function will increase/decrease the magnitude of the function to fit the range of the window. The **Set Mean** button will scale the function so that the value you enter becomes the mean, or average value.



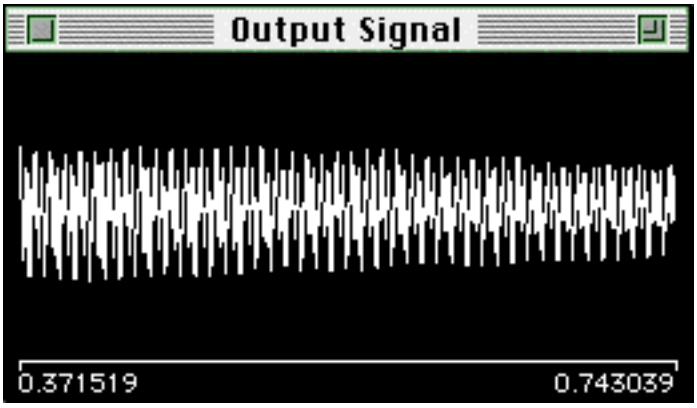
The boxes in the upper right and lower right corners allow a very primitive type of zooming. There is no facility for selecting, copy, paste or cut. One can read or write control functions as soundfiles by clicking the **Read** and **Write** buttons. The **Time:** legend refers to the time in the input soundfile and the other legend **(Scaling:)** is updated depending on how the control function is applied.

SOUNDFILE MENU

This simply allows you to select between the open soundfiles. The first ten soundfiles open are given command key equivalents (Command - 1 to Command - 0).

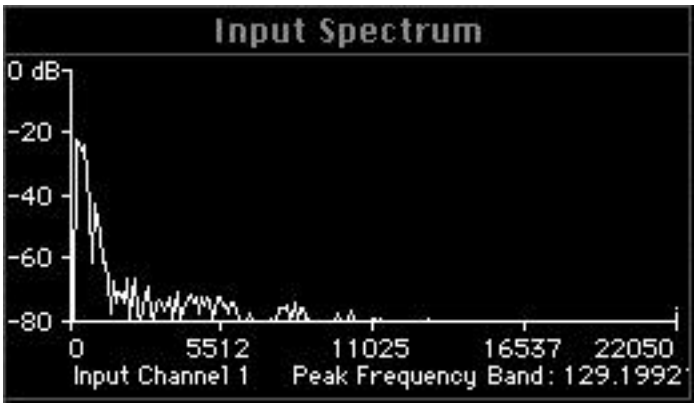
CONTROL MENU

Show Signal



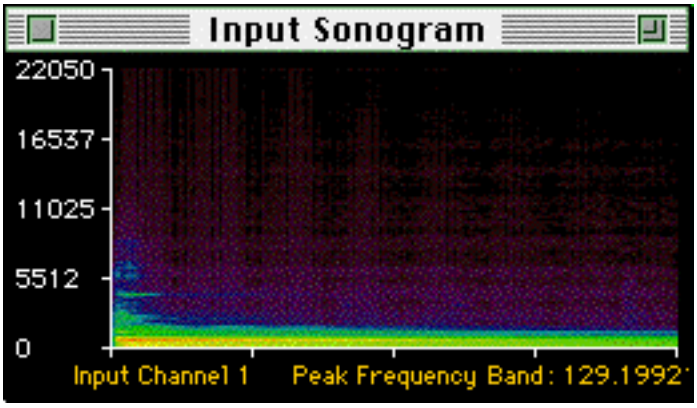
This will bring up a window to show the sound whenever SoundHack reads or writes sound (except during file copying and normalization). There are separate windows available for all active soundfiles.

Show Spectrum



This will bring up a window to show the spectral data in all spectral operations (most everything but varispeed, which I do in the time domain). There are separate windows available for all active soundfiles.

Show Sonogram



This will also show spectral information but over time. Intensity is represented by different colors where purple-red is the lowest intensity, green is mid intensity and bright red is the highest intensity. This display will slow down any Macintosh dramatically.

Pause Process

This allows you to pause during a long process. This is especially useful if you would like to here the sound you have processed so far. If you are running a convolution, it sometimes takes a while to pause (up to 3 minutes on a slow Mac II).

Continue Process

This will resume processing where you left off.

Stop Process

This will kill your process and close the output soundfile.

Load Settings...

This loads a previously created settings file. This is also done automatically on startup to the "SoundHack Preferences" file in the ":System Folder:Preferences" folder.

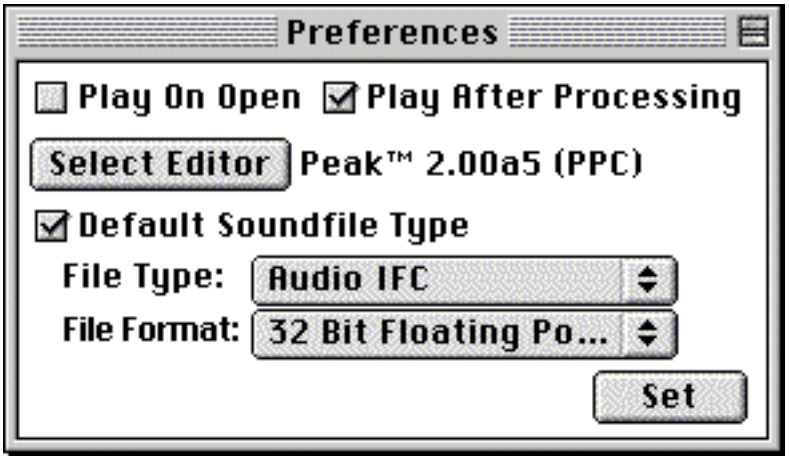
Save Settings...

This will create a settings file which contains the current settings from the binaural, convolution, spectral analysis, spectral dynamics, mutation, phase vocoder, varispeed, gain and preferences dialog panels. This is done automatically when you quit SoundHack to the "SoundHack Preferences" file in the ":System Folder:Preferences" folder.

Default Settings

This will reset all internal settings to a default set. Recommended if you suspect your "SoundHack Pref" file is corrupt.

Preferences...



This allows you to set a few things. You can set SoundHack to automatically play a soundfile when it is opened. This is a very nice way to set things if you are using SoundHack as your web browser helper application. If you set an editor, then you can use the **Close & Edit** command (described earlier). The **Default File Type**, when set, will give you the selected file type and format whenever you create a new soundfile. **Play On Open** will play the file as soon as it is opened, **Play After Processing** will play the output file as soon as the current Hack process is done with it.

FUTURE PLANS

```
1.0   SoundHack is finished. Tom and Betsy take a holiday.
.90   Last major release, only bug-fixes and minor updates from
      now on. Make a suggestion for final features. Time is running out!
      Port to IRIX, Linux? Solaris, Rhapsody, Wintel. AppleEvents and
      scriptablity. MPEG. Spectral plugins?
```

.89 Mark Dolson phase vocoder enhancements. Simple graphic filtering. SDS, SDI, TX16W, OMF, Sonic, ProTools, AIFF resource soundfile formats. QT coder: log frequency sonograms, new color encodings. Buffered read and write routines.

BUG FIXES AND REVISIONS

.88 QT coder - changes soundfiles into QuickTime™ movies into soundfiles. 24&32 bit SDII files, IMA4, MACE3, MACE6, aLaw and µLaw AIFC files, aLaw Sun files. 32-bit float AIFC files compatible with Mills' Csound. Peak sample saved in all 32-bit float files. 32-bit float playback. QuickTime/AIFF files fixed. Increased sample rate conversion quality. Motorola's libmoto library integrated into app. Looping playback. WAVE loops and markers. Enhanced Draw Function dialog. Functions work much more predictably with varispeed (no more fast high pitch - slow low pitch). Tiny soundfile playback fixed. 3rd octave band grouping added to spectral dynamics and irregular mutation.

.87 Spectral files are now treated as soundfiles. Simple spectral analysis/resynthesis added. Quick normalization added. Many cosmetic/GUI changes. New binaural filters added. Spectral dynamics will trigger on an averaged amplitude value (good for hiss removal). Draw function debugged for the umpteenth time. Preferences are now savable.

.86 The shareware version now is just the 68k version. Multiple soundfiles supported. Another sample rate conversion bug (errant pointer deallocation) squashed. CD Audio import added. Channel splitting added.

.85 Soundfile playback now works for all soundfiles which are 8-bit (twos, raw or µLaw) or 16-bit linear (twos). The apple events for open and quit application added. Spectral dynamics fixed again.

.84 Separate display windows for most signals added. Sonogram window added. Off screen drawing keeps windows updated. Most all dialogs now modeless.

.83 Popup bugs fixed, convolution brightening added.

.82 Soundfile recording has been added.

.815 24 and 32 bit AIFF and AIFC supported.

.81 Spectral extraction added. Spectral dynamics now allows threshold to be set relative to the peak amplitude in a block of sound.

.803-6 Various bug fixes. A full disk now stops any process. Samples which are too loud now clip instead of going out of range. Greater post convolution gain is now possible. Normalization no longer clips. The packmode no longer changes when a new soundfile type is selected. Mixed stereo/mono mutation no longer crashes!

.80 Split into shareware (0.74) and commercial (0.80) versions. Added Mutation functions. Ported to the Power Macintosh.

.74 Fixed early ending sample rate conversion problem. Fixed the spectral display. Added attack/decay time to spectral dynamics. A moving binaural effect is added. Allow raw data files to have text packmode and text files to have raw packmodes. Fixed various GUI complaints (0.743) Double-clicked soundfiles now open correctly in all versions.

.73 Fixed varispeed clicking for the third time. Fixed the output soundfile dialog. Ported the code from Think C to Codewarrior. Spectral display is now broken and needs to be redone (disabled for the present).

.72 Fixed 16 bit Microsoft RIFF/WAVE file bug.

.71 Fixed a minor bug which caused spectral dynamics crashing. Also fixed a typo (thank you Phil Burk).

.70 Added spectral dynamics process. Added AIFC and Microsoft WAVE file support. Added 4-bit ADPCM. Opens double-clicked documents. More types show up in "Open" menu item. Fixed stupid error about number of bands in the phase vocoder and the spectral dynamics processor. It used to refer to the number of FFT points used, it now correctly refers to the number of filter bands used.

.68 Added import and export of SND resources. Limited soundfile playback to AIFF and AIFC until Apple comes up with a stable sound manager. Sound Manager 3.0 is required for playback of 16-bit files. Added a limited spectral gate to the phase vocoder. Fixed old bug in pitch shifting.

.67 Added soundfile playback.

.66 Added windowing selection for both convolution and the phase vocoder. Added "Vary by Pitch" /"Vary by Scale" selection in varispeed.

.65 Fixed crashing at the end of long varispeed calculations, cleaned up memory allocation (especially for convolution), added ring modulation (spectral convolution), cleaned up dialog annoyances. Made output window resizable, and made it white on black (like a scope). Added bibliography screen.

.64 Added varispeed processing, added many things to the function dialog.

.63 Added TEXT soundfile format. Added interpolation, sinc and number of cycles to function dialog. Added ramp enveloping of impulse to convolution. Added popup menus.

.62 Added function window for multirate phase vocoder.

.61 Sped up pitch transposition significantly, allowed pitch transposition by semitone and time warping by desired length.

.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.

.5 Gain change added, 8-bit linear, 8-bit mulaw and 32-bit float added. (Jan 1992)

```
.4    Binaural filtering and process control added.
.3    Convolution and NeXT/Sun file formats added.
.2    Sound Designer II files and Header change added.
.1    First release of SoundHack, converts AIFF files to IRCAM and back.
      (Fall 1991?)
```

BIBLIOGRAPHY

Apple Computer, Inc., Inside Macintosh, Volume 1-6, Addison Wesley, Reading, Mass., 1985-91.

Begault, Durand R., Control of Auditory Distance, Ph.D.. dissertation, University of California, San Diego, 1987.

Begault, Durand R., 3-D sound for virtual reality and multimedia, Academic Press Professional, Cambridge, MA, 1994.

Blauert, J., Spatial Hearing, MIT Press, Cambridge, Mass., 1983.

Dolson, Mark, "The Phase Vocoder: A Tutorial", Computer Music Journal 10:4, 1986.

Ellis, Dan, "pvanal.c", part of the Csound distribution, MIT, 1991.

Gordon, J. W. and Strawn, J., "An Introduction to the Phase Vocoder", Digital Audio Signal Processing: An Anthology, editor J. Strawn, Kaufmann, Los Altos, Calif., 1985.

Mark, David and Reed, Cartwright, Macintosh C Programming PRIMER, Volume I, Addison Wesley, Reading, Mass., 1989.

Moore, F. Richard Elements of Computer Music, Prentice Hall, Englewood Cliffs, NJ, 1990.

Polansky, L. "More on Morphological Mutations: Recent Techniques and Developments, "Proceedings of the ICMC., San Jose, 1992.

Polansky, L. and McKinney, M. "Morphological Mutation Functions: Applications to Motivic Transformations and to a New Class of Cross-Synthesis Techniques." Proceedings of the ICMC. Montreal, 1991.

Reed, C. E. and Passin, T. B., Signal Processing in C, John Wiley, New York, NY, 1992.

Settel, Z. and Lippe, C. "Real-time Musical Applications using FFT-based Resynthesis", Proceedings of the 1994 International Computer Music Conference. Diem Aarhus, Denmark 1994.

Vaseghi, S. and Frayling-Cork, R., "Restoration of Old Gramophone Recordings", Journal of the AES, 40:10, 1992.

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