

Cool Edit version 1.51

Brought to you by Syntrillium Software Corporation
P.O. Box 60274, Phoenix, AZ 85082-0274, USA

Cool Edit is a full featured wave editor for Windows, and was created by [David Johnston](#). Please read on.

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Please see the [Technical Notes](#) for information on setting up Cool Edit to take full advantage of your system.

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Cool Features

- > Multiple instances (copies of Cool) can be loaded, copied to, and pasted from in any mixture of wave formats
- > Cool-Edit can synthesize just about any sound using the noise and tone generation functions along with the various wave transformation functions
- > Very clean sample rate conversion possible with Convert Sample Type.
- > Batch run any series of commands on a set of files, save as any file type (great for file type conversion of hundreds of files) by using the batch processor.
- > Any sample can be used as an "instrument" and set to music, like a dog barking Jingle Bells, or Tarzan yelling the national anthem
- > Waves of any size can be edited, limited only by hard drive space
- > Supports PCM, Microsoft ADPCM and IMA/DVI ADPCM .WAV, Sound Blaster .VOC, raw PCM, SMP, ASCII Text, AU, Apple AIFF and more. See data formats
- > Practically any file format can be loaded successfully into Cool by using the raw PCM file type, and then Adjusting the sample rate, bit rate, and number of channels, as well as performing Byte Swaps to get the byte ordering and interpretation correct.
- > Add 1960's sounding flanging and special effects using the Flange function.
- > Continuous echo of all or part of a sample with filtering for successive echoes possible with Echo. Simulate 3D echoes with the Echo Chamber. Simulate realistic reverb using the Reverb function.
- > Speed up or slow down samples without affecting pitch, or raise and lower pitch without affecting tempo by using Stretch.
- > Supports descriptive information, and bitmaps when saved in .WAV format
- > Supports a cue list, and a play list when saved in .WAV format for playing portion of a wave in any order, with looping
- > Built in CD player when the [MCI] CD Audio driver is loaded, and a valid music CD is in the drive
- > Brainwave synchronization function available to create sound files that alter your state of mind
- > Record and Playback waves using any Windows compatible sound board, even works with the PC Speaker driver
- > Spatially locate sound sources to appear as if they are coming from different directions by delaying one channel a few microseconds
- > Most functions have programmable Presets to save and easily retrieve your favorite effects
- > You can feel free to experiment, and easily back up a step if you don't like what you did by using the Undo function
- > 8-band Quick Filter for quickly adjusting equalization.
- > FFT filtering and Spectral View make it easy to filter and analyze audio with high precision.
- > Compress/Expand/Limit dynamic range using the Compressor function. Can also act as a noise gate to silence audio between spoken words.
- > Skewed waveforms can be centered properly using the DC Bias filter, and waves can be normalized, faded in, faded out, or panned by using the Amplify function.
- > Frequency analysis at the cursor can be continually be performed on the waveform at the cursor to better see exactly what frequencies are present, and where.
- > Support for unlimited wave formats by allowing others to write custom file formats that Cool Edit will support. See the File Filter API.
- > Support for unlimited effects by allowing others to write custom effects modules that Cool Edit will support. See the Effects Modules API.
- > Noise Reduction possible to reduce broad band noise up to 20dB and constant noises (like that annoying 60 cycle hum) over 70dB.
- > Cool Scripts will remember everything you did to arrive at a certain waveform, or transformation and remember it for the future.
- > Customizable Toolbar (with a little work) possible by customizing the cool.ini file.
- > Assign keyboard keys to play specific parts of your waveform by using the cue list.
- > Ring Modulation, or any type of modulation possible via Generate Tones or Paste Special.
- > Ultimate Distortion effects possible with the Distortion.
- > Generate telephone signals using the DTMF Tone Generator.

File menu

The file menu displays file handling options.

File Menu Options:

New

New Instance

Open

Open As...

Open Append...

Re-Open

Close

Save

Save As

Save Selection

Exit

Answers to common Questions (a.k.a. Troubleshooting)

Q: When repeatedly hitting Play too fast, or starting and stopping Monitor Source too quickly, my system hangs or crashes.

A: Try increasing the STACKS line in CONFIG.SYS to STACKS=12,512. On some configurations, if the stacks are set too low, there will be problems starting and stopping audio too quickly.

Q: I had saved some presets using a previous version of Cool Edit, and now they're gone. Where did they go?

A: They are most likely still in your COOL.INI file, but in creating new versions, sometimes it is impossible to keep the parameter orders and such the same for some functions. Look in your COOL.INI file and find your previous presets and print them out or write them down. Perhaps you can re-enter them from looking at the data. Also, you can *try* to just copy the entries from the previous function section to the new one. Sections that are 100% compatible are [Channel] from 1.33 and the new [Channel Mixer] section; [NewFlanger] from 1.33 and the new [Flanger2] section. Entries from these can be copied straight across. Entries from 1.33's [Filter] section can be copied straight across to the new [Filter2] section, but be aware that any filters designed are going to be assumed as if they were done at 44.1 KHz. This means a bandpass filter at 5khz done on a 22 KHz wave will now be assumed to be a 10 KHz filter at ALL sample rates. The new [Amplify2], [Stretch2], and [Tones2] sections are not compatible with the previous 1.33's [Amplify], [Stretch], and [Tones]. Sorry for the inconvenience.

Q: I had written a script using Cool Edit 1.33, and now it won't run properly. Help!

A: Sorry about that. In making modifications and everything more generalized, some incompatibilities may have arisen. When making the conversions, I tried to favor keeping Scripts running properly over remembering presets properly. You can print out the SCP file containing your invalid scripts and then try to reconstruct it by recording a new script and following along what you had done previously.

Q: I cannot load normal wave (*.WAV) files. Why?

A: Do you have the file WAVE.FLT in the same directory as COOL.EXE? The *.FLT files are required for any file loading and saving. Also, no to programs can be running at the same time called "WAVE". If you are running another program whose filename is WAVE.EXE or WAVE.DLL, then rename Cool Edit's WAVE.FLT to something different like WV.FLT.

Q: I get an error message when I try to record. What's up?

A: Check the capabilities of your sound card. You may be trying to record at a rate not supported by your hardware. Some older boards can play at higher rates than they can record. You may also be trying to record in 16-bit mode for an 8-bit card. When COOL is started, the playback abilities are displayed in the audio format box (but this does not necessarily mean you can record at the same rates).

Q: I get an error message when I try to play a wave. How come?

A: Your board probably does not support the audio data sample rate, or sample size. Try checking "Play 16-bit files as 8-bit" in the Settings box. This will convert any 16-bit audio data to 8-bit as the file is played.

Q: I just installed a 16-bit audio card, but my 16-bit sound files still sound awful. Should I take my card back?

A: No. Your card is probably fine. Check to see that the "Play 16-bit files as 8-bit" box is **not** checked in the Settings dialog. If it is checked, your files are being converted to 8-bit before being played. Also be sure you are using the right DMA settings. The lower DMA channels can only support 8-bit audio. Please check your sound board manuals for this information.

Q: Why does it take forever to do things like Filter, and to use Spectral View?

A: These functions use the Fast Fourier Transform (FFT) to convert the waveform from temporal data to frequency data. The FFT does "zillions" of floating point operations to accomplish this. If you do *not* have a coprocessor (this includes 486SX users!) then these operations are going to take a very long time.

A coprocessor will speed these operations up by **at least** a factor of 10, sometimes 20!

Q: Why does it take so long to save as ADPCM Wave format?

A: ADPCM is a compression scheme, which compresses 16-bit samples to 4-bits while retaining much of the quality at higher sample rates. If the multiple pass option is chosen, which provides the highest quality, each block of wave data is compressed seven different ways, and the way which sounds most like the original signal is then saved.

Q: I am using Windows NT. Why doesn't my screen redraw when I make a change?

A: Normally, COOL is set up to "multitask" when the waveform is being drawn, so as not to lock up the PC when drawing large waves. NT automatically multitasks, which throws things out of sync. Make sure the NT Compatibility option is checked under Settings if you experience screen redraw problems.

Q: How long did it take you to make this program?

A: It took me almost a year of programming in the evenings and weekends to produce what you see here. I had actually started writing some of the waveform transformation functions six months prior when I wrote a DOS only version of Cool, which only worked with Sound Blaster files. I first ported that functionality to Windows, then added on functions and features from there, and eventually ended up with this creation. The brainwave function (Transform->Wave) was written around April to May of 1992, and eventually released to ShareWare in July of 1992. I experimented for several months with different sound/relaxation techniques before finding the one which worked extremely well.

Q: Why are some functions not selectable?

A: If you have the unregistered version of Cool, you chose which functions you wanted to use for the editing session. The others will not be selectable until you choose them on the next editing session. Also, some functions only work on stereo files, such as Wave and Channel Mixer (which becomes Invert for mono waves).

Q: How can I see my wave size information in samples instead of time?

A: Double-clicking on the time (or samples Start and End) window will toggle the display between time and samples. This, and double-clicking on the wave to select all, are the only functions that do not have a corresponding menu item or button associated with them. Other shortcuts are double-clicking on the waveform type display to change the waveform interpretation (i.e. interpret the 44.1 KHz wave as a 22 KHz wave), and double-clicking on the green bar to bring up the viewing samples data entry box.

Q: Why do my recordings sound choppy?

A: You probably have too small of a buffer size in the Settings dialog. A minimal buffer size is about 4 seconds. Using a compressed hard drive on a slow PC could also eat up so many CPU cycles, that there isn't time left to do recording. Either try adjusting the buffer size up or down, or record at a lower data rate (i.e. 8 bits instead of 16, or 32K instead of 44K). I tried very hard to ensure that recordings would sound perfect, without any data loss. If you have problems, see if other recording software has the same problems. If so, you may have a hardware incompatibility between you sound card and your main board, video board, or other installed boards.

Q: My computer crashed while Cool was running, and now I lost a lot of hard drive space. What happened?

A: Cool uses temporary files for the editing of waves, and the undo buffer. Since some of these files may have been open (the editing temporary buffer is always open) during the crash, the files were never closed properly. Exit Windows, and type **CHKDSK /f C:** (use whatever drive letter is missing the space). If you get "Convert lost chains to files?" or something like that, answer yes. The missing data will now be in the root directory as *.CHK files, which you can then safely delete.

Q: I keep getting "parity error" messages, and the program keeps crashing. What's going on?

A: Some older style PCs do not handle memory properly, and Cool just makes the problem worse since it is very CPU intensive. The "parity error" message appears when one of the bits that were saved into DRAM have changed unexpectedly (indicating bad memory). Most boards have the first 64K of memory

on the motherboard, so the problem could be either in your memory arrays, or on the motherboard.

Q: Cool does something really weird that I didn't expect, and I think it's not working right.

What's a person to do?

A: A person should send me mail explaining the bug they've found, and I will respond by either finding a fix in the program, or just letting them know that the program is supposed to do that.

Q: Cool seems to hang when I have multiple copies all drawing their screens at the same time, or if I resize Cool Edit while it is in the middle of redrawing

A: When "NT Compatibility" is not checked in the Options:Settings dialog, Cool Edit may freeze up if multiple copies are open, and two or more copies are updating their waveform display area at the same time. This does not happen if you are only working on a single copy at a time, but can happen if, for example, one copy of Cool Edit just finishes a calculation and is updating its waveform display, and another copy just finishes loading a new waveform and is now displaying its waveform data at the same time. Again, if you are working simultaneously on many instances of the program, where more than one copy is in the process of calculating and/or loading at the same time, uncheck the "NT Compatibility" box in Options:Settings. This will force Cool Edit to redraw the entire waveform before allowing other processes to run. Doing this will also disallow the ability to play a waveform before it is finished being displayed. Also, do not re-size Cool Edit while it is in the middle of updating the screen after a re-sizing (this is impossible to do if NT Compatibility is turned on). Currently, this may cause a lockup as well. I am working on revising the display code to resolve these problems.

Q: Cool Edit doesn't seem to know the format of waves that have headers, so Cool Edit should know them. Huh?

A: If you find you cannot load or save .WAV format waveforms, if you try to load a .WAV waveform file and the "Choose Sample Rate" dialog appears, or have any trouble with loading and saving in the proper format, then remove all copies of the .FLT files on your hard drive, and copy over the ones distributed with your current version of Cool Edit into the same directory as COOL.EXE. If different versions (older or newer) of the .FLT files are mixed with the distributed versions, these problems could arise. This problem could also result from running a separate program with the same name as the .FLT file (e.g. wave.flit will not be used if an application called wave.exe is running). If this is the case, simply rename the .flt file (e.g. rename wave.flit to wav.flit).

Q: Some functions don't work when I have low memory on the hard drive... ?

A: If the TEMP environment variable points to an invalid directory, some functions may fail. Be sure the TEMP environment variable is set to a valid directory with plenty (at least 1 meg) of hard drive space. Alternatively, you can add the TEMPOVERRIDE= line to the [Size] section of cool.ini (found in the Windows directory) and set it to your temporary drive and directory. You should have as much free space on the temporary files drive as twice the size of the largest file you will be working with, as a rule of thumb.

Other Programs from Syntrillium

Some programs the author of Cool Edit has written in the past are also available. If you can't find them on the net anywhere, we will gladly send you a copy for about \$5 for the disk and postage.

MindSync for DOS v.1.01

This is the original brainwave synchronization program I wrote for DOS and the Sound Blaster. It only generates brainwave files in 22 KHz stereo 8-bit VOC format. Brainwave synchronization is the same as the *i* Wave function in Cool Edit.

Kaleidoscope Screen Saver v.2.8

This is a great screen saver for use with Windows. Lots and lots of options let you customize, or completely randomize the kaleidoscopic variations. It's hard to describe the beauty of the living, flowing artwork that *is* the Kaleidoscope screen saver. Can also change patterns based on incoming music! Revive your old CD collection and actually *see* your music.

RPN Calculator

Reverse Polish Notation calculator (like HP calculators). This calculator has dozens of great functions, including Hex/Binary/Decimal/Octal/n-ary base translations, trig functions, factorial, prime number, exponents and logs, and even biorhythms.

After Dark Screen Savers

I have written three after dark screen savers: Kaleidoscope (scaled down version of the Windows counterpart), Organic (fractal designs that look like cells), and String Arts (colorful designs resembling string art -- a popular art form of the 60's and 70's where colored string is strung between nails to make beautiful patterns).

Wallpaper Maker

Create colorful wallpaper 'tiles' that look superb when tiled as Windows wallpaper. Never draws the same tile twice. See patterns in the tiles that are only visible when viewed as a group of tiles.

Classic Collection

Various DOS based graphic programs. If you're looking for some *classic* programs that are fun to use and watch, these may interest you. Use your mouse to control birth and death rates in a Game of Life, or place wagers on which species will win out in a different life simulation. Steer your way through a warp speed starfield simulation. Experiment with Iterated Function Systems to grow plants and trees. Weave your way through the Mandelbrot set (yes, I know, there are thousands of Mandelbrot generators out there now). Shoot marbles around the screen, and experiment with the effects of gravity. Generate screen mazes. Draw spiro designs, watch string art designs, or point set fractal art. Even includes a character based 'snow' program. Not much documentation, but most programs include the '/h' or '/help' switch for help.

Package Contents

The following files comprise the complete package for Cool Edit 1.51:

COOL.EXE	Main program
COOL.HLP	This help file
ADPCM.FLT	File filter to read and write Microsoft ADPCM *.WAV files
AIF.FLT	File filter for Apple AIFF files
AU.FLT	File filter for Next/Sun files
COOLACM.FLT	File filter for Microsoft ACM formats
DVI.FLT	File filter for DVI/IMA ADPCM *.WAV files
PCM.FLT	File filter for raw PCM files
SAM.FLT	File filter for 8-bit signed audio
SMP.FLT	File filter for SampleVision files
TEXT.FLT	File filter for ASCII text sample data
VOC.FLT	File filter for standard SB Voice files
VOX.FLT	File filter for Dialogic ADPCM 8KHz files
VOX6K.FLT	File filter for Dialogic ADPCM 6KHz files
WAVEAU.FLT	File filter for CCITT mu-law and A-law *.WAV files
WAVEPCM.FLT	File filter for PCM *.WAV files
AMPLIFY.XFM	Amplify transform
CHAMBER.XFM	Echo Chamber transform
CHANMIX.XFM	Channel Mixer transform
COMPRESS.XFM	Compressor/Expander transform
DELAY.XFM	Delay transform
DISTORT.XFM	Distort transform
DTMF.XFM	DTMF tones generator
ECHO.XFM	Echo transform
ENVELOPE.XFM	Envelope transform
EQUALIZE.XFM	Quick Filter transform
FILTER.XFM	Filter transform
FLANGE.XFM	Flange transform
NOISE.XFM	Noise generator
NONOISE.XFM	Noise reduction transform
NORMAL.XFM	Normalize transform
REVERB.XFM	Reverb transform
STRETCH.XFM	Stretch transform
TONES.XFM	Tones generator
WAVESYNC.XFM	Brainwave Synchronizer transform
CTL3D.DLL	DLL 3-D looking controls
README.TXT	A readme file.*.SCP Various script files

Version History

1.51

File open/save dialogs contain MRU (most recently used) list of directories.

CD Tracks can be chosen by typing the track number (great for CD's with more tracks than can be displayed at one time).

Added Zero Crossing function to adjust cursor position or highlight to zero crossings. Use F4 to quickly snap to zero crossing.

Tab functionality added to Cue and Play lists.

Fixed little annoying highlight bug that caused some of the highlight to 'stick' when switching between overlapping Cool Edit windows.

Sped up initial wave display after load.

Added keyboard support for adjusting highlight (Arrow Keys, and SHIFT+Arrow Keys), paging left and right (Page Up, Page Down), and zooming to the beginning or ending of the file (Home and End keys).

Took 'fact' information out of standard Windows .WAV files (wavepcm.flc) for greater compatibility across applications that depend on special formatting of the .WAV file.

Added A-Law, mu-Law, byte-reversed, and unsigned formats to the PCM raw data format. Also added optional header (.DAT) files for specifying format information for raw data.

Added support for Microsoft ACM (Audio Compression Manager) file formats.

Added a 5-bit IMA ADPCM compression for a higher quality ADPCM taking only 25% more space.

Rebuilt record level meters: Display is logarithmic (and labeled) in decibels (dB); Right-mouse-button configuration menu; Clipping indicators; Peak meters easier to see; Meter displays while Playing and Recording as well; Meter spans the entire width of the window for arbitrary precision.

Added user defined undo levels (maximum is 32 levels of undo).

Fixed 'memory hog' problem that was present in the Reverb function.

Quick Help no longer disables the Alt key.

Undo Music was corrupt, it's fixed now.

Drive letter only (eg "D:\") undo directory now works fine - it was causing a 'not enough memory for undo' message erroneously. Also fixed problem when temporary directory specified for a non-existent drive.

Space bar start/stop playing disabled when monitoring audio.

Destination directory always set properly in batch processing, even if the directory is directly typed in instead of using the browse button.

Demodulation added to Generate Tones for demodulated modulated signals.

Made High Quality Convert mode, and Filter function available for Cool Edit Lite users.

Functions will work on entire waveform if no highlight is made.

Moved location of 'fact' chunk in .WAV files to before the data instead of after to be compatible with other wave programs that do not expect this chunk to come after the data.

New password generator being used that is case and 'space' insensitive in the user name field - previously the user name had to be typed in *exactly*.

1.50

Application startup time sped up for owners of CD-ROM drives by not querying the CD-ROM every time the app is started.

Clicking on the 'black' area of the green slider bar will page the display left or right.

Cue items may be added to the cue list while audio is playing (F8 will also add the current cursor location or highlight to the cue list).

Cue list markers and ranges are displayed as blue brackets, or red arrows respectively indicating their locations within the wave.

Current play time is displayed as a wave is played now.

New 'Merge' button added to cue list to merge any two cues into a range (hold down on CTRL to select more than one cue entry).

Added Open Append, and opening of multiple files simultaneously.

Add 'Hand' icon to Graphs to more easily move graph points.

Fixed bug in 8-bit 'dither' conversion from 16-bit that occurred only with clipped audio.

Made separate Undo temporary directory so the undo buffer can be on a separate hard drive if necessary.

Added multi-level undo, instead of just one level of undo.
Convert Sample Type is now undoable.
Added a maximum display size when loading, so the initial display can be limited to a few seconds when a large (several minute) file is loaded.
Added option to keep Cool Edit from writing extra RIFF .WAV information in Options->Info.
Fixed bug in single channel (of a stereo wave) Filtering and Noise Reduction.
Enhanced Noise Reduction function by allowing more FFT sizes.
Fixed bug in Noise Reduction that caused it to stop processing when 99% complete.
Added 6000Hz VOX file format.
Added new Reverb function.
Added Length setting to Stretch function.

1.34b

Fixed EQ presets for Echo function, and Stretch presets.
VOX file format will prompt "Convert to 8000Hz...?" instead of auto-converting.
Fixed bug that caused GP Fault when cursor was out of view and Play was pressed in Play From Cursor mode.

1.34a

Adjusted play bar speed for SBPro cards to be (hopefully) the right speed now for certain wave formats.
Fixed 8-bit stretch function (it was truncating all waves to only the lower half).
Fixed stretching of waves near clipping so distortion does not occur.
Fixed AU file filter so it works properly for saving 8-bit audio, and so it will save properly instead of just writing out the header at times.
Fixed Stretch so choosing an overlap of zero percent will not get errors.
Fixed Quick Filter volume locking and presets (presets were saving as the inverse of what was displayed).
Fixed possible problems with Undo using multiple instances where Undo in one instance would corrupt another instance's undo buffer.
Fixed playing on Sound Blaster cards to prevent "Divide By Zero" errors.

1.34

Custom effects modules (the *.XFM files). See Effects Modules API to build your own!
Last used settings for practically every dialog is now remembered between program runs in the [Profile] section of COOL.INI.
Repeat Last Command function by using F2 (with dialog) and F3 (immediate with previous settings).
Added variable quality sample type conversion.
Added a pause button to pause during playing or recording.
Added a fully configurable distortion effect for getting grunge and blown speaker sounds.
Added a three dimensional echo/reverb effect to simulate acoustics in rooms of any size.
Added a Generate DTMF tones function for making telephone signals.
Added a batch processor to the scripts for running a script on multiple files.
Added new file formats for Next/Sun (.AU), Raw 8-bit signed (.SAM), SampleVision (.SMP), ASCII Text (.TXT), Dialog ADPCM for voice (.VOX), and A-Law/mu-Law for Waves (.WAV) (see Wave Formats).
Auto-play and file format info features added to File-Open dialogs to play sounds through installed sound drivers for preview and view the file format before opening.
Moved file filter options selection to Save As dialog instead of always displaying options when saving.
Single channel editing possible for most functions (that do not change size of waveform).
Improved noise reduction speed for stereo waveforms and added more options. Also displays a noise profile for visual inspection.
Sped up display times in most areas.
Sped up Normalize, Amplify and Envelope on slower machines by integerizing routines.
Added Pause button to progress meter so you can free your system in an emergency without losing your edits.
Broke out Normalize and Invert into separate functional units. Normalize can be used in a script now.
Added decibel scales and logarithmic fades to Amplify.

Modified brainwave synchronizer to have separate high and low settings for intensity and centering.
Generate Tones now has separate initial and final modulation parameters and independently adjustable frequency multipliers that can also vary over time.
Improved reliability of Monitor Source.
New settings options for temporary drive.
Play may begin from cursor or from left edge of screen.
Frequency analysis window displays information in stereo with left=Cyan and right=Magenta.
CD Player controls improved, now you can Insert as well as Eject. Also handles over 10,000 CD titles and song lists. That should be enough for anybody! Also, the IDs used are compatible with the MUSICBOX.INI file, and any CD titles in MUSICBOX.INI will be displayed by Cool Edit.
Added auto zoom button to cue list so double-clicking on cue item opens view to just that cue range, or zooms into the cue point.
Zoom In without any selection zooms in at cursor now, and highlight does not go away when zooming in.
Wave files may be up to 1 Gig in size, which should be sufficient for most applications.
Open and Save As can be recorded into a script for creating more complex scripts that use temporary files.
Current file size and number of samples updated while recording.
Time accuracy increased to 3 decimal digits, and new Frames format added (use Settings to change number of frames per second).
Uses fewer resources per instance, so more instances may be open at once.
Right click extends selection (a shortcut to using shift+left click).
Improved stretch function (preserving tempo or preserving none) by using a better interpolation method.
Added new Fractional Interval Overlap elongation method to stretch (preserving tempo or preserving pitch).
Notes are turned off now when you are not listening to a Music preview.
Filtering has been improved/fixed to be more accurate with less distortion.
Fixed bug in finding proper file format of a waveform (that was a tricky one!).
Fixed bug where Cool Edit crashed when loading in NT while a CD was in the drive.
Progress bar updates ETA time more often, at shorter intervals.
File types added to Extensions section so programs like File Manager have Cool Edit associated with all formats Cool Edit understands (unless format was otherwise associated with a different program).
Progress meter updates ETA time more often for a more accurate result.
Lots and lots of other little tweaks, enhancements, and elbow grease went into this version as well.

1.33

Fixed bug in Filter Transition from initial to final settings... Final frequencies were off.
Frequency analysis calculates fundamental frequency much more accurately.
Added Compressor function to compress/expand dynamic range.
All graphs have double-clickable points to edit the graphical input point directly.
Clicking on any of the graph points displays the point's values below, instead of just displaying the mouse position.
Ring Modulation possible by using "Modulate over Source" checkbox. Instead of generating sine waves, the source is modulated by a the sine wave (or any waveform you choose).
Any type of Modulation possible by using Paste Special's Modulate option.
Noise reduction function added.
Sped up some stereo operations for Filter, Spectral View, and Noise Reduction.
Added smoother 16-bit to 8-bit conversions when copying 16-bit files into an 8-bit waveform.

1.32

Filter has adjustable size (previously used 1024, now smoother filtering can be done by using sizes of 2048 or 4096).

1.31f

References added to the Brainwave Synchronization help.
Cue list and Play list remember their positions better when cutting and pasting shifts their positions.

1.31c-1.31e

Larger files list display.

FFT filter much smoother now (previously had some high frequency noise when doing extremely narrow sharp filters).

Adjustable spatial separation for Generate Noise.

New "smooth" option added to Wave brainwave synchronizer for an alternative "feel" to the sounds. Also the "gamma" range (frequencies above 200 Hz) has been noted.

New independent left and right meters for Monitor Source, as well as peak indicators.

Clicking on the Cool window in the waveform area will **not** move cursor location if Cool Edit is not the active application.

New display for stereo waveforms, plus a more precise and consistent display for all waveforms.

Toolbar buttons draw a little quicker now.

Flange and Special EFX have been combined into a new more powerful Flanger, with a Special Efx option.

1.31b

Space bar will act as a 'play' button.

Warning issued when loading a file takes up ALL available temporary drive space.

Pasting low sample rate waves into high sample rate waves works properly now.

1.31a

Fixed Stretching to less than 50% for preservation modes.

Fixed displaying of waves more than 15 megs in size.

Sped up music generation by using temporary files for each note. Long music files can be generated up to 10 times faster, or even more!

Added Open As to open waveforms into any format desired.

Added Select Entire Wave to quickly select the entire waveform, even if the view is zoomed in.

The Paste and Paste Special functions now have a progress bar like the rest.

1.31

Added the Scripts feature, which allows Cool to remember everything you do, and will let you replay it.

One script file can contain many individual scripts.

Wasn't loading from command line properly before.

Fixed memory hog/crashing bug in Generate Tones.

Added customizable Toolbar by modifying cool.ini file.

Added play bar to see what part of the wave is currently being played

Added **Autocue** button to Play List to play a single play list item at a time.

Trim is now undoable. Please turn off the Undo feature whenever you are doing really really large operations if you want Cool to run faster, since it will not have to save Undo information.

Added Save Selection to save the currently highlighted portion of the wave to disk.

Improved method for stretching while preserving pitch or tempo for cleaner sounding waves called

Interval Overlap. See Stretch.

Improved 'time left' reporting when recording.

Increased maximum file size from 248 megs to 536 megs.

1.30b

Yah, I know... There weren't supposed to be any more bugs. But I found some:

Using FFT functions sometimes crashed (like FFT, Frequency Analysis, Spectral View, or Music). That has been fixed!

Undo paste when Hilight After Paste was not checked didn't work. It does now.

File filters have been modified to allow any format to be broken up into multiple files. For example, the WAV file format can have several file filters for different compression schemes.

IMA/DVI ADPCM compression filter added to package (dvi.flr), which allows compressing of 16-bit audio down to 4, 3, and even 2-bit (with noticeable loss in quality at 2-bit).

1.30a

Found more bugs: fixed Undo for Paste (special/loop/regular), and Undo Delete.
Fixed Re-Open on PCM files (would hang).
Added Undo for Music.
Added presets for Envelope and Filter.
Added initial/final settings for overtones in Generate Tones.
Revised Music dialog by adding Constant Duration checkbox to replace Adjust Duration.

1.30

Find Frequency is now a graph of the frequency spectrum.
Fixed the remaining known bugs, and completed spec for building custom file filters.

1.29b

Custom file filters (WAV, VOC, PCM, and AIFF currently). See [File Filters API](#) to build your own!

1.29a

Selectable wave devices (handles multiple sound cards).

1.29

Edit Left/Edit Right for cutting, pasting, and copying only one channel of a stereo wave.
Adjust sample rate added.

1.28c

Equalizer changed to Quick Filter, since FFT Filter does true, precise filtering.
Graphs have more flexibility with points.
Cue ranges can be associated with keyboard keys, so waves can be played from the keyboard.

1.28a

[Spectral View](#) and [FFT filtering](#) added.

1.28

File -> Close added.
Customizable presets added to a variety of wave functions.

1.27

Double-clicking on slider bar allows entry of starting and ending samples.
Pre-compute trig tables added to Settings.
Undo function added.

Versions 1.03 to 1.26

Cool-Edit is sizeable. You can make your window big or little or whatever.
Wave files may be Dropped into the editor from the File Manager, or other applications supporting Drag and Drop.
New Instance added to File Menu
Added automatic conversion of 16-bit audio to 8-bit while playing for boards that do not support 16-bit audio.
Fixed a major bug having to do with the progress meter. You can now go on to do other things while Cool works in the background on long operations.
Fixed an annoying bug when working with large files that caused the green slider to work inappropriately.
When opening multiple instances of the same wave, different names are given to each instance.
New and Improved Pink noise source. (Ideal for use with the Wave function).
[Cue list](#) and [Play list](#) supported for looping and custom play order.
New Setting for viewing mode of Dots (original) or Lines (dots of sample are connected by lines).
Digital [Delay](#) transformation added to easily spatially locate sounds to the left or right, or for special effects.
New settings option for Save/Save As interpretation of the diskette icon.
Are You Sure? Dialog box added to Silence button.

New waves can be made at any valid sampling rate, by use of the new **Custom** option.

DC Bias Filter added to Amplify to adjust waves skewed by a voltage.

CD Player has been added to play CD's for recording. Song titles may also be entered and restored automatically on future playbacks.

Brainwave frequency (Wave) function has adjustable graph for easier input of multiple frequencies over time.

Sound Blaster VOC file compatibility was added. Not FULL capability yet, because it does not support loops (it only loads the loop once), and ASCII text. It *does* however support Sound Blaster Pro formats for 44.1 KHz-mono and 22 KHz-stereo. It will also load non-standard sample rates if the VOC file was recorded at one.

Echo Equalizer added to Echo function to have each successive echo equalized for truer echo effects, or just snazzy special effects.

The Listen option was added to the music dialog so that the notes could be played through your MIDI setup as a preview. You can record from the music played as well (if your board supports recording from the on-board synth.)

Envelope option was added to have more control over amplifying waves by using an amplification envelope.

A few new tone flavors have been added for making more natural sounding instrument sounds, which can be used with the Music option.

New options in settings dialog for play/record buffer size, and highlighting after pasting.

Color DIBs can be added to the .WAV file format.

Equalizer settings changed to decibels instead of percentages, making it easier to equalize, since the logarithmic scale is more natural.

Monitor Source meter is in much better *real time* since it now does not depend on the selected buffer size.

Versions 0.5 to 1.02

New algorithm implemented for raising/lowering pitches, and speeding up/slowing down waves.

Preset buttons added to Stretch dialog.

Highlighting is faster for slower video cards.

You can use the Shift+Mouse Button to extend the highlighted selection.

Music option was added to put your favorite samples to music (as if you were playing them on a keyboard) using a very easy to use music editor. Only short riffs are supported right now.

Stretching is cleaner sounding.

The Transpose function has been added to the Stretch option to musically raise or lower pitches of selected samples.

A linked list approach has been taken for working with the temporary disk file when editing. Now Inserts and Deletes are almost instantaneous when working with very large files. (Fixed cut/paste bug from 1.01)

The RAW file format was added for saving only the wave data without any headers.

Edit keys Delete (Delete Wave Selection), Ctrl+Insert (Insert), Shift+Insert (Copy) and Shift+Delete (Paste) were added.

Toolbar was added for quicker access to functions.

The previous Save menu item was changed to Save As, and a new Save was added to save the wave currently being worked on without being asked for the file name.

The Silence option was added to the Transform menu to quickly silence the selection portion.

Extra RIFF information embedded in the .WAV file is now remembered between saves, and can be edited using the Info dialog.

MS ADPCM Compression is supported for loading and saving compressed files now.

A slight bug in overlapping was fixed. Overlap pasting past the end of a wave that used to be longer would revive data past the end of file.

The Stretch transformation was added to allow stretching and compressing of waves, adjusting pitches, or adjusting tempos.

Viewing mode of the Beginning and Ending samples, as well as the time window (which displays how long the viewed portion of the sample is, or how long the selection is) can be changed between samples and time by double clicking on the display.

Generating silence in 8-bit mode works properly now, centering the silence line at value 128.

You may now associate the COOL.EXE program with .WAV files in the file manager. Command line loading works. File will automatically play once loaded if another file is not already playing.

Little Yellow Arrows work on files larger than 30,000,000 samples.

Gliding waves from one frequency to another "overshot" before. For example, gliding from 12Hz down to 7Hz would actually glide down twice the frequency, (instead of -5Hz, -10Hz) resulting in a glide from 12Hz down to 2Hz instead. It's fixed now.

Multiple COOL's can be opened at once (which reduced the executable as well) for working with multiple waves at once. It's very easy to copy and paste between waves, even of different formats.

Also, waveform display is instantaneous once loaded in (for example, restoring a minimized long waveform displays instantly instead of recalculating). This allows such effects as the sound level meter running while zooming in and out of a waveform.

Displaying long waveforms now "multitasks" with Windows, so other things can be done while it is loading.

The progress meter was fixed slightly to allow multitasking with other applications without crashing Windows.

When saving a file, the extension .WAV is assumed if no extension is typed.

8-bit stereo pasting (with Overlap) works properly now (slight bug caused screeching problems). Also, overlap pasting past end of file works properly too.

A "Musical Source" checkbox was added to the brainwave transformation to eliminate clicks heard when waving musical files.

A Recording level meter was added. Source input can be monitored through the Options menu for setting recording levels.

Lifting up on button after dragging a selection (and leaving the main window) will now act properly, and disengage the selection process

Recording to Sound Blaster Pro works properly. It no longer gives errors and chops up recording. In fact, using COOL to record does a much nicer job than the recorder program provided with Sound Blaster (at least mine), and there are no "clicks" or other distractions to contend with.

Using "Large Fonts" on some monitors correctly displays main console, as well as progress meter now.

Thanks go out to:

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Other Cool Fans who made suggestions and found bugs here and there.

All those who downloaded Cools 1.05 to 1.50

Other Acknowledgments

Cool Edit uses Microsoft Corporation's CTL3D.DLL dialog library, which Microsoft has made publicly available to be used without restrictions. Please note that Microsoft does not support this program in any way, and are not liable for anything that may happen.

All format types have been reproduced from publicly available documentation. I make no claims that they are in any way correct, and as such, may not be compatible with other interpretations of the same format. It is up to you to first test to see that when saving audio files in a particular format, that the file generated is compatible with other systems that read the same format.

What Does the Future Hold?

We plan to keep on upgrading Cool Edit, making it more easy to use and useful to more people. We have some definite plans for the next version and we're sure you'll like them! Suggestions for future versions are always welcomed.

Common Procedures (How To...)

If you are asking "How do I...?" then you should find (some of) the answers here! (Even though it is a terribly small list - and it hasn't grown for several versions now).

HOW DO I:

Convert cleanly from one sample rate to another

Open a file for editing?

Scroll wave left or right?

Create sound effects?

Mix multiple waves together?

Copy a Mono wave to only one channel of a stereo wave?

Spatially locate a mono sound to the left or right?

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Opening Waveform Files

There are a number of ways to get a waveform into Cool-Edit for editing.

- Choose **File -> Open** and find the wave file from the dialog box.
- Double-click on a waveform file from the File Manager (if you have chosen COOL as your editor of choice from the **File -> Associate** dialog).
- Click on a waveform file, or multiple files, in the File Manager, and drag them into Cool-Edit.
- Copy a waveform from any wave editing program, then choose the **Edit -> Paste** option, or press Shift+Insert while in Cool-Edit.
- Create a new Cool-Edit program item in the Program Manager. In the Command Line field, enter the path to the wave file to open after the program executable name. When this icon is double-clicked, Cool-Edit will automatically load the wave file you specified in the command line, and play it.
- Choose **File -> New** to create a new waveform of the appropriate sample rate. You may now press **Record** to record a new wave to be edited.

Converting Sample Rates

With version 1.51, the best method for converting sample rates is to use the Convert Sample Type function. This function incorporates the methods outlined below and performs them automatically.

There are various methods for converting sample rates. One way is to simply copy the waveform you wish to convert, open a new instance of Cool, and say New. Choose the new sample rate, bit rate, and so on. Then Paste. When Cool Pastes samples of a different rate, it converts by doing a linear interpolation of the points in-between when upsampling (converting to a higher sample rate), and it just picks out the samples needed when downsampling. This method is quick and dirty, which means it is fast, but is not the cleanest method, and you will get some artifacts and "ringing" depending on the sample.

A slightly cleaner method is to use the Stretch function (stretch To cleanly convert, you must run a filter over the sample, and use the Stretch function (in Preserve Neither mode). If you are downsampling (e.g. 22 KHz to 11 KHz), you would compress the wave to the ratio of the destination sample rate over the source sample rate (e.g. 11 KHz / 22 KHz = 0.5 = 50%). Then you would adjust the sample rate (Edit->Adjust Sample Rate) to your new rate, keeping the bit size (8 or 16), and the number of channels the same. The stretch function uses a weighted average approach when downsampling, which is cleaner than just pasting the sample in.

The best method, for downsampling or upsampling without noticing any difference in quality (in fact, upsampled waves sound better than the original at a lower sample rate), you must use the filter, which can be very slow. The rest of this article explains how to use the Filter and Stretch functions to upsample and downsample.

If you are downsampling (converting from a higher sample rate to a lower one), you must Filter the wave first to eliminate any frequencies above the destination rate's Nyquist frequency. The Nyquist frequency is a frequency exactly one half the sample rate. For example, to convert a 44.1 KHz sample to 22 KHz, you must filter out all frequencies above 11 KHz. To do this, choose Filter. Select 'Passive' as the filter type, and make the graph such that all frequencies above one half the destination's sample rate are set to 0%, and all frequencies below are set to 100%. If you are cutting the rate exactly in half, as in our example, you may use the pre-defined filter "Low Pass 1/2", which does just this.

Once filtered, use the Stretch function to compress the wave. The ratio of compression is equal to $100 * \text{destination_sample_rate} / \text{current_sample_rate}$. In our example, this would be 50. If you were downsampling from 44.1 KHz to 11 KHz, this would be 25, or 32 KHz to 22 KHz, this would be about 68.8 (68.745% for 32075, or 68.906 for 32000).

Finally, choose Edit->Adjust Sample Rate to tell Cool what the new sample rate is.

If you are upsampling (converting from a lower rate to a higher one), you must first use the Stretch function. The ration of expansion is equal to $100 * \text{destination_sample_rate} / \text{current_sample_rate}$. For example, if we are converting from 22 KHz to 44.1 KHz, this would be 200. The stretching will do a weighted average, which results in a linear interpolation of the samples to generate the "in between" samples. This interpolation adds a lot of high frequency noise, which can simply be filtered out in the last step.

Change sample rates by choosing Edit->Adjust Sample Rate to tell Cool what the new sample rate is. This should be done before filtering, so the filter knows what the current sample rate is.

Choose Filter to filter out all frequencies above the Original wave's Nyquist frequency (or one half the original sample rate). In our example, we would filter out everything above 11 KHz, since that is one half of 22 KHz, or original sample rate. All frequencies below 11 KHz should be set to 100%, while all frequencies above 11 KHz should be set to 0%. If you were upsampling from 8 KHz, you would set all frequencies below 4 KHz to 100% and above 4 KHz to 0%. The filtering will take the linearly interpolated

samples, and smooth out all the jagged edges. The result is a sample that now sounds better than it did at its original sample rate.

Your sample should now be cleanly converted to the new sample rate, and sound spectacular, with no ringing, or noise artifacts.

Mono to Stereo Conversion

Use the Edit->Convert Sample Type function to convert Mono to Stereo. There are other methods that work as well.

You can copy the wave at it's current volume directly to one channel or the other.

- Copy the wave in question to the clipboard by highlighting it and choosing **Edit -> Copy**.
- Go to the stereo wave in question, and uncheck **Edit Left** or **Edit Right** in the **Edit** menu. Now paste the wave with **Edit -> Paste**. The wave will be pasted to the channel that remained checked in the menu.

or

If you wish to place separate waveforms on each channel of a stereo wave and mix at different volume levels, you can use the **Edit -> Paste Special** option.

- Choose **File -> New** and create a new stereo wave of the sample rate you wish.
- Open a new instance of the program, and open the mono wave you want to place on the left channel. If you want to place a stereo wave, you may wish to use the Channel Mixer to mix both channels at 50% first.
- Highlight the section you wish to place on the left channel, and choose **Edit -> Copy**.
- Select the new blank stereo waveform, and choose **Edit -> Paste Special**. Make sure Overlap is checked, looping is turned off, Lock L/R is turned off, and the left volume is 100% while the right volume is at 0%.
- To do the right channel, copy the section as you did for the left.
- Place the cursor at the start of the new waveform, and choose **Edit -> Paste Special**. Change the volume levels so that the left volume is 0% and the right volume is 100%.

Spatial Location

Spatially locating a mono sound makes the sound appear as if it is coming from the left or the right when listened to with stereo headphones, even though the actual volume levels for left and right of the wave are identical. What happens is one channel is delayed a few milliseconds. When a sound reaches one ear a few milliseconds before reaching the other ear, the brain interprets the delay as hearing the sound coming from the direction of the non-delayed signal.

There are two functions that can do this: **Delay** and **Wave**. Use **Delay** to place a sound anywhere from left to right, and **Wave** if you wish to have the sound source oscillate between left and right.

- First make sure the signal to be delayed is mono in stereo format. That is, both left and right channels are identical. With a mono source, copy the wave, create a new stereo wave, and paste. With a stereo source, use the channel mixer so that both channels get mixed at 50% for each channel.
- Highlight the entire selection you wish to spatially locate. Choose **Transform -> Delay** or click the stopwatch icon. Choose a delay of up to 2 ms. Delaying one channel will make the sound appear as if it is coming from the other channel. You can use the preset for Spatial Left or Spatial Right as well.
- If you want the source to move left to right, choose the **Transform -> Wave** function. Enter the cycles per second (e.g. 0.1 = left to right and back in 10 seconds) in the initial and final boxes (the same value for both boxes). Choose the center position (in the center, left, or right) using the Centering control. Choose an intensity of about 50 or so, greater intensities mean the sound goes further left or right off of center. Also check the Musical Source box if your sound is not noise related.

Creating Sound Effects

Two types of sound effects can be created using Cool-Edit: Noise based, and Tone based. To create a sound effect, you must first generate some noise or tones to base your effect upon. The basic method is to create a few seconds of tones or noise, then use the transformation functions to manipulate the wave and create the desired effect. With these two sound sources, noise and tone, practically any effect can be created.

Noise Effects:

Waterfall, wind, and rain
Thunder, snare drum, cymbals, jet engines
Fantasy sounds such as time tunnel vortex, etc.

Tone Effects:

Siren, pipe organ, piano, and other musical instruments
Space ship sounds, whining, whistles, etc.

- First generate a few seconds of noise or tones using the **Generate** functions.
- Experiment with the different settings (e.g. white noise, pink noise, overtones, etc).
- Add some silence to the end of the sample by clicking the cursor at the end of the wave, and choosing **Generate -> Silence**. This will give some room for transformations that "bleed over" such as echo.
- Try effects such as **Flanging**, **Filter**, and **Quick Filter** with noise sources, or **Stretch** and **Echo** with tone sources.
- Try reversing, or copying and loop pasting portions of the wave.
- The possibilities for sound effects are endless, and is only limited to your imagination!

Function Presets

Many of the functions have presets available for easily recalling your favorite settings. New presets can be added at any time. All preset information is saved in the file cool.ini, usually in your Windows directory.

Double-Clicking on any preset will instantly set all controls in the dialog box to that preset.

Whenever you have settings you would like to keep, you may enter name for your settings, and press the **Add** button. Be careful though, since there is no rule against you adding two presets with the same name. This can get confusing if the presets are different.

To remove a preset from the list, choose the preset, and press **Del**.

To modify an existing preset, **double-click** on the preset name, make your modifications, then press **Del** immediately followed by **Add**. This will delete the old preset and add your current settings in under the same name.

If all presets are removed, the default presets will automatically be reloaded.

If you are finding that your presets are not being saved the next time you use the function, check to see that the cool.ini file is not more than 64K in size. This should never happen, but it may be possible if you have hundreds and hundreds of presets.

If you obtained a new version of Cool Edit, and find that your presets have disappeared or double-clicking on them brings up something mysterious and not the settings you wanted, then I apologize. In making new versions it is sometimes necessary to change the order of entries in the INI file to make room for new items, etc. Just remove all your presets to get the Cool Edit defaults and start from there. Again, if this has happened to you, I'm sorry.

File Filters API

File filters (the *.FLT files you see bundled with Cool Edit) are actually separate DLLs (Dynamic Link Libraries) that contain the code necessary to read and write various file formats. If you are interested in writing your own file formats, let me know and I will send you a complete description of the File Filters API along with a sample file format. This application and the file filters were written in C (not C++) using Microsoft Visual C/C++ 1.5.

If you have written file filters in the past, and you find Cool Edit 1.34 does not recognize them, then you must change the valid library number from 1967 to 134 and recompile.

Effects Modules API

Transform and Generate effects modules (the *.XFM files you see bundled with Cool Edit) are actually separate DLLs (Dynamic Link Libraries) that contain the code necessary to perform the effect. If you are interested in writing your own effects modules, let me know and I will send you a complete description of the Transform and Generate Effects API along with a sample transform module. This application and the effects modules were written in C (not C++) using Microsoft Visual C/C++ 1.5.

Undo

If the Undo function is enabled, (see [settings](#)), you will be able to back up one step if you make a mistake. Undo information is automatically saved in a temporary file called ~NDO`nnnn` in your temporary directory before you do anything that will change the waveform being edited. This could slow down wave operations on very large waves. The temporary directory used for undo information can differ from the standard temporary directory for Cool Edit, and it is specified in the Settings dialog.

You will be warned if there is not enough disk space to save the Undo information before continuing any wave operation. Up to three levels of undo are retained so you can undo the last three actions.

Press Alt+Backspace to quickly undo the last operation.

Closing a Waveform

Close will close the wave currently being edited, and show the initial startup information in the status boxes.

When a waveform is closed, the associated temporary file is removed, thus freeing up the hard disk space that was being used previously.

Scrolling Wave

The green and/or black bar above the waveform indicates which portion of the entire wave is being viewed at that moment. When **Zoom In** is chosen, the bar gets smaller, since the portion being viewed with respect to the entire wave is smaller.

You may click, and drag the green bar at any time to scroll the portion being viewed left or right at any time. Clicking in the black region outside the green bar will scroll the bar exactly one screen to the left or right.

Double-clicking in the slider bar region will bring up the Viewing Samples text entry box, to type in starting and ending samples.

Mixing Waves

Waves of any sample type can be mixed together into a resultant wave of any sample type. This means you can mix a mono 8-bit 11 KHz wave together with a Stereo 16-bit 22 KHz wave without any problems.

- Create a new instance of the program by choosing **File -> New Instance**.
- Choose **File -> New** in the new instance to choose your final sample type. You may generate a few seconds of silence to give yourself a workspace.
- Mix all your waves into the new instance one at a time by highlighting the portion you wish to mix, and selecting **Edit -> Copy**. You may create more new instances and open waves into them if you wish.
- Position the cursor where you wish to place the start of the copied waveform, and choose **Edit -> Paste Special**. Make sure the Overlap box is checked. Choose the mixing volume, and press OK.
- Do this for each wave you wish to mix.

Edit menu

The file menu displays all the waveform editing options.

Edit Menu Options:

Undo

Repeat Last Command

Copy

Cut

Paste

Paste Special

Select Entire Wave

Edit Left

Edit Right

Delete Selection

Trim

Viewing Range...

Adjust Sample Rate...

Byte Swap...

Spectral View / Waveform View

Convert Sample Type

Amplify
Channel Mixer
Compressor
Delay
Distort
Echo Chamber
Echo
Envelope
Filter
Flange
Invert
Music
Noise Reduction
Normalize
Quick Filter
Reverse
Reverb
Stretch
Wave

Transform menu

The Transform menu displays all the waveform transformation options. Waveform transforms are encapsulated in the .XFM files.

Transform Menu Options:

Reverse

Silence

Invert

Music

The following are embodied in XFM files, and may change at any time. If the function you need is not listed, go to the function directly and view it's help information

Quick Filter **

Amplify

Channel Mixer

Compressor **

Delay **

Distort **

Echo **

Echo Chamber **

Envelope **

Filter

Flange **

Noise Reduction **

Normalize

Reverb **

Stretch

Wave

**** Function not available in Lite version.**

Generate menu

The Generate menu displays all the waveform generation options. Generate functions are encapsulated in .XFM files.

Generate Menu Options:

Generate Silence

The following are embodied in XFM files, and may change at any time. If the function you need is not listed, go to the function directly and view it's help information

Generate Noise

Generate Tones

Generate DTMF Signals **

**** Function not available in Lite version.**

Options menu

The Options menu displays various additional options.

Options Menu Options:

Loop / Play Toggle

Level Meters

Settings...

Info...

Frequency Analysis...

CD Player

Cue List

Play List

Cool Scripts...

Select Wave Device

The **View Indicator** is the green bar above the waveform being viewed which changes size depending on the level of Zoom.

The **Toolbar** is the row of icons just below the menu for quickly accessing commonly used functions.

Quick Help messages appear below the toolbar to give a brief description of the function associated with the tool icon.

Cool Edit Quick Reference

Navigating the Main Screen

Keyboard and Mouse Shortcuts

File Functions

Editing Functions

Transformation (Effects) Functions

Audio Generation Functions

Options Menu

Updated Features

Navigating Cool Edit

Commonly used functions are represented as icons in the **Toolbar**.

- Just holding the mouse over any of these icons will bring up a Quick Help message describing the function in more detail.

Waveforms are displayed in green.

- Any portion may be selected by clicking on the waveform, and dragging left or right.
- Holding the **SHIFT** key while doing so will increase the currently selected portion.
- **Double-Clicking** on the waveform will select the entire visible wave.

Choose **Zoom In** to expand the selected portion to full width. **Zoom Out** will give a larger view of the waveform, while **Full View** will display the entire waveform in the workspace.

The **Play** button will play the portion of the wave that is currently being viewed, or the portion that is highlighted.

The **Pause** button will temporarily pause the playback or recording of audio. The button turns into a **Continue** button when audio is paused. If recording, the red record bar turns yellow to indicate a paused state.

The **Record** button will start recording at the current insertion point. Any waveform data after that will be recorded over.

Use the **Stop** button to end waveform playback or recording.

The **Record Level Meter** below the Play/Record buttons will display in real-time the current peak amplitude of the audio being monitored, recorded, or played.

- **Double-Clicking** on the Record Level Meter will start and stop monitoring.
- **Right-Clicking** the display will bring up the Level Meter configuration menu.

The View Indicator depicts which portion of the entire waveform is being viewed in the workspace below.

- **Sliding** the green bar (when present) left or right scrolls various parts of the waveform below into view accordingly.
- **Clicking** to the left or right of the green bar will scroll the audio to the left or right one screen when zoomed.
- **Double-Clicking** on the View Indicator will bring up a Quick View screen to allow direct entry of starting and ending samples for the viewing range.

The **Yellow Arrows** indicate the point of insertion. Clicking anywhere in the workspace will move the point of insertion to the mouse.

The **Red Arrows**, when present, represent the Cue List entries for a cue marker.

The **Blue Brackets**, when present, represent the Cue List entries for a selection range.

The **Time Display** windows show the current Starting and Ending points of the current selection, or portion being viewed.

- **Double-Clicking** on these windows will toggle the display between time in seconds, and in samples.

The **Wave Format** window displays the format of the wave in Channels, Sample Rate, and Bits Per Sample.

- **Double-Clicking** on this window will bring up the Adjust Sample Rate dialog to change the waveform's characteristics.

When a wave is being played via the **Play** button, or automatically via a keypress (see Cue List) a vertical bar shows the current playing position.

If a valid audio CD is in the CD-ROM drive, and the [MCI] CD Audio extensions are loaded, pressing the CD icon will display the CD Player Controls at the bottom of the window.

Shortcuts

KEYBOARD

Delete	Delete
CTRL+C	Copy
CTRL+Insert	Copy
CTRL+X	Cut
SHIFT+Delete	Cut
CTRL+V	Paste
SHIFT+Insert	Paste
CTRL+T	Trim
Home	Shift view to the leftmost portion of the waveform
End	Shift view to the rightmost portion of the waveform
Page Up	Shift view one screen to the left
Page Down	Shift view one screen to the right
Left Arrow	Nudge left side of highlight to the left
Right Arrow	Nudge left side of highlight to the left
SHIFT+Left	Nudge right side of highlight to the left
SHIFT+Right	Nudge right side of highlight to the right
ALT+Bksp	Undo
CTRL+A	Select Entire Wave
CTRL+S	Select currently viewed portion
ESC	Cancel selection
ALT+P	Play waveform
ALT+S	Stop play
SPACE	Play waveform/Stop waveform
F2	Repeat last command (show dialog if command has one)
F3	Repeat last command now (using last used settings)
F4	Zero-cross adjust
F8	Add the current cursor location or highlighted selection to the cue list
ALT+I	Waveform Info
ALT+Z	Frequency Analysis Window

MOUSE

Left Click and Drag on waveform to highlight and select a range of samples
Right Click (and drag) on waveform to extend selection
Shift+Left Click (and drag) on waveform also to extend selection
Double-Click on view indicator (green bar) to enter viewing range directly in samples
Click to the left or right of the view indicator to page one screen left or right
Double-Click on Levels Meter (black bar beneath play buttons) to start/stop monitoring
Click on the Clip Indicator (to the right of the level meter) to clear it
Right Click on the level meter to bring up its configuration menu
Double-Click on Sample type display to change sample type interpretation
Double-Click on time windows to change time format
Double-Click on title bar to Maximize/Restore
Rest mouse over toolbar button to get explanation of button's function

Opening a Waveform

Various waveform data types are supported. Cool Edit will convert the waveform type to its own internal temporary file type for faster editing, and larger edit file size. The wave can subsequently be saved in any of the supported formats you wish.

Checking the **Auto Play** box will cause any wave selected to be played if your system has a driver that understands the format of the wave being highlighted. If using this option causes your audio drivers to crash, then don't use it. Cool Edit requests the system to play the file, and some system drivers do not know how to handle files of various formats. Your system may crash if your Windows audio drivers try to interpret a sound file that is not understood. Getting new sound compression drivers should fix these sorts of problems.

Checking the **Show Info** box will show the waveforms basic information like file format, uncompressed file size, and running time.

Each wave type supported is embodied in a filter (*.FLT) file. See [Wave Formats](#) for descriptions of the various formats.

When opening a wave AS another format, the quick and dirty method is used. See [Open As](#) for more information. To append another waveform to the end of the current one, see [Open Append](#). Multiple waveforms can be opened at once by selecting multiple files in the Open Waveform dialog box. If multiple files are chosen, all file types are converted to that of the first file that is opened (alphabetically the first file in the selection).

Appending a Waveform

Append any waveform to the end of the current waveform. If the waveform being appended is of a different type, it will be quickly converted as it is being copied. It is advised that you only append waveforms of the same sample rate for best results, since no interpolation is done when resampling waveforms of differing sample rates. See [Open](#) for more information on the Auto Play and Show Info options.

Each wave type supported is embodied in a filter (*.FLT) file. See [Wave Formats](#) for descriptions of the various formats.

Wave File Formats

All the waveform file formats that Cool Edit understands are placed in the .FLT files. Each .FLT file can contain a coder and a decoder, and all options necessary for the particular wave format. Sometimes large formats like .WAV are encapsulated in many .FLT files instead of just one. The following is a list of the various file formats for which I have written codecs.

Windows PCM waveform (.WAV)

All WAV formatted files follow the RIFF (Resource Information File Format) specification. Most of the special information (See [Info](#)) is saved with the wave file in these formats. The standard Windows PCM waveform contains PCM coded data, which is pure uncompressed pulse code modulation formatted data.

Microsoft ADPCM waveform (.WAV)

Microsoft ADPCM compressed waveform format consists of 4-bit per channel compressed data. Each 4-bit sample is expanded to 16-bits when loaded. For this reason, it is best to save 16-bit files in this format rather than 8-bit files as the quality will be much greater. In the end, the 16-bit data can still be quickly converted to 8-bit during playback on cards that don't support 16-bit. Choosing the Multiple Pass option will take longer to save, but the quality will be better. The time taken to read an ADPCM file is the same no matter which option you choose.

IMA/DVI ADPCM waveform (.WAV)

This standard compresses 16-bit waves to 4-bit using a different (faster) method than Microsoft ADPCM, and has different distortion characteristics, which may be better, or worse, depending on the original sample being compressed. Again, it is better to save 16-bit audio in this format than 8-bit. This format allows for 3-bit compression as well at a slightly lower quality. Very few sound drivers support the 3-bit ADPCM, and I have found none that actually work properly. In building this format, I followed the specification to the letter without making any assumptions. In the past, I have seen audio drivers that did not play DVI compressed data properly, but lately this has been changing as other manufacturers are providing DVI audio drivers that can read the files saved by Cool Edit in this format just fine. (Although I have yet to see a 3-bit DVI audio driver that plays stereo waves properly.) If you have other software that does not play files saved in this format properly, please contact the vendor and try to obtain the latest driver they have.

I have also implemented a 2-bit and a 5-bit version of the compression by using the index tables {-1, 2, -1, 2} and {-1, -1, -1, -1, -1, -1, -1, -1, 1, 2, 4, 6, 8, 10, 13, 16, -1, -1, -1, -1, -1, -1, -1, -1, 1, 2, 4, 6, 8, 10, 13, 16} respectively. I could not find the specification for these compression ratios, so it may not be compatible with other IMA/DVI 2-bit or 5-bit compressed files. I found that the preceding index tables provided the best quality.

CCITT mu-Law and A-Law waveforms (.WAV)

These formats compresses original 16-bit audio down to 8 bits. The quality is somewhere between 8-bit and 16-bit. Thus, a-law and mu-law encoded waveforms have a higher s/n ration than 8-bit PCM, but at the price of a little more distortion than the original 16-bit audio. The quality is definitely higher than you would get with 4-bit ADPCM formats.

Sound Blaster voice file format (.VOC)

Sound Blaster and Sound Blaster Pro voice files can be loaded. This format only supports 8-bit audio, mono to 44.1 KHz, and stereo to 22 KHz. If the file you are loading contains loops and silence blocks, they will be expanded while loading.

Apple AIFF format (.AIF)

This is Apple's standard wave file format. Cool Edit only supports the PCM encoded data, even though this format (like the WAV format) can contain any one of a number of data formats. Assuming you got the wave over to the PC from a Mac, you can name the file with the .AIF extension, and load it using this file filter. If saving files destined for the Mac in this format, you should add the four character code "AIFF"

in the file's resource fork on the Macintosh side so other Mac programs will know what type of file it is.

ASCII Text format (.TXT)

Data can be read to or written from files in a standard text format, with each sample separated by a carriage return and channels separated by a tab character. Options allow data to be normalized between -1.0 and 1.0, or written out and read in raw sample values. An optional header can be placed before the data. If there is no header text, then the data is assumed to be 16-bit signed decimal integers. The header is formatted as **KEYWORD:value** with the keywords being: SAMPLES, BITSPERSAMPLE, CHANNELS, SAMPLERATE, and NORMALIZED. The values for NORMALIZED are either TRUE or FALSE. For example,

```
SAMPLES: 1582
BITSPERSAMPLE: 16
CHANNELS: 2
SAMPLERATE: 22050
NORMALIZED: FALSE
164 <tab> -1372
492 <tab> -876
...
```

8-bit signed raw format (.SAM)

Data with the .SAM extension is assumed to be 8-bit signed raw data with no header. Sample rate is assumed to be 22050Hz, but the actual sample rate can be changed once loaded using Edit->Adjust Sample Rate. This format is popular for building MOD files, since audio in MOD files is 8-bit signed. Many MOD editors allow samples to be inserted from files, or exported to files in this format.

Next/Sun CCITT mu-Law, A-Law and PCM format (.AU)

The Next/Sun format supports many data formats, but only mu-law, A-law and linear PCM data are supported. The most common use for the AU file format is for compressing 16-bit data to 8-bit mu-law data.

SampleVision format (.SMP)

The SampleVision format only supports mono 16-bit audio. If your data is in a different format, you will be asked if you want to convert it before saving. This format supports loop points, which can be edited using the Cue List. The Label of the cue must be in the format **Loop *n,m*** where *n* is the loop number from 1 to 8, and *m* is the mode being 0=no looping, 1=forward loop, 2=forward/back loop. The Play List is used to enter the number of times to loop the cue range. Add the cue range to the Play List, then enter the number of times to loop.

Dialogic ADPCM (.VOX)

This is yet another 4-bit ADPCM file format. It has been optimized for low sample rate voice, and will only save 8000Hz Mono 16-bit audio. This format has no header, thus any file format with the extension VOX will be assumed to be in this format.

Raw PCM Data (.PCM) (*.*)

This format is simply the PCM dump of all the data for the wave. No header information is contained in the file. For this reason, you are asked to select the waveform sample rate, resolution, and number of channels. By reading audio data in as PCM, practically any audio file format may be read in! You must have some idea first of how many channels, and the sample rate. You can also interpret the data as A-Law or mu-law. The waveform will sound bad in different ways depending on which parameters you have mixed up. Once the wave is loaded, and sounds fine, you may hear clicks at the start, end, or sometimes throughout. The clicks are various header information being interpreted as a wave. Just cut these out, and *Voilà!* You have read in a wave in an unknown format! When saving raw data files, an optional header can be written to a separate .DAT file to make reloading easier.

ACM Waveform (.WAV)

Any file format supported by the Microsoft Audio Compression Manager can be loaded or saved. When

saving, only the ACM formats that are compatible with the format of the current wave will be displayed under Options.

Choosing a Directory

Choose the directory you wish to copy files to. That's all!

Opening a Waveform As Any Format

This works just like Open except that the sample format can be specified before opening. If the sample format selected is different than the native format of the wave being opened, then it will be converted, using the "quick-and-dirty" conversion method. This means that if the target sample rate is different, the conversion will not try to pre-filter or post-filter the samples. This function works great for opening files as different bit rates and number of channels, but for differing sample rates, open the file normally, then use Convert Sample Type

After choosing the filename, choose the new sample type. The waveform will be opened, and converted as it is opening.

Re-Opening a Waveform; Revert to previous waveform

This will reload the previously loaded waveform, replacing any changes that were made to the waveform being edited. Be aware you will lose any changes you have made since the last time the wave was saved.

Editing Individual Channels;Single Channel Editing

For stereo waveforms, it is sometimes handy to be able to copy from, and paste to individual channels. Normally, both **Edit Left**, and **Edit Right** are checked, meaning both channels are being edited simultaneously. To edit only one channel, uncheck the channel you want to preserve, so only the other channel is being edited. When you do this, most of the commands are still active. When pasting, the audio data is overlapped with what is already there, since inserting only on one channel will put the stereo-ness completely out of phase. If this is your desired effect, use the Delay function.

Viewing Range

Selecting **Edit -> Viewing Range** or double-clicking on the green/black samples portion bar will bring up the viewing range window.

Enter the leftmost and rightmost samples that you wish to have displayed. Use this function to highlight a specific number of samples by double-clicking in the wave editing area after selecting the viewing range to "select all". Remember, Select All only selects the viewing range, not the entire waveform.

New Instance

... Or "Give me another window"

Selecting New Instance will open another Cool-Edit window as if you opened another instance from the Program Manager. This may be handy when you need a secondary window to do some editing in, and don't want to disturb the original wave. There is *almost* no limit on the number of Cool Edit windows that can be showing - the limit is based on the amount of available RAM.

Frequency Analysis

This will bring up an analysis dialog, containing a graph of the frequencies present at the insertion point (yellow arrow cursor), or at the center of a selection. A Fast-Fourier-Transform is performed to determine the frequencies present. Also, the most prominent frequency is interpolated, and displayed below. By moving the mouse over the graph area, the frequency, and percentage (just the height of the mouse position) are displayed.

The information in this dialog is like one "slice" or line in the Spectral View of the waveform.

If stereo data is being viewed, the left channel will be shown in Cyan while the right will show in Magenta (kind of Left=Blue, Right=Red like the icon).

Check the **Normalize** box to re-scale the graph so that the highest intensity sample reaches the top of the graph.

Spectral Viewing

Choosing **Spectral View** will display waveforms by the by their frequency components. This is a handy function for analyzing your audio data, to see which frequencies are most prevalent throughout your data.

The more abundant a frequency is, the brighter the color displayed will be. Colors range from dark blue (next to no frequencies in this range exist) to bright yellow (frequencies in this range are very strong). On 256 color displays, there will be more gradations between the colors.

Lower frequencies are displayed near the bottom of the display, while higher frequencies are displayed near the middle on to the top. The display is linear. White lines on the left and right divide the display into 1/2, 1/4, 1/8, and so on. To top of the display represents frequencies at just below the Nyquist frequency, or 1/2 the sample rate. So if a bright spot appears near the top of the display for a signal sampled at 22 KHz, the frequency being represented is near 11 KHz.

Choose **Waveform View** to return to original waveform view mode.

Select Wave Device

If you have multiple sound cards, or multiple output devices (such as a sound card *and* the PC speaker), choose the input and output devices you wish to use.

If your system is equipped with MIDI devices, you may also choose the MIDI in, and MIDI out sources.

The capabilities of the recording and playback devices are displayed in the given tables.

The settings are remembered in the [cool edit] section of your WIN.INI, which means if you install a new sound driver or card, Cool will not access it until you choose it from this dialog.

FFT Filtering

You may filter out undesired frequencies, or just keep certain desired frequencies by using the **Passive** mode.

Use the **Logarithmic** mode to boost or dampen frequency components.

When the **Lock** is not set, you can choose both an **Initial** and a **Final** filter. Filtering will gradually go from the initial state to the final depending on the Transition settings.

Graph Values in dB or percentage (depending on logarithmic or passive mode) can be entered to the right of the graph. These values are independent of the shape of the graph, which means that loading a preset *will not* change these two values.

The **Precision Factor** determines how accurate you want the filtering over time. A low factor, means the filter settings will change roughly, or in chunks, from the initial to the final settings. With higher factors, the filter's transitions are much smoother. In any case, the higher the precision factor, the longer it will take to filter your selection, but the nicer it may sound. The FFT (Fast Fourier Transform) function takes a large group of samples, and filters them all at once. The precision factor determines how many samples from the entire group are actually saved in the final product. A factor of two means that half the samples are saved back. A factor of 10 means that 1/10 of the samples are saved back. Since there can be only one filter setting for the entire group of samples, you will want smaller groups of samples if the settings are varying widely over short periods of time.

The **Points** parameter specifies the size of the FFT to use. For cleaner sounding filters, use higher values. The maximum value currently is 8192, and the value must be a power of two. Recommend values are 1024, 2048, 4096, and 8192.

Windowing Function

This is the windowing method used when filtering. Different windowing methods give different frequency responses. The Hamming and Blackman windows give great overall results. The windowing method determines the amount of transition width and ripple cancellation, and are in order from smallest width and greatest ripples to widest width and least ripples. The filters with the least ripples are also those that more precisely follow the drawn graph, and have the steepest slopes, even though they are wider, and pass more frequencies in a band-pass operation. Try different Windows if you are not getting the effect you desire.

If **Morph** is checked, the transition from the initial filter settings to the final filter settings will actually "morph" from one to the other. If this is not checked, the settings simply change linearly over time, which means if you have a spike at 10K for the initial filter, and a spike at 1K for the final filter, the spike at 10K will gradually decrease, and the spike at 1K will gradually increase over time. If morphing is on, then the spike will "ooze" from 10K down to 1K, passing many of the frequencies in between.

Really nice effects can be heard by simply choosing the **Passive** mode, and having an initial setting with first half of the filter at 100%, and the second half at zero for the initial filter, and the right 1/10th or so at 100% with the rest at zero for the final filter. This selects high frequencies for the initial configuration, and low frequencies for the final configuration. To get a nice blending from high to low, choose **morph** to blend the two together by including all the frequency combinations between the two filters. To see exactly what is happening as the filtering changes from the initial configuration to the final, choose Transition to view the actual settings that will be used over the duration of your selection.

Currently, the noise floor when using this filter is at about -70dB. That is, if you cut out a frequency completely, and leave complete silence, there is noise at -70dB. I am hoping to add an HQ feature to bump this up to effectively -90dB.

This function supports Presets.

For best results, filter using 16-bit samples. If your source is 8-bit, use Edit->Convert Sample Type to convert to the 16-bit format. You can convert back to 8-bit when done if you are trying to save space.

To keep the transitions from one filtered block to the next smooth, a percentage of the previously filtered block is mixed with the block currently being filtered. This will ensure that there is no added high frequency noise when using very narrow filters.

FFT Transition Settings

Choose how you want your sample filtered over the time of the sample. The left of the input graph represents the start of your sample, while the right hand side represents the end. You can choose how your selection will be filtered over the length of the selection. The low points represent filter settings close to your initial filter, while the higher points represent the settings close to your final filter. All points in between are a combination of your initial and final filter arrangements.

The actual graph of the filter being used at any point can be seen below. The filter setting shown corresponds directly to the position of the mouse in the input graph. By watching how the filter settings change as you move your mouse up and down in the graph area, you can decide whether you want a morphing transition, or a linear transition. Each type of transition will give different filter settings for the points between your initial and final filter settings.

Morphing is generally any technique used to transform one object into another. In the case of filter settings, it is a way to smoothly transform one setting (represented by a graph) to another by estimating all the possible combinations of the two settings. Over time, the first setting becomes the second. At some point, the setting will stop looking like the initial configuration, and start looking like the final configuration. In the filtering world, this means that frequencies between the ones selected to be filtered will also be filtered.

In going from the initial to the final filter configuration, the points in the "in between" settings are just the average between the two settings. For example, a filter setting exactly between the initial and final would be the exact average of the initial and final filter settings.

Musical Incantations

Now you can put your clippings to music, or just harmonize a wave using a particular chord. To choose a clipping for your sample, select the range you wish to use as a quarter note. If no range is selected, the clipboard data will be used. Note that the clipboard data will be filled with your sample automatically once music is generated, thus selecting music a second time will automatically use your last sample.

This function is by no means a complete MIDI authoring studio. It is just meant as a quick and simple way to put a sample to music. The only MIDI support is in the preview playback when the Listen button is pressed.

How to build a song

Simply drag the notes and rests you desire to the music bar above. To sharpen or flatten a note, drag the sharp (#) or flat(b) symbol on top of the note you wish to transpose. You can move notes up or down after they have been placed, or pick them up to insert in a new position. To remove a note, pick it up and drop it off away from the bar.

Use the scroll bar to work on individual portions of the song at a time. You can scroll to write a piece as long as 256 notes.

Tempo

The tempo is given in quarter notes (beats) per minute. Your sample's length is the length of a quarter note. If your note is longer than the period determined by the tempo, then the notes will overlap.

Key

You may choose to have your music interpreted in any of the standard key signatures. The key of C is the standard (white keys).

Constant Duration

If chosen, all notes will be the same length as the original sample, regardless of pitch. The operation that does this takes longer to calculate, but high pitched notes will be the same length as lower pitched notes. The Interval Overlap method is used with an overlap of 80% and an interval of 30 Hz. If not checked, the note is created by directly stretching or compressing the original sample, resulting in higher pitches being shorter than lower pitches.

Exact Tune

Choose Exact Tune to tune your sample so that when played at A (above middle C), the frequency of your sample is at 440Hz. If this is not checked, your sample's original frequency will be played at A (above middle C).

Chords

The triplets of numbers to the right is the chord selection box. You can choose to make a chord out of 2, 3, or 4 notes, then choose the chord from the list. Finally, pick up a chord object (the 3 notes on top of each other) and drop it on a note above. The note you drop it on will be the starting note of the chord, and the other notes will automatically appear above it in the right ratios.

Clearing Chords, Sharps, or Flats

If you want to clear a sharp, flat, or chord from a note, use the faded looking quarter-note object, and drop it on the note you wish to bring back to normal.

Saving Your Songs

If you make a cool song you want to keep, give it a name in the **Song Title** box. In the future, you can choose your song from the list of song titles that you created. The actual song data is saved in the file SONGS.INI in your Windows directory.

Listen

If you have MIDI play capabilities, you can listen to a preview of your song before actually creating it. Play begins at the leftmost note visible on the staff, which means play begins at the position you are scrolled to, and continues on to the end of the song. The music is played through channels 1 and 13 for Extended and Base level compatibility. The instrument can be chosen by typing its instrument number to the left. You can record music played by the listen preview button. Simply hit the record button first, then go into the music dialog and press Listen. When the song is done, hit Cancel, and then Stop to stop the recorder.

Pink

What the heck is this button for? Well, it automatically plays the chosen instrument through the MIDI, using pink noise as the source for randomness. Maybe once out of 1 million tries, it may actually write a cool song? Tempo, Octave, and Key all affect the play of "pink" music. The only purpose for this button I have found so far, is to open the Play List first, then open the music dialog. While listening to a relaxing soundscape from the play list, you can listen to relaxing random music at the same time. (?) (!)

Saving a Waveform

Waveforms can be saved in many formats. Having many format options gives you the chance to save the waveform in the smallest amount of space while still retaining a high level of quality. PCM formats are the only kind that save the entire wave completely with no data loss. ADPCM formats lose some of the original waveform information and the waves sound a little distorted. Mu-Law and A-Law formats also lose some of the information, but it may not be as noticeable as ADPCM, but it depends on the type of audio that is being saved.

Some file formats support various options that can be modified by pressing **Options**.

See [Wave Formats](#) for a list of supported formats.

Saving the Highlighted Selection

Saving a wave selection is identical to Save except that the highlighted selection is saved, and not the entire waveform.

Immediately Saving a Waveform

Choosing **Save** from the **File** menu will save the current waveform back to disk, overwriting the original without confirmation.

The disk icon in the toolbar can be set to save immediately, or to bring up the **Save As** box depending on the setting in the Settings section.

Repeat Last Command

Pressing **F2** or this will repeat the last function that modified waveform data.

Press **F3** to repeat the command immediately, bypassing any settings dialogs.

Settings

Disk Icon Interpretation

The Disk icon can be interpreted as:

- Save Now (saves what you are working on under the same filename seen in the title bar without asking if you wish to overwrite the original) or
- Save As (Always brings up the Save As dialog box to enter the title to save the file as)

Viewing Mode

When zoomed in closely to a wave, individual samples can appear as dots, or as a continuous line.

Play/Record Buffer Size

The number of seconds to reserve memory for recording and playback. Increasing this will allow more multitasking while audio is being played, but it takes more memory. If this value is too small, there may be too much choppiness in your recordings and playbacks. If your recordings are getting all "chopped up", or you cannot Stop after you've started recording, increase the buffer size, or switch to a faster hard drive (Use a non-compressed hard drive for example).

Temp Directory

This is the location that Cool Edit will use to save all of the temporary files, which includes the disk image of the file currently being edited. It is best to choose a large, fast hard drive for this. If you are not certain that the drive is *not* compressed, then do not check the **Compressed** box, otherwise check this box. If you have trouble recording high quality audio for more than a minute (or even less), then make sure the Compressed Drive box is not checked. When checked, DOS is asked for the number of bytes remaining on the hard drive since this value does not change linearly with time on compressed drives.

Undo Directory

This is the location that Cool Edit will use to save all the Undo files. Up to 3 undo files will be present per instance of Cool Edit being run, to accommodate the three-level undo. This directory can be different than that of the Temp Directory. It makes no difference whether the drive used for Undo is compressed or not - but a non-compressed drive will still be faster.

Highlight After Paste

After doing any Paste operation (this includes Paste Special), you can have the inserted selection automatically highlighted, or just have the cursor at the end of the pasted selection. Not highlighting after pasting makes it easier to do multiple pastes one after the other.

Play 16-bit files as 8-bit

If your sound board is only capable of 8-bit audio, you can still create and edit 16-bit audio files. When you choose Play, the audio data will be converted to 8-bit before being sent to the sound board.

Enable Undo

If this is checked, the undo function is enabled and running. After making a change, you can choose **Edit -> Undo** to back up one step. You may want to disable the undo feature if you know you are not going to undo something. For example, if you are running a function on a 5 minute file, you may not want to wait while the undo information is saved.

NT Display Compatibility

If this is checked, then the program "locks" while the green waveform is being drawn. With Windows NT, other apps are not paused when this is going on, and the extra code needed to "multitask" with other apps does not need to run. If you are running Windows NT, and this is not checked, you may experience various screen redraw problems.

Smooth 16-bit to 8-bit Conversions

If checked, when 16-bit audio is pasted into an 8-bit waveform, or 16-bit audio is Opened As 8-bit, the 16-

bit audio is dithered down to 8-bit to retain more of the original signal. Dithering can keep some of the audio audible that is below the threshold for 8-bit audio. When doing this, you may notice a bit more noise in the background, but the noise is constant, and you can hear some of the 16-bit audio in the noise that was otherwise inaudible. If this dithering is not done, you will notice strange artifacts in quiet audio. Most prefer to have a constant background noise rather than have strange sounding artifacts in the background. Distortion sounds "crackly" if no dithering is used, or else the sounds have a constant hiss if dithering *is* used.

Enable Toolbar Help

When the mouse is over a toolbar button for more than a few hundred milliseconds, a small help box appears to describe the button's function. Anyway, if it is annoying for you, you can turn it off here.

Play from cursor

When no selection is highlighted, audio can be either played from the current cursor location to the end of the view, or always from the left edge of the view to the end of the view.

Maximum Display on Load

This is the maximum number of seconds of audio to display when a file is first loaded. When working with large files, you may wish to limit the initial display area to 10 or 20 seconds so you don't have to wait for the entire waveform to draw. Setting this value to zero means there is no limit on the initial display size.

Time Code Display

Double-clicking on the time boxes will change their display format. When in Hours:Minutes:Seconds:Frames format, you can customize the number of frames per second that will be displayed.

Spectrum Analyzer Resolution

This is the number of bits of accuracy to display the spectral analysis when Edit->Spectral View is chosen. Each bit will take approximately twice as long to draw, but the resolution will be more precise. The number of data points displayed is equal to 1/2 of two to the value given. In other words, a value of 7 will display 64 points along the y axis, a value of 8 will display 128, 9 for 256, 10 for 512 and so on. On most displays, a value of 10 is as high as you will ever need to go.

Cue List

A cue list is a list of time offsets into the wave file. A cue can be either a point, specifying a cursor position, or a range, specifying a selection. You can easily jump to a cue position in a wave by double-clicking on the position in the list, or selecting the cue position, and pressing Goto. Cue ranges can later be arranged in a play list to be played back in any order, with a specific number of loops if desired. A maximum of 256 cues may be entered.

<- Add

Add the currently highlighted selection, or cursor position to the cue list. Items will be displayed in temporal order, with the earliest cue position at the top of the list. Pressing F8 when editing a waveform will add the current range or cursor location to the cue list.

Remove

Remove the selected cue position from the list.

Label

Short text label describing the selection.

Description

A textual description of the wave data if necessary. Also can be used as a comment.

Goto

Goto the selected cue position, or highlight the selected range. Double clicking a cue item acts as if Goto were pressed for that item.

The cue list is saved in the .WAV file format in the 'cue ' chunk. Additional information about the cue position, like label, description, and length of sample, are placed in the 'adtl' list in the 'labl', 'note', and 'ltxl' chunks.

Merge

Merge will create a cue range that spans the two cue items selected (whether they are ranges or markers themselves). To select more than one cue item in the list, hold down on the CTRL key when selecting, or click and drag over more than one cue item. The name used for the new merged item will be the same as the earliest item chosen in time (the highest item in the list). The information typed into the Name and Description fields for the second item being merged will be lost.

Markers

The cue list can be used anytime to mark your current selection so you can return to it later. If you would like Cool Edit to remember your highlighted selection, or just your current cursor point, click Add in the cue list, and quickly type a name for your selection. In the future, if you want to return the cursor to that point, or re-highlight that selection, double-click the name or choose the name and click Goto. One great use for markers is to highlight a wave from the zero crossings. Go to the start of wave portion you wish to highlight, and zoom in as far as needed to position the cursor exactly on the zero-crossing point. Add that position to the cue list. Now zoom out, go to the end of the wave portion, and once again zoom in to find the ending zero crossing. Now, double-click on marker name in the cue list, hold down on the Shift key to extend the selection, and click on the ending zero crossing. Voila! You can choose "Zoom In" now to see your wave portion if you like... it is now selected.

Visual Representations

Cue markers will be displayed in the main waveform as red arrows above and below the wave. Cue ranges will be displayed as blue brackets above and below the waveform.

Assigning Cue Ranges To Keys

If you wish to assign any cue range you have added to a key on the keyboard, give the cue range a label of the form **KEY N**, where **N** is any key on the keyboard (capital letters only). When you go back to

editing the waveform, pressing the key will play the cue range you selected. You can assign any portion of the waveform to any key on the keyboard. If you have any problems when playing audio by pressing the assigned keys, try increasing the STACKS line in CONFIG.SYS to read STACKS=12,512.

Play List

The play list is a listing of cue ranges that can be played in any order, and looped a specified number of times. The play list is used in conjunction with the cue list. Maximum size of a play list is about 4,000 entries.

<- Add Before

Add the currently highlighted selection from the cue list to the play list. The selection is inserted before the currently highlighted play list item, or at the end if nothing is selected.

Remove

Remove the selected play list item from the list.

Loops

The number of loops to loop the selected cue range in the play list.

Play

Play the cue ranges in the order listed, looping selections if necessary. Play begins at the currently highlighted item in the play list, or the entire list is played if [end] is selected, or there is no selection.

Autocue

Play the currently highlighted item in the play list (or the first item if nothing is highlighted), looping if necessary, and stop on the next item in the play list. Thus, every time **Autocue** is pressed, the next item in the play list is played.

The play list is saved in the .WAV file format in the 'plst' chunk

Reverse

This does just as you'd expect. Select a portion of the wave using the mouse, and select Reverse from the Transform menu to reverse your wave.

Trim

Trim is the exact opposite of delete, which means everything is deleted *except* the portion that is selected. Only the selected portion is kept. This is handy to quickly pick out the part of a recording you want to keep.

Use CTRL+T to quickly trim the selected portion of the wave.

Editing Channels of Stereo Waveforms

When fully implemented, you will be able to make edits (copy, paste, etc. and so on) to only one channel, completely independent of the other.

Digital Delay

Delay either channel up to 50 milliseconds with the option to mix in the original signal with the delayed signal. Great for effects such as spatially locating a previously mono wave source to the left or to the right, so that the sound will appear to emanate from that direction when listened to with stereo headphones. Delays of longer than 50 ms may be entered for creating a single echo.

Delay (ms)

The actual amount of time to delay the channel in question.

Mixing

You may choose to have the resultant wave the delayed signal, keep it the original signal, or mix the two. A value of 50 will mix the two evenly.

Invert

The delayed signal may be an inverse of the original if this is checked. More special effects!

This function supports [Presets](#).

Tunnel Preset

The Tunnel preset can be used with mono as well as stereo waveforms. The settings provided give a nice tunnel/tubular effect.

Spatial Left Preset

If a mono wave source was converted to stereo (so that the left and right channels are the same), then choosing this will make the sound appear as if it is coming from the left, since the right channel is delayed just enough so your brain interprets the sound as coming from the left. You must use headphones to hear the effect.

Spatial Right Preset

The same as Spatial Left, but it locates the sound to appear as if it is coming from the right.

Stretch

You can choose the initial stretching percent (less than 100% will compress the wave), and the final stretching percent. By having two separate values for start and end, the wave can be stretched linearly from one ratio to another. This gives the effect of slowing down and speeding up, or raising and lowering pitch.

Preserve Pitch

Lower percentages will slow down the tempo, while higher ones will increase the tempo. The pitch remains the same throughout.

Preserve Tempo

The tempo or speed of play will remain the same. The only thing affected will be the pitch. Higher percentages will lower the pitch, and lower percentages will increase the pitch. Try using differing initial and final percentages to raise and lower the pitch without affecting the tempo. First the selection is adjusted, preserving the pitch, then the selection is squeezed or expanded, with no preservation.

Preserve None

The tempo will slow, while at the same time the pitch will lower if percentages above 100 are used. For lower percentages, the tempo will speed up and the pitch will increase.

Ratio, Length

The desired Ratio or final Length can be entered. Entering one automatically changes the other. If the initial and final lengths are different, then the actual final length will be exactly $(\text{initial} + \text{final}) / 2$ when in Preserve Pitch mode.

Transpose

When the number of desired keys up or down are chosen, the numerical values for transposing musically are entered into the stretch sliders. For example, to make your sound as if it were the next key higher (if played on a keyboard, and black keys included) choose 1# for 1 sharp. The 'b' values will flatten or lower your sound.

Interval Overlap

When preserving pitch or tempo, the waveform must be elongated or truncated smoothly and preserve as much of the original information without adding noticeable distortion. This is one of two methods that can be used to achieve this. The amount of distortion introduced is not dependent on the type of sample (e.g. music or speech). When stretching or compressing, the appropriate chunk from the original wave is output to the transformed wave, and overlapped with the previously transformed chunk.

The **interval rate** determines the size of a 'chunk' of audio data. Interval rates will become an audible *hollow* sound when large rates (above 50Hz) are used. If the rate is too low, echoing will be very noticeable when raising pitch, or slowing down tempo, or chopped syllables will be noticeable when lowering pitch, or speeding up tempo. Values of 20Hz to 40Hz usually produce good results.

The **overlapping** determines how much of the previous chunk is overlapped with the current chunk. The overlapping can produce a *chorus* effect. To reduce the chorus effect, lower the overlapping percentage. When the overlapping is reduced, a *choppiness* to the sound may appear. Adjust the overlapping to your taste to strike a balance between choppiness and chorusing. Overlapping can be as high as about 400%, but only use this for really high speed increases (like 200% or more).

Fractional Interval Overlap

This second method of preserving pitch or tempo uses a more straightforward algorithm which guarantees the timing to be as close as the interval rate allows. This usually causes more of a reverb/echo effect with the audio, but in cases where stretching is being done at rates lower than 60 percent, or higher than 175 percent, this method can produce more desirable results..

The **interval rate** determines the size of the chunk of audio data in the final stretched wave. If there are any artifacts, they will be heard at this frequency. For example, an interval rate of 60Hz will cause a 60Hz distortion artifact. Values of 15 down to 7 or so seem to work best. The higher the value, the more precise the placement of stretched audio over time, but also the artifacts are more noticeable as rates go up.

The **overlapping** determines how much the current chunk overlaps with the previous and next chunks. The maximum overlapping allowed can be as great as 1000%, in which up to 10 sections of the wave are overlapped together.

If using either of the Interval Overlap methods, you can improve the quality of the stretched audio of mono tonal (pure tone) samples by choosing an Interval Rate that is evenly divisible into the frequency of the sample. Use the Frequency Analysis window to find the sample's base frequency, then divide by an integer to get the Interval Rate. For example, if the tone was reported to be 438Hz, dividing by 20 gives 21.9Hz, so using 21.9Hz for the Interval Rate will greatly improve the quality by reducing the phasing artifacts. For very non-tonal or noisy samples, the Interval Rate does not matter as much.

Zero Splicing

This is the third method of preserving pitch or tempo. Less distortion is noticed when using samples that were derived from one source, and contain a relatively low number of fundamental frequencies. For example, a single instrument, or speech will work OK, while an orchestra or music will not work as well. When stretching or compressing, the wave is broken up into chunks that begin and end when the waveform crosses zero, or the midpoint. Chunks are repeated, or thrown out depending on the compression ratio and the cutoff frequency. Chunks smaller than the **cutoff frequency** will not be thrown out or repeated. Try cutoff frequencies between 50Hz and 300Hz for best results using this method.

This function supports [Presets](#).

Raise Pitch Preset

This will adjust the settings so that the pitch of the sample will be raised by 1/4 octave.

Lower Pitch Preset

This will adjust the settings so that the pitch will be lowered by 1/4 octave.

Speed Up Preset

This will speed up the sample, so talking will be faster, but the pitch will be unaffected.

Slow Down Preset

Slows down the sample.

Note: An alternative method of stretching while preserving pitch or tempo is to do an FFT, and modify the amplitudes and phases of each frequency in the frequency domain, and then do the inverse FFT to translate back. This method would be very complex, and be very very much slower than the methods currently being used. In comparison, the methods presented come strikingly close in quality to some FFT methods.

Compact Disc Player

If you have the **[MCI] CD Audio** driver loaded, you can control the CD player with a standard set of control icons. You can also name your CD, and the individual songs as well for display the next time you use the same CD.

Tracks List

Click on any track number to start playing that track.

Time Readout

Displays the current time in minutes:seconds into the current track.

Title Display

Displays the title of the CD. The title defaults to the length of the CD. You can type in the proper title of the CD here. If a track is currently being played (if the track number is highlighted in the track list), then the title reflects the title of the current song. As you are entering song titles, you may use the TAB key to jump to the next song to easily enter the titles for the entire CD. Titles are saved in the file(s) COOLCDx.INI where x ranges from 0 to 99. Once one of the COOLCDx.INI files reaches about 64K in size, the next file is used for storing new data. This keeps the size of each INI file below 64K which is a Windows requirement. Song titles in the file MUSICBOX.INI in your Windows directory will also be displayed properly, and can be edited from within Cool Edit as well. Using this method, thousands and thousands of CDs can be entered completely without fear of losing your favorite tracks.

When working with the CD player, the any track may be chosen by typing the track number.

[Stop]

Square Box: Stops CD playing. Play will resume at the start of the CD.

[Pause]

Vertical Bars: Pauses the CD. Play will resume at the same location. This button will turn into a **Play** button when pressed so that play can be resumed by pressing it.

[Play]

Right Arrow: Starts the CD either at the beginning of the disk, or at the paused location. This button will turn into a **Pause** button when pressed, so that play can be paused by pressing it.

[Scan Back]

Double Left Arrows: Rewinds the CD 10 seconds.

[Scan Forward]

Double Right Arrows: Forwards the CD 10 seconds.

[Mark]

Red X: Mark the location currently being played.

[Goto Mark]

Arrow to Red X: Go to the location that was marked earlier.

[Eject]

CD Ejecting: Spit out the CD if that is possible on your player. The icon changes to [Insert] if no CD is in the drive, and if your drive can do it, the CD will insert when you press the button.

Selecting **Exit**, or double-clicking the system menu box will close the program. At close time, any temporary files that were created will be removed.

RIFF Info

Extra information can be included in your .WAV files using the RIFF LIST INFO and DISP type 1 formats. This information should (depending on future wave editors) stay with your sound through it's lifetime. Other wave editors should preserve some of the fields you see here. If the "Fill * Fields Automatically" box is checked, then the Software Package and Creation Date fields will be automatically filled by Cool Edit. If you do not want this extra information to be tagged with your wave files, uncheck this box. Be sure to put proper information in it's place!

Display Title

This should describe the sound, or text (if there are words in the wave). This field should be as short as possible, since it will be displayed in OLE objects and the like.

Icon

Any DIB or BMP file can be inserted, but preferable a 32 X 32 16-color would be best. The Media browser uses this size to display a picture representing the sound. Other OLE compatible applications can use the above display title, and/or the bitmap to represent your waveform.

Original Artist

The one who created the sound initially. Examples are: Beatles, Pat Sejek, Fred Flintstone

Name

The title of the wave. This is your chance to put a name with your audio "artwork". Examples are: Thunderstorm At Night, Forest Stream

Genre

The Genre of the original work. With audio, let's try things like musical classifications, etc. Examples are: Cartoon Voice, New Age, Instrument

Key Words

In the future, sounds may be searched for by key words. Please separate key words by a semicolon followed by a space. For Example: Violin; Hayden; Johann Strauss

Digitization Source

Where was the sound digitized from. A tape deck, CD, or maybe directly from a microphone? Maybe describe the board used here too, like Sound Blaster Pro, or MediaVision. For Example: DDD CD to MediaVision Pro 16

Original Medium

Where did the sound come from originally. Examples: Live Band, Flute, Moog, Voice

Engineers

Store the name(s) of the engineer(s) who worked on the file, or edited the file. Please separate names by a semicolon and a space. When a new person edits the file, they can add their name to the list. For example: John Cravitz; Fred Millstone

Digitizer

Who is the technician that did the actual digitizing? They should put their name right here.

Comments

This is for making any comments you wish. Feel free to include any special effects or enhancements you made to any preexisting waves so that the editing history can be tracked. Please do not use any line returns. End each sentence with a period. For Example: It took me 12 hours to get this recording right. John added echoing effects using Cool Edit.

Subject

This Describes the content of the file. Feel free to include a description of the instruments used, where someone can find the song recorded, etc. Line returns are OK, and are created by pressing Ctrl+J. Sometimes copyright information is placed here as well. For Example: The shakuhachi of Japan.<Ctrl+J><Ctrl+J>The shakuhachi was developed in the 15th century from a Chinese end-blown flute, called the chiba.

Source Supplier

The name of the person, or organization who supplied the original source material. Let's use this field for the names of record companies, or whoever supplied you with the source. Examples: MCA Records, Ann Wilson (if recorded live)

Copyright

Any copyright information for this file should go here. Example: (c)1992 G. Willikers Corporation. All rights reserved.

Software Package

The software used to digitize and edit this file.

Creation Date

The date that the subject matter was created. The date should be in the format yyyy-mm-dd, using '0' as a place holder in single digit values. For example, if the date the original recording was made was July 30, 1988 then it would be written as: 1988-06-30

Embedding Bitmaps in WAVs

Any DIB or BMP bitmap can be inserted into the wave file. The bitmap will only be saved if the file is saved as WAV. Icon sized bitmaps (32 X 32) are preferable, since they are easier to display by other OLE aware applications. OLE applications may represent your wave file using the bitmap you select.

Cut

Cut will copy the selected range to the clipboard, and remove it from the waveform being edited. Once in the clipboard, it can be pasted or special pasted to waveforms, or pasted to other applications that support the CF_WAVE clipboard format.

Use CTRL+X or SHIFT+Delete to quickly cut the selected wave.

Envelope

By using an envelope, you have great control over which parts of your wave are amplified, and by how much. A point at the top of the graph is 100% (normal) amplification, and at the bottom is no amplification (silence). This function is handy when modifying Tones generated with this program to create more realistic sounding instruments and effects.

Amplification

Adjust this value to amplify more than 100%. This value changes the values represented by the graph.

Click in the graph area to add control points. You can drag control points up and down. Drag a control point off the graph area to remove it.

This function supports [Presets](#).

Note: This function relies on the AMPLIFY.XFM module for doing the actual amplification.

Generate Silence

Enter the number of seconds of silence desired to generate that amount of silence at the insertion point.

Pretty simple, eh?

Silence

Choosing this will silence out the selected material.

Just about as simple as Generate Silence, huh?

Generate DTMF Signals

This function generates Dual Tone Multi-Frequency (DTMF) signals used for dialing telephone numbers over the PSTN. These signals are recommended internationally by the International Telegraph and Telephone Consultative Committee (CCITT) as the signals for push-button telephones. The DTMF signals generated by telephone push-button keypads are different from the Multi-Frequency (MF) tones generated by the telephone network to transmit information.

Dial String

Enter the *phone number* you wish to generate the tones for in this box. Other characters may be entered such as the '*' and the '#' symbols, as well as extra digits 'a', 'b', 'c', and 'd'. Entering the pause character (defined below) will insert a pause of the length defined.

Tone Time

All tones will last the milliseconds entered. The standard time for DTMF tones is 100ms.

Break Time

This is the number of milliseconds between successive tones.

Pause Time

This is the number of milliseconds to use for a pause (when the pause character is used in the string).

Pause Character

When the character entered here is also typed in the Dial String, it will be interpreted as a pause, and silence will be inserted for the duration specified by the Pause Time value above. The default pause character is the hyphen "-", but any character may be entered here and used as the pause character.

DTMF Signals

DTMF (normal push-button telephone type) signals will be generated using combinations of the frequencies 697Hz, 770Hz, 852Hz, 941Hz and 1209Hz, 1336Hz, 1477Hz, 1633Hz.

MF Signals (CCITT R1)

MF (internal to telephone networks) signals will be generated using paired combinations of the frequencies 700Hz, 900Hz, 1100Hz, 1300Hz, 1500Hz, and 1700Hz.

Amplitude

The volume level of the tones being generated, with 100% being maximum volume without clipping.

The presets in this function save everything, including the dial string. To see how effective these tones are, try typing in your favorite phone number to generate the tones for it. Then hold the receiver of your phone next to the speaker and play the wave. It will dial the number you had entered!

Generate Tones

Generating tones is a great way to provide a base sound to create spectacular special sound effects.

Base Frequency

The main frequency (F) that will be used for sound generation.

Modulate By

Enter the variation in frequency you wish to hear. For example, choosing 100 will oscillate the tone being generated between 50 minus and 50 plus the base frequency.

Modulation Frequency

This is the rate (times per second) at which the frequency modulates. Entering a value of 10, for example, will generate tones that warble at the rate of 10 times per second.

Start Phase

This is the phase, in degrees from 0 to 360 that the waves (sine waves for example) will start out at. Generally, keep this at zero, but if you are doing phase critical operations, you may wish to start at a different phase. Generally, one can not tell the difference between a signals that start at different phases. Keeping this value at zero will ensure that the tones generated will start at a zero crossing.

Flavor

Choose the type of waveform to use. Sine waves sound soft, while Triangle and Sawtooth waves are sharper. Each flavor has a particular sound unique unto itself.

Modulate (over Source)

If a selection was highlighted, it will be modulated by the tones based on the normal tones settings. Instead of generating new tones, the currently highlighted wave data will be "ring modulated" by multiplying the tone by the data underneath. This is great for adding really weird special effects.

DeModulate

If a selection was highlighted, it will be demodulated by the tones (this is the exact opposite of choosing Modulate over the source material). Demodulating by the exact same tone parameters will always retrieve the original signal after modulation. Since modulation involves multiplying every sample by the tone samples, demodulation involves dividing the signal by the tone samples. Because of this, if a sample is near zero, it will be truncated to exactly zero instead of being amplified to infinity. Modulating a signal by a high frequency sine wave (like 8KHz) allows the sound to be sent long distances under water for example. The receiver of the 8KHz modulated signal would then demodulate by the same frequency (and the proper starting phase must be chosen as well) in order to get the original signal back. Since low frequency audio does not travel as far under water, modulating it first allows it to travel great distances. Another example of using modulation and demodulation would be in voice encryption. The sender would modulate by a known frequency pattern, and the receiver would then demodulate the 'scrambled' signal to retrieve the original message (the receiver must know the exact parameters the sender used to modulate the signal of course). When demodulating, phase is very important and you will most likely need to adjust the Start Phase of the demodulating signal. If a signal was not already modulated, demodulating by any signal will most likely produce very ugly distorted noise.

Stereo Phasing

The left channel can be out of phase with the right channel. If you wish this, choose the amount of phase shift here. A value of 180 will be completely out of phase.

Duration

This is how many seconds of tones you wish to produce.

Frequency Components and Multipliers

You can choose up to 5 overtones, and mix them at any proportion, and change the multiplier used to gain the overtone's frequency. If Lock is not checked, all these can change over the duration of the signal from initial to final settings. The value entered below the overtone slider is the frequency multiplier. The actual frequency will be this many times the base frequency. Moving the slider from 0 to 100% changes the ratio this slider's frequency with respect to the others. Lots of really great effects can be generated with just these 5 overtones. Just experiment and have fun!

Lock

If checked, the overtones, base frequency, modulation, modulation frequency, frequency multipliers, and overtone intensities are constant -- they do not vary over time. Uncheck this box to dynamically change the proportion of any overtone over time by choosing the initial and final proportions. You can also dynamically change the frequency multipliers, base frequency, modulation, and modulation frequency for interesting effects.

Initial and Final

You can choose to adjust the initial or final overtone proportion settings when the overtones vary over time. You can also choose different base frequencies, modulations, and modulation frequencies.

Flip

Check (or uncheck) Flip to swap the initial and final settings. This is very useful if you wish to generate another tone *after* the last tone that was just generated. Whatever overtone settings the last tone had as the final settings can be used as the initial settings on this tone by clicking Flip. All you need to do is click Final, and enter new final overtone values. This way the tones will flow together smoothly.

Volume

The Volume sliders are used to select how intense the tones will be in each channel. Both channels can be controlled independently when generating stereo tones.

This function supports [Presets](#).

Experiment with all the settings for various wild effects.

... and Have Fun!

Paste

paste will insert the wave from the clipboard at the current insertion point, replacing any waveform data in the selected range. If the format of the waveform data in the clipboard differs from the format it is being pasted into, it will be converted accordingly before pasting occurs.

Use CTRL+V or SHIFT+Insert to quickly paste the waveform that is on the clipboard.

Delete Selection

Once a range is Selected, it can be removed by choosing this option. The deleted portion is *not* copied to the clipboard. It is gone forever.

Quick Filter

The 8-band quick filter allows one to customize to suit most filtering needs. The "equalizer" works pretty much the same as a standard audio equalizer does. Except that the bands are not the same as you would expect. The highest frequency band *will* increase or decrease the high end, but it will also increase frequencies all the way down to the lowest as well, but it will increase the high frequencies more than the low ones. The effect is close to an equalizer, but not quite. Basically, this is a handy function for changing the tone of your sample (such as noise) to make it more pleasing to the ears.

Equalizer Bars

Adjusting these increases or decreases the frequency component specified beneath the bar.

Volume Bars

The final volume after equalizing may be adjusted to suit your needs. Checking the **Lock Vol** checkbox will lock the left and right scroll bars.

Flat

This simply places all equalization values plus volume adjust at 100%

Lock Initial/Final

When locked, the entire selected range is equalized with the setting shown. If unchecked, the initial and final equalization settings may be adjusted, so the selection can smoothly glide from the initial equalization setting to the final setting over the range selected.

View Initial

When Initial/Final is not locked, choose this to select the initial equalization settings.

View Final

When Initial/Final is not locked, choose this to select the final equalization settings.

This function supports [Presets](#).

To produce a semi low-pass filter, set the higher frequency scroll bars to zero to cut out higher frequencies. A high pass filter can be done in the reverse fashion, by zeroing out the lower frequencies.

Very interesting effects can be made by selecting widely varying initial and final equalization settings.

Note: Setting the lower bands to very high values can, and most times will, result in clipping if the volume adjustment is not turned down.

For serious filtering, see [Filter](#).

Copy

Waves can be copied to the clipboard to be pasted later using this program, or any other application that supports the CF_WAVE clipboard wave format. Before copying, the portion that is to be copied must first be selected.

Extra large waves may not be able to be copied. If this happens, try breaking up the selected range into smaller portions and copy/paste them in multiple steps.

Use CTRL+C or CTRL+Insert to quickly copy the wave.

Amplify

Amplify will increase or decrease the volume of the selected sample.

Initial Amplification

This is the amplification that will affect the beginning of the selection. Choose a separate final amplification for fading up/down effects. An amplification value of 100 will keep the signal unchanged.

Final Amplification

This is the amplification that will affect the ending of the selection. Setting both the initial and final amplifications to the same value will amplify the entire selection the same amount.

Lock Left/Right

Left and Right channels may be amplified at separate values. If the Lock is checked, then the scroll bars for the left and right channels are locked to the same value. Effects such as panning from left to right can be achieved by choosing separate values for the left and right channels.

Logarithmic Fades

Also known as Power fades. When checked, the power of the signal fades at a constant rate. When not checked, the sample values fade linearly. As seen on screen, linear fades look like a flat slope, while power fades usually look like a hill that starts steep and gets less steep as time goes on (or the opposite depending on whether you are fading in or out).

dB Scale

When checked, amplification values are entered in decibels, otherwise they are entered as a percent of the original waveform.

DC Bias Adjust

Adjust the waveform so it is centered on the center line (0 %). If samples are recorded with a DC Bias, they will appear to be above or below the center line. They must be centered before doing other waveform transformations, and choosing this will center the wave properly. To skew the entire selected waveform above the center line, enter the percentage to move the waveform up in the adjustment box. For example, 50% will move the entire waveform up half way, and a -50% will move it down half way.

Normalize

Pressing the Normalize button will calculate the greatest amplification for the sample that will *not* result in clipping when set to 100%. If the left and right scroll bars are not locked, separate left and right values will be computed, potentially amplifying one channel more than the other. To normalize to less than the maximum range, enter the percentage of maximum to normalize to. For example, choosing 50% will compute values needed to amplify the file no more than 50% of maximum, resulting in a 3dB attenuation from maximum output. If two sounds normalized to 50% are overlapped, the resultant wave is guaranteed not to exceed the boundaries, and will not clip. All this button does is recalculate the amplification values for you based on how much normalization is needed. To normalize in one step, use the [Normalize](#) function.

To achieve a fading in effect, choose an initial amplification of 0, and a final amplification of 100. For fade outs, do the opposite by setting the initial to 100 and the final to 0.

Note: This Normalize button only calculates the values needed for the desired normalization. If you are recording a script, only the final values will be remembered. If you want to add normalization to a script, use the Normalize function instead.

This function supports [Presets](#).

Fade In Preset

Initial amplification is set to zero, and final is set to 100 for a fading in effect.

Fade Out Preset

Initial amplification is set to 100, and final is set to zero for a fading out effect.

Pan L->R, Pan R->L Preset

Initial and final values are set so that the sound starts at one channel and pans to the other.

Echo... echo... o...

Create echoing and reverb effects. This function creates a continuous echo. Each successive echo decays in amplitude by the falloff ratio. To create the effect of a single echo, use the [Delay](#) function instead.

Falloff Ratio

Each successive echo will be a certain percentage less than the previous one. Choosing a falloff ratio of zero would result in no echo at all, while choosing a ratio of 100 would produce an echo which never gets quieter.

Delay

This is the number of milliseconds to place between each echo. A delay of 100 milliseconds is equivalent to a 1/10th of a second pause between echoes. Choosing very small values of delay produces quite interesting effects.

Initial echo volume

This is the volume at which the echoes will be mixed with the original sample. Choosing smaller percentages (30% or so) is nice if the effects of the echoing at 100% make the sound incomprehensible.

Continue beyond selection

Choosing to continue beyond selection will echo the highlighted selection over the rest of the unhighlighted area, stopping at the right-hand edge of the wave that can be seen in the window. If the window is zoomed in, the echoing will stop before the end of the file, since it will stop at the right hand side of the portion on screen. By using this option a single word, for example, can be highlighted and echoed over other audio, without echoing the other audio as well.

Echo Left to Right

Selecting this option will make the echoes travel back and forth between the left and right channels.

If you wish to echo the right channel only, select an initial echo volume of 100% for the right, and 0% for the left.

Equalizer

The echo "quick filter" lets you choose approximately which frequencies get removed from the echo first. A setting of zero will leave the frequency band unchanged. You can choose the frequencies that are "absorbed" as the echo progresses. The echoed sample is re-filtered through the quick filter on each successive echo. Setting all values to zero turns off the equalization, since no frequencies are to be absorbed.

This function supports [Presets](#).

Reverb

Create high quality reverb effects. Use this function when you are looking to reproduce the effects of a certain environment - everything from your coat closet to a grand amphitheatre. Unlike echo, which generates specific echoes at specific times, the reverb function creates a very much spread out, random phase trailing of the original audio and no specific echoes can be heard at any particular time. The effect is very warm and natural. To simulate specific rooms that have echoes and reverb, use the Echo function first to get the 'size' of the room sound, then use Reverb to make it sound more natural. This function is ideal for converting Mono audio to sound as if it is Stereo. Converting a Mono sample to stereo where both the left and right channels are identical should be used as the source, then by adding some reverb, even as little as 300ms, will open up the sound so it is perceived of as being in true stereo.

Total Reverb Length

This is the total length of the reverberation. The signal will trail off and finally cut out at about -96dB after this amount of time. Values below 400 produce a small room environment. Values between 400 and 800 simulate medium sized rooms, and values above 800 simulate concert halls up to giant amphitheatres at delays around 3000 ms.

Attack Time

The amount of time it takes for the reverb to gain full strength is known as the attack time. For smaller reverb lengths, the attack time should be smaller. In general, a value of about 10% the total reverb length works well. But interesting effects can be gained by using longer attack times with shorter reverb lengths for very subtle reverb. Or, very short attack times can be coupled with long reverb lengths for other special effects.

High Frequency Absorption Time

In natural environments, higher frequencies are attenuated more than lower frequencies. Using this parameter, the exact time it takes for the highest frequencies to be completely cut out can be chosen. Faster Absorption times simulate rooms that are occupied and have furniture and carpeting like night clubs or theatres. Slower times (especially over 1000ms) simulate more empty rooms, like gymnasiums and empty auditoriums, where higher frequency reflections can be heard.

Perception / Timbre

This is another parameter to help give subtle qualities to the environment making it sound more realistic. It can be thought of as changing the width of the room and adjusting other room irregularities. With lower values, the reverb is smoother without as many distinct 'echoes'. Higher values cause more variation in the reverb amplitudes and add more spaciousness to the reverb by creating distinct reflections over time. In general, higher values (up to 60%) can be used for simulating large rooms, and lower values (down to 0%) for small rooms. But these are only suggestions. Interesting canyon effects can be created by setting this value to 100, and using a total reverb length of 2000 or more.

Mixing - Original Signal

This is the amount to mix the original signal into the final result. If you are trying to achieve some special effects with reverb, you may want to reduce the volume of the original signal. Or, if the reverb is so great that audio begins to clip, reduce both the original signal and the reverb mixing strength. In general, the more reverb you add, the lower the original signal volume should be. In most cases, a value of 90% or so should be fine.

Mixing - Reverb

This is the amount to mix the reverberated signal into the final result. A value of 100% is most natural, but you may wish to decrease this for a reverb that exists more in the background, or increase to simulate being far away from the audio source where only the reverb can be heard in greater strength than the original audio.

Combine L&R

In general, this should be checked for more realistic reverb, and faster calculation times. When checked, the left and right channels in a stereo source are combined before reverb is performed. This should especially be checked if you know that both channels are identical, otherwise it is just a waste of computer time. When Combine L&R is not checked, separate stereo reverb is calculated for each channel individually. The original signal will remain in the respective channels, but the reverb will carry through to both channels equally. With stereo audio where there is different information in the left and right channels, this box should not be checked. The stereo audio will be dramatically enhanced by the reverb, and sound fuller and more rich in most instances. Also, when this box is not checked, calculations take exactly twice as long to compute, since separate reverb is being calculated for the left and right channels before being recombined to the final stereo output.

I have built a few presets to get you started. But for best results, experiment with the different parameters and I'm sure you'll find just the reverb you are looking for. The "Large Occupied Hall" gives a very nice live theatre atmosphere. The "Concert Hall Light" setting gives a nice professional performance reverb, enhancing a non-reverberated vocal singing track quite nicely.

This function supports [Presets](#).

Flanger

What *is* flanging you ask? Just try it out and see! The term is coined from the flanging mechanism on the old style tape recorders which, when fiddled with, would slow down the playing of the tape, and speed it back up again when desired. That is how they got those funky psychedelic sounding recordings in the 60's. Here's how you can do it today.

Original - Delayed slide

This slide decides at what proportions to mix the original and flanged signal. If the Original is at 100%, no flanging is heard. If the Delayed is at 100%, a cute wavering (like a bad tape player) sound is heard. Portions of both signals need to be present for there to be canceling out, and reinforcing of wave patterns between the two signals.

Initial Delay

Flanging will start with the delayed signal this many milliseconds behind the original.

Final Delay

Flanging will end with the delayed signal this many milliseconds behind the original. If the delays are the same, the effect disappears, since the delayed signal will not change.

Stereo Phasing

The right channel can be at a separate delay than the left channel. A phasing of 180 will put the right channel at the initial delay value when the left channel is at the final delay value, and vice versa.

Rate settings

The **Frequency**, **Period**, and **Cycles** settings are all interrelated, and refer to the rate at which the delay cycles between the initial delay and the final delay. The flanging will cycle *frequency* times per second, or *period* seconds per complete cycle, or a total of *cycles* complete cycles over the entire selection. Various effects can be heard by using different settings. For example, if 0.5 cycles is chosen, the selection will start with the initial delay, and end with the final delay. If a frequency of 4 is chosen, the flanging will cycle from the initial delay to the final delay and back again 4 times per second.

Invert

Invert the delayed signal when flanging, which causes the waves to cancel out periodically, instead of reinforcing. If the mixing is at 50/50 then whenever the delay is at zero, the waves will cancel out to silence.

Special EFX

A mixture of both normal and inverted flanging, with the delayed signal summed, and a future signal subtracted out. So this option will mix not only a delayed signal, but a future one as well.

Sinusoidal

If checked, the transition from initial delay to final delay and back will follow a sine curve. Otherwise, the transition is linear, and delays from the initial setting to the final setting at a constant rate. With sinusoidal checked, the signal is at the initial and final delays more often than it is at delay in-between.

By trying different combinations of Invert, Special EFX, and Sinusoidal, you should be able to create just the effect you want. These three options give a lot of control over the flanging effect, so experiment with them all!

This function supports [Presets](#).

Select Entire Wave

This will select the entire waveform from zero to the end of the wave. It makes no difference if the view is zoomed in or not. Double-clicking on the waveform selects the visible portion of the wave, while choosing **Select Entire Wave** (or pressing CTRL+A) will select the entire waveform independent of the portion being viewed.

Wave (Brainwave Synchronizer)

Audio may be Waved to produce files, that when listened to with stereo headphones, will put the listener into any desired state of awareness. For example, by listening to waved files, you can easily achieve states such as deep sleep, theta meditation, or alpha relaxation. Because of the nature of this function, it only works on **Stereo** waveform data, and to be effective, must be listened to with stereo headphones. The Wave function spatially locates the audio left and right, in a circular pattern over time. In order to spatially encode the signal, either the left or right channel is delayed so that the sounds will appear at each ear at different times, tricking the brain into thinking they are coming from either side. When this is done at frequencies of 3Hz and above, the brain will start synchronizing at the same frequency, increasing its output of Delta, Theta, Alpha, or Beta frequencies.

Audio can also be panned binaurally by delaying one channel with respect to the other, and changing this delay over time at the rate specified.

LOW SETTINGS

These settings all correspond to the lower part of the graph. If points are dragged to near the bottom of the graph, these settings will be active.

HIGH SETTINGS

And these settings correspond to points near the top of the graph.

Frequency Graph

Time is represented along the horizontal, so as you go to the right of the graph, you are setting the frequency characteristics of the highlighted sample later and later in time. The settings chosen will vary between the low settings and high settings depending on where the graph dictates the signals should be.

Click on the graph to add new control points. Drag a control point up or down, or off the screen to remove. Choose the highest and lowest frequencies that are represented on the graph with the scroll bars. Gliding about 4 to 5 Hz over 2 minutes works nicely. If large variations are done in short time spans, the effects are not as pronounced. For example, after 5 minutes of Theta waves, if 30 seconds of alpha waves are generated, and returned to theta, the listener will become slightly awake, and aware of his surroundings for that brief moment. The effect is like all of the sudden changing gears, and you stop thinking about whatever it was you were thinking about, and become aware that you were thinking about it, but aren't any more.

Frequency

This is the brainwave frequency that will be encoded. Different brainwave frequencies will stimulate the brain to sync to differing levels of consciousness (e.g. sleep, meditation, awakeness, etc.). See the bottom of this article for more information on specific frequencies.

Intensity

This is the intensity of the brainwave encoding. Higher intensities work well with lower brainwave frequencies. Beta waves should have intensities below 25 or so, while Delta waves work better with intensities above 60.

Centering

You may choose to have your brain think the synchronization frequencies are coming from the left or right. This may affect the left or right hemispheres more intensely, but that's only a guess. Mixing a file that has been waved to the left with one that has been waved to the right (in the same frequency range within 2 Hz) has interesting effects.

Musical Source

If the selection being waved is musical, checking this will calculate the wave patterns in such a way as to eliminate clicks and pops. If the source is noisy (waterfall, ocean, nature recordings, etc.) do NOT check this. If you do, it will actually add interference. Since noise is based on "randomness", the clicks and

pops are inaudible.

Smooth Wave

When checked, the actual audio appearing at the left and right channels is smoothed out, but the spatial encoding is identical. The left and right channels will delay and un-delay following a smooth curve such that the delay difference between the left and right channels follows a sine wave, and the brain will hear the audio traveling around the head in a circle. When Smooth Wave is not checked, the net delays are the same, but are achieved by holding one channel constant (at no delay) while the other channel is delayed following half a sine wave. Then the other channel is delayed while the first is held constant. The boundary between holding constant and delaying is discontinuous in that the dD/dt (difference in delay over time) jumps from zero to a positive delay value without hitting any values in-between. When Smooth Wave is checked, the dD/dt is always continuous. This will also cause less noticeable distortion in either channel when heard independently.

For special spatial panning effects, choose wave frequencies of 1Hz or less. A mono source (left and right the same) will appear to move from left to right and back at period of $1/\text{frequency}$. For example, a frequency of 0.1Hz will pan the audio in a "full circle" over the period of 10 seconds.

Please read on for more information about [Brainwave Synchronization Files](#)

Channel Mixer

On stereo waveforms, the channel mixer allows one to have total control over the left and right channels. The default values will leave the wave unchanged. For mono waveforms, the wave is inverted (that is, crests become valleys, and valleys become crests).

New Left Channel

The slide bars give the percentage of each channel, left and right, that will go into the final wave after mixing. Choosing an L of 0, and an R of 100 will make the left channel equal to the right channel.

New Right Channel

These two slide bars do the same, but for the right channel.

Invert

Choosing invert for either channel will invert the channel. Peaks become valleys, and valleys become peaks. By inverting *both* channels, there will be no perceived difference in sound when listened to. But, inverting only one channel will greatly change the sound when listened to.

This function supports [Presets](#).

Vocal Cut Preset

This will sum the left channel with the inverse of the right, and place the result into both channels. On music where the vocals are heard equally loud on both channels, the vocals will disappear, or come close to disappearing.

By playing with the combinations, effects of swapping channels, creating a mono sounding wave that is equal to the left, right, or a mixture of both channels, and creating waves whose left channel is the inverse of the right can be done.

Invert

This function simply inverts the samples, so all positive offsets are negative and all negative offsets are positive. On stereo waveforms, both channels are inverted.

Compressor/Expander/Limiter/Noise Gate

The compressor function varies the output level based on the input level. This allows one to expand or compress the dynamic range of a sample, limit the dynamic range so all audio is at roughly the same level, or create a noise gate where all audio below a certain level is clipped to zero. This is all accomplished by use of a transfer function that is drawn using the **graph**. The graph depicts input level along the x-axis (left and right) and the new output level along the y-axis (up and down). A line from lower-left to upper-right (default) leaves the signal unchanged, since every input value goes to the exact matching output value. Other weird transfer functions can be drawn as well, for example, boosting all input that has a level of around -20dB, and leaving everything else unchanged. Or, drawing an inverse line (a line from upper-left to lower-right) will dramatically boost low amplitudes while dramatically suppressing high amplitudes, that is, all quiet sounds are loud, and all loud sounds are quiet.

Invert

The invert button will change the graph to one that will function as the exact opposite. For example, if a transfer function with a compressor characteristic is being displayed, pressing Invert will change the graph to one with the corresponding expander characteristic. For a graph to be invertible, it must have points in the two corners (-100,-100 and 0,0) and it must be always increasing in output (i.e. you cannot go down in output volume as you go from left to right). All segments must be sloping upwards from left to right.

Attack Time

Attack time determines the time it takes for the new output signal to reach the proper output volume. If there is suddenly a quiet portion that drops 30dB, it will take this much time before the output actually drops to its corresponding volume level. If the sum of Attack and Release times is too short (less than about 20 ms total), audible effects can be heard, such as a "vibrating" sound at a frequency of $1000/\text{time}$. So if the Attack and Release times are each set to 5 ms (making 10 ms total), then a vibrating sound at 100Hz can be heard. Thus, a total value of about 30 ms is about the lowest you can go without getting these effects.

Release Time

This is the time it takes the end of a previous output level to reach the proper output volume. For example, where the Attack is the time it takes the start of a pulse to reach the desired output volume, the Release is the time it takes for the end of the pulse to reach the desired level.

Samples/Group

The number of audio samples to group together into one volume level change. A value of 1 is the best, so each sample gets its own volume change. Larger values will change that many samples together at a time. You can go larger without noticeable changes in quality. The only reason for using larger values would be for speed, as larger values calculate much faster. Use larger values for pre-viewing how a compressor is going to sound, then Undo, and use a value of 1 when the compressor is set just the way you want it.

Joint Channels

In Stereo, each channel can compress independently, sometimes causing the surrounding background noise to get louder on one channel at a time, which may sound strange. For example, a loud drum beat in the left channel will make the background noise sound louder in the right than in the left. If Joint Channels is checked, both channels are used to find a single input dB value, and both channels are amplified the same amount, together. For example, a loud drum beat on the left channel will cause the right channel to go quieter as well if compressing.

Compressors are used for the compression of the dynamic range of an audio signal. It is generally an amplifier with two gain levels: the gain is unity for input signal levels below a certain threshold, and less than unity for signals with levels above the threshold. Compressors can be used to eliminate the variations in the peaks of an electric bass output signal by clamping them to a constant level, thus providing an even solid bass line. To maintain the original character of the instrument it is necessary to

use a compressor with a long A/R time compared to the natural decay rate of the electric bass. Compressors can also be useful to compensate for the wide variations in the signal level produced by a singer who moves frequently, changing the distance from the microphone. **Limiters** are compressors with a compression ratio of 10:1 or greater because their output levels are essentially clamped to the threshold level. A limiter can be used to clamp all audio to a prescribed output level, or just all audio above a certain threshold.

Expanders are used to expand the dynamic range of an audio signal, opposite of the compressor. It can also be considered an amplifier with two gain levels: the gain is unity for input signal levels above a certain threshold, and less than unity for signals with levels below the threshold. The expander is used to expand the dynamic range of an audio signal by boosting the high-level signals and attenuating the low-level signals.

Noise Gates are a special type of expander that can be used to reduce noise below a threshold level. It attenuates heavily signals with levels below the threshold. It is used to totally cut off the signal level during a musical pause so as not to pass the background noise present. It can also be used to silence the pauses in speech.

RMS mode

This is a new graph interpretation method that more closely matches the way people hear volume. This will cause the output to be exactly the RMS amplitude specified in the graph. For example, a limiter (flat horizontal line) at -10dB will cause the RMS amplitude of the result to average -10dB (where 0dB is a maximum amplitude sine wave without clipping).

Peak mode

This is the method that has been used in previous versions of Cool Edit. This method is a little more difficult to use, but it equates to the RMS value times two. That is, if the RMS value was -20dB, then the equivalent peak value would be -40dB. This occurs because the RMS value calculated was mapped to a peak sample value for output. This method is basically here for backward compatibility with previous versions of Cool Edit.

Noise Generator

Generate random noise in a variety of colors. Each color has its own characteristics. One use for generating noise is to create a waterfall-like sound which is ideal for Waving. It is also great for making weird effects by flanging and equalizing.

Color

Noise can be a variety of colors, which describe its spectral composition.

Brown noise has a spectral frequency of $1/f^2$. Which means, in English, that there is much more low-end, low-frequency components to the noise, which results in thunder and waterfall like sounds. Brown noise is called that because, when viewed, the wave follows a Brownian motion curve. That is, the next sample in the waveform is equal to the previous sample, plus a small random amount. This gives the appearance of a mountain range when graphed. The wave pattern is very predictable.

Pink noise has a spectral frequency of $1/f$ and is found mostly in nature. It is the most natural sounding of the noises. By equalizing, rainfall, waterfalls, wind, rushing river, and other natural sounds can be generated. Pink noise is exactly between brown and white noise (which is why some people used to call it tan noise, but pink was more appealing). It is neither random, nor predictable. It has a fractal like nature when viewed. When zoomed in, the pattern looks identical to when zoomed out, except at a lower amplitude.

White noise has a spectral frequency of 1. In other words, equal proportions of all frequencies are present. Because the human ear is more susceptible to high frequencies, it sounds very "hissy". White noise is generated by choosing random values for each sample.

Style

Noise can be generated in a variety of styles for your listening pleasure.

Spatial Stereo noise is noise generated by using 3 unique noise sources, and spatially encoding them to appear as if one is coming from the left, the other from the center, and the last from the right. When listened to with stereo headphones, the mind perceives sound coming from all around, not just in the center. To choose the distance from center of the left and right noise sources, you can enter a delay value in microseconds. About 900 to 1000 microseconds corresponds to the maximum delay perceivable, and a delay of zero is identical to Mono noise (left and right channels are the same).

Independent Channels noise is generated by using 2 unique noise sources, one for each channel. The left channel's noise is completely independent of the right channel's noise.

Mono noise is generated by using 1 noise source, with the left and right channels set equal to the same noise source.

Inverse noise is generated by using 1 noise source as well, but this time with the left channel's noise exactly inverse of the right channel's noise. When listened to with stereo headphones, the effect is that of the sound coming from the center of the listener's head instead of out in space somewhere.

Intensity

With higher intensities, the noise becomes more erratic, and sounds harsher and louder.

Duration

This is the number of seconds of noise to generate. If long periods of noise are desired, it is faster to generate a short period of noise (about 10 to 20 seconds), delete excess noise at the beginning and ending of the noise so that the waves are starting and ending at the midpoint, copy, then loop paste as many times as needed.

If a selection range is highlighted, it is **not** replaced by the noise generated. Noise gets inserted at the insertion point represented by the yellow arrows.

Cool Scripts

Scripts are similar to Macros. Your exact mouse moves, and tweaking of parameters is not stored, only the final result when you click "OK". Undoing something is recorded into the script, since you may have copied something, then undid it, which can be a very useful technique.

Multiple scripts can be kept in one script file, and identified by name. At any time, you can edit the script file directly to take out steps, rename scripts, remove unwanted scripts, etc.

There are various types of scripts, which depend on when you initiated the recording:

- Scripts that start with File->New, and always start with a blank, empty waveform.
- Scripts that start when a waveform is opened, and work at the current sample rate, etc. Actions begin at the insertion point in the waveform, and may affect any part of the entire wave if present.
- Scripts that start with a highlighted waveform portion. All actions in the script pertain only to the portion that is highlighted, leaving the rest of the waveform untouched.

Scripts that run during all of the above conditions will be displayed, but only the ones recorded under the same circumstances will be allowed to run. In other words, if a script recording started when a portion of a wave was highlighted, then you will only be able to run that Script when something is highlighted.

Scripts are very useful for remembering how you generated a particular sound effect. Use the script to reproduce the sound effect without having to save the entire waveform. This is especially useful when generating large brainwave "theta" files, which can take monstrous amounts of space. By generating the file once, with the scripting turned on (record), you can generate the file again at any time in the future, and save all that hard drive space. You can also pass along scripts to your friends across email or BBS systems, since they take nearly no memory to store.

When running a script, you can either stop at each dialog box, or have the script automatically run through completion by using the "Stop at Dialogs" checkbox. Stopping at each dialog box is handy if you wish to 'tweak' the parameters while the script is running.

After recording a script, you may enter a description at the bottom of the dialog to go with the script you just recorded. This description will appear when the user of the script highlights the script to run. Note: the only time you can edit the description is after recording, not before, and not after it has been added to a script collection file. But, you can still edit the description at any time by pressing the **Edit** button to edit the text file directly.

A single script can be run on a batch of files by pressing the **Batch Run** button. For more information, see [Batch Processing](#).

Pause at Dialogs - At each dialog, the script will stop to allow you to modify the values to the function. Pressing Cancel at this point will stop the script, pressing OK will continue it.

Alert when complete - When the script is finished, a dialog box will signal the completion of the script if this option is checked.

Execute Relative to Cursor - When running a script that was recorded when a waveform was loaded but there was no highlight, it can be run by playing back all the operations relative to the beginning of the file or to the beginning of the cursor. For example, the Sound Effects scripts require you open a waveform (it can be blank) first. Checking this option will insert the effect at the cursor, otherwise the effect will be inserted at the start of the file.

Important Note: Other buttons and functions are not disabled while the script is running. Therefore, do not use the other functions until the script has stopped playing.

FXNS2.SCP Sample Collection

Description These are five sample functions I came up with. Cross Fading is useful if you are going to

loop the sample. The last portion of the sample is overlapped with the first portion, and the amount of overlap is different for each Cross Fade script. Full cross fading fades the last half of the sample with the first half. Soft cross fading fades the last 5% with the first 5%, and hard cross fading fades the first 0.4% with the last 0.4%. Make Piano Keys will take the highlighted sample and stretch and compress them to vary the pitch, making 13 copies of the original, each at a different pitch. Each pitch is assigned to a key on the keyboard through the cue list. This turns your keyboard into a simple sample player. Reverse Echo is just that -- the echo function, but the echoes go in reverse.

How To Use These sample functions work on a highlighted selection. Open a waveform, and highlight the portion you wish to operate on, then run the script.

SNDEFX2.SCP Sample Collection

Description Here are some nifty sound effects, and a small song (very small) I wrote. If you run the Cool Song script, you can then go to the Music function, and enter a name for the song to save it under. This script generates a short note, and then uses the Music function to build the song. The other four effects are just weird effects using the tones or noise functions with other transformations. If you've watched "Dr. Who", you may recognize the Cool Lasers sound effect.

How To Use These sound effects work in a currently opened waveform. Open a New (blank) waveform in any sample rate setting you desire for the quality you would like, and run a script. These sound effects can also be inserted into an existing waveform, and will be inserted just as if the Paste command were used.

MINDSNC2.SCP Sample Collection

Description Included are four "Tones" synchronization scripts, which have a binaural beat pattern (two differing tones in each ear) overlaid with the corresponding "Waved" pink noise. Choose Loop Play and listen to the audio as long as you like. Each Tone script stimulates a different brainwave frequency, from Delta to Theta to Alpha, and an "Earth" tone of 7.83Hz. The "Music" scripts have "Waved" music overlaid with the pink noise for a relaxing effect. The Creativity Theta Session is similar to the sample theta session described with the Wave function, and lasts 1/2 hour. The session starts at Alpha, goes down to Theta, and stays there with a few bursts into Alpha and back.

How To Use Open a new blank waveform of any Stereo sample setting you wish. I suggest using at minimum 22 KHz 16-bit stereo, but the synchronization effects will still work at lower sample rates and 8-bit. Once you have a blank waveform to work with, run one of the scripts.

Batch Processing

A single script can be run repeatedly over a group of source files. The script must have been recorded in a "Works on Current Wave" mode, that is, before the script was recorded there must have been an open waveform (perhaps blank) and no highlighted selection. The **Batch Run** button will only be selectable if a script of this type is selected from the Scripts list.

Any number of **Source Files** can be chosen as long as they are in the same directory (sorry about the limitation). Press the **Browse** button to choose these wave files. The wave files can also all be in different formats if desired.

After each file has had the script run on it, it will be saved to the **Destination Directory**. You can choose the **Output File Format** that all the waves will be saved as, as well as enter the appropriate options for the file format if the format supports options.

File names can be modified slightly before being saved. The filename extension will change to that of the file format being saved automatically (e.g. *.AIF). If another filename extension is desired, or some modification to the filename portion is desired, the filename template can be modified. Use the question mark '?' to signify that a character does not change, and a '*' to denote the entire original file name or entire original file extension. Here are some examples of how filenames will be saved given the original file name and the filename template:

zippy.aif	*.wav	zippy.wav
toads.pcm	q*.voc	qtoads.voc
funny.out	b???????.*	bunny.out
biglong.wav	?????.wav	bigl.wav
bart.wav	*x.wav	bartx.wav

Choosing **Overwrite existing files** will always perform the script on the file in question and save it to the destination, even if a file of the same name already exists at the destination. If this box is unchecked, and the destination filename already exists, the batch will not even attempt to run the script on the file in question, but skip it instead.

In most cases, you can check the **Disable Undo** option to disable the Undo function during the batch run. Unless the batch was written expecting the Undo function to be enabled, this is a completely safe thing to do, and it speeds up processing because undo information does not continually need to be saved.

If a source file is unreadable, or in a RAW type format without any header information, then the batch needs to know what file format to assume for the data. This ensures that the batch will run continuously without interruption by dialogs asking for input data formats on head-less data.

Customizable Toolbar

If you wish to re-arrange the toolbar buttons, or make certain buttons visible when the toolbar is shorter, you can do this by modifying your cool.ini file. Cool.ini is located in your Windows directory. In a section of the INI called [Toolbar] you can define button order, and priority (which buttons get displayed first if there is not enough room for all buttons on the toolbar). I apologize for there not being a visual way of re-arranging buttons at this point, but it will be possible in the near future. Until then, you must edit cool.ini.

The easiest way to quickly get a button that you desire on the toolbar *when* you want it is to:

Swap the priority ratings of the button you want to see with a button you do *not* want to see. For example, if the Paste button is more important than the Loop button, and Paste has priority 37, while Loop has priority 10, then swap the priorities by giving Paste a priority of 10 and Loop a priority of 37. This way, the Paste button will be displayed before the Loop button as the toolbar grows in length by resizing the application.

Positioning buttons is accomplished by changing the order in the list. You will notice that each button has an Item and Priority key associated with it. For example, the Loop button has Item33=Loop and Priority33=10. If you wanted to move this button to the left, give it a lower key value, such as Item9=Loop and Priority9=10. When you do this, make sure that the original button associated with Item9 is changed as well, since there cannot be two keys of the same name. The easiest thing to do is just to swap, and give the Item9 and Priority9 values to Item33 and Priority33. For example, if Item9=Copy and Priority9=36, then you would change it to read Item33=Copy and Item33=36. You could also bump up all key numbers by one until you bump Item32 to Item 33. (i.e. Item32... becomes Item33..., Item31... becomes Item30... ,etc.). Again, in the future, all this will be automatic and done visually by dragging and dropping the buttons.

If your toolbar becomes messed up, and you want it back to the original, just delete the entire [Toolbar] section along with all Item and Priority items, and Cool will re-build the section with the defaults.

About the Creator

That's me, David Johnston... I have been working with various sound programs since about 1991, and have been toying with electronic noise makers (synthesizers) since the 70's. In high school I was even a pseudo-member of the band with a home-made programmable "sequencer" that I used to make Pac-Man noises for the song Pac Man Fever. I also started programming while in high school, when the school got a Commodore PET computer. Shortly thereafter I bought an Atari 800 (summer 1982) and started my programming career. I have since programmed on Apple II's, Commodore 64's, Macintoshes, IBM PC's (the original on up to today's), numerous mainframes and workstations. I attended Eastern Montana College for my first two years, then moved to the Seattle area and got my BS in Computer Science and Engineering from the University of Washington in 1991. I now work for a large software company, and write fun programs on my home PC when I get a chance. This is one of them! With the ShareWare registrations I have been able to purchase a Pentium 90, so development on Cool Edit has thus accelerated. I gave my ol' 386-25 to my brother-in-law who is starting college.

Syntrillium Software Corporation was founded by Robert Ellison and myself, and we intend to continue bringing innovative and imaginative software to all those who like to have *cool* software on their computers.

We offer another ShareWare program which is a kaleidoscope screen saver that actually is really cool! It will even change patterns based on audio for your own personal "laser show". Another useful utility we offer is an RPN calculator for Windows which I use almost every day for one reason or another. A swirling illusion screen saver is also available that makes walls appear to breathe, or objects in general appear to swish around and distort once you look away from the patterns.

We have tried to include all the wave editing features we have found useful, and are welcome to any suggestions for new waveform transformation functions, user interface, file formats (helps if you can provide a description of the new file format you want), and features. There are still big plans in for upcoming versions of Cool Edit. The "wave" function in this program is quite unique and I personally hope you check it out. It works only with stereo waveforms, and is the same function as in the freeware program out called MINDSYNC (wave.exe or wave87.exe) which I finished in July of 1992; see [Wave](#) for more details. This program should be compatible with every sound card on the market that has Windows drivers available for it. Please read the special notes in the Contents section if you have problems.

We hope you find lots of uses for Cool Edit, and find it fun and enjoyable to use. Feel free to let us know what you think of it. It's because of input from numerous users out there that the program has evolved to what it is today.

The Demonstration version of Cool is limited to using only 2 groups of functions per editing session. There are no limits on length of files, time in use, or number of uses though!

Syntrillium Software Corporation
P.O. Box 60274
Phoenix, AZ 85082-0274
USA

FAX: 602-941-8170
email: Syntrill@aol.com

You may send USD \$50 for a registration key that allows the use of all the effects and features of Cool Edit.

You may send USD \$25 for a 'Lite' version (for those on a budget or for those who don't need all the extra bells and whistles). The particular bells and whistles left out are: Compressor, Delay, Distortion, Echo, Echo Chamber, Envelope, Flange, Noise Reduction, Quick Filter, Reverb, and Generate DTMF Tones.

And those who wish to give that little bit extra will be placed on our *preferred* user list for USD \$100. Preferred users will be eligible for special offers, and other goodies.

If you use this program a lot, I hope you consider registering it, even if you've never sent in money for a ShareWare program before. It's quite easy, just write a check out to me and drop it in the mail. When you register, you will be sent a registration number that will *unlock* the software, allowing unlimited use of all functions all the time, or you will receive the Lite version on a 3.5" floppy. If for any reason you do not believe the program is worth \$50, send whatever you *do* think it's worth, and we will send you a registration number anyway. Please include a reason why you don't think it's worth the asking price though. If you would like a printed manual sent to you, just ask and we'll send one down. A written manual is free to all registered users, but it usually takes about a month or so after releasing before manuals are ready.

Australian customers can register through the WOFTAM BBS at (03) 761-9904 or write:

Woftam BBS
P.O. Box 100
Mt. Evelyn. Victoria
Australia 3796

If you have the Demo version, and want to Register, see the registration form below.

REGISTRATION INFORMATION

Cool Edit v.1.51 Registration Information

You can register Cool Edit by fax, by mail, or through the CompuServe ShareWare Registration service. Regardless of the method you choose, Syntrillium will e-mail or fax (your choice) your registration number within 2 business days and send you a letter confirming your registration along with a receipt if you use a credit card. If you have any questions about ordering, please do not hesitate to e-mail Syntrillium Software at syntrill@aol.com.

1) **FAX:** Fill out the [registration form](#) and fax it with your Visa or Mastercard information to Syntrillium Software at 602-941-8170.

2) **MAIL:** Mail the [registration form](#) with a check drawn in US dollars or your credit card information to:

Syntrillium Software Corporation
P.O. Box 60274
Phoenix, AZ 85082-0274
USA

3) **CompuServe:** Use GO SWREG, select Register ShareWare, answer the questions that CompuServe asks, specify a Registration ID of 6445 when prompted, and then select Display Selected Titles.

REGISTRATION FORM

Cool Edit v.1.51 Registration Form

Name: _____

Address: _____

City: _____ State/Prov: _____ Zip: _____

Country: _____

Email, CIS, AOL: _____

Phone #: _____

FAX #: _____

Visa or Mastercard #: _____ Exp Date (required): _____

Choose the level of registration you prefer:

\$50 Basic Registration: You will receive a registration number that will unlock your copy so you can use any function at any time.

\$100 Preferred Registration: You will receive a registration number and automatically receive ShareWare updates as they are released. You are also placed on our *Preferred* list which will entitle you to special offers, discounts, and other goodies in the future.

\$25 Lite Version: You will receive a registration number that will turn your copy of Cool Edit into Cool Edit Lite. If you have no need for all the fancy effects offered by Cool Edit, then register at this level. This version has all the features of Cool Edit except for the following: Compressor, Delay, Distortion, Echo, Echo Chamber, Envelope, Flange, Noise Reduction, Quick Filter, Reverb, and Generate DTMF Tones.

Please inform me of updates and where to find them on the net.

You may answer these if you like! Just fill in the blanks.

Where did you find Cool Edit? _____

What do you use Cool Edit for? _____

Brand of PC: _____ OS (win 3.1, NT...): _____

CPU (386, 486...): _____ Co-processor? _____

Sound Card: _____

RAM: _____ Hard Drive size: _____

----- THANK YOU!!!! -----

Syntrillium Software Corporation
P.O. Box 60274
Phoenix, AZ 85082-0274
USA

Fax: 602-941-8170
Email: syntrill@aol.com

Cool Edit v.1.51 Upgrade Form

Why upgrade? You may want to upgrade from the Lite or Basic version of Cool Edit to a higher registration level. Currently registered users of Cool Edit may receive an upgraded registration number for ShareWare versions 1.51 and higher free of charge. If you currently are registered for Cool Edit Lite and would like to upgrade to the Full version, or if you would like to upgrade from a Basic registration to the Preferred level, then check the appropriate box below.

Name: _____
Address: _____

City: _____ State/Prov: _____ Zip: _____
Country: _____

Email,CIS,AOL: _____
Phone #: _____
FAX #: _____

Visa or Mastercard #: _____ Exp Date (required): _____

Choose the upgrade you would like

- \$0 Updated Registration Number:** You will receive an updated registration number that is guaranteed to work on future versions of Cool Edit. If you wish to use future versions of Cool Edit at your *current* registration level, then you should send in for the updated registration number that works on versions 1.51 and up.
- \$25 Upgrade from Lite to Basic Registration:** You will receive a registration number that will upgrade your copy so that you can use all the functions of the full version of Cool Edit, including Compressor, Delay, Distortion, Echo, Echo Chamber, Envelope, Flange, Noise Reduction, Quick Filter, Reverb, and Generate DTMF Tones.
- \$50 Upgrade from Basic to Preferred Registration:** You will receive a registration number and automatically receive ShareWare updates as they are released. You will also be placed on our *Preferred* list which will entitle you to special offers, discounts, and other goodies in the future.
- \$75 Upgrade from Lite to Preferred Registration:** You will receive a registration number that will upgrade your copy so that you can use all the functions of the full version of Cool Edit, including Compressor, Delay, Distortion, Echo, Echo Chamber, Envelope, Flange, Noise Reduction, Quick Filter, Reverb, and Generate DTMF Tones. You will also automatically receive ShareWare updates as they are released, and you will be placed on our *Preferred* list which will entitle you to special offers, discounts, and other goodies in the future.
- Please inform me of updates and where to find them on the net.**

----- THANK YOU!!!! -----

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New Wave

When creating a new waveform, you must specify the waveform properties. Using higher sampling rates, stereo, or higher bit resolutions will result in higher quality sounds at the expense of requiring more memory.

Sample Rate

The sample rate describes how many times per second to take a *snapshot* of the audio. The human ear can perceive sounds up to about 17,000 cycles per second, or 17 KHz. When choosing a sample rate, frequencies of up to 1/2 the sample rate can be produced effectively. So to reproduce frequencies up to 10Khz, a sample rate of at least 20Khz must be chosen. Any sample rate may be entered in directly, or a common sample rate can be chosen from the list.

8,000 Hz	Telephone Quality
11,025 Hz	Poor AM Radio Quality
16,000 Hz	Reasonable compromise between 11 KHz and 22 KHz
22,050 Hz	Near FM Radio Quality
32,075 Hz	Better than FM Radio Quality (Some boards support 32,000 instead)
44,100 Hz	CD Quality
48,000 Hz	DAT Quality

Channels

Mono waveforms support one channel of audio information. Stereo files take twice the space because there are two channels of information represented, a left and a right channel.

Resolution

This describes the number of bits to use for each sample on each channel. Choosing 8-bit resolution will provide 256 unique "volumes". The PC-Speaker, for example, provides only 4-bits of resolution because it can support 16 unique volume levels. Choosing 16-bit resolution will provide 65,536 unique "volumes", for a 96 dB signal-to-noise ratio. Much quieter sounds can be reproduced at 16-bit resolution than at 8-bit resolution, which only has a 48 dB signal-to-noise ratio. Compact disk players have a 16-bit resolution.

Note: Certain combinations of sample rate, channels, and resolution may not be available on your system. To see the maximum capabilities of your system, look at the status window when starting the program. you may create and edit files that cannot be played on your system, you just may not be able to play them properly.

Adjust Sample Rate

You can change how Cool-Edit interprets the actual waveform data by adjusting any of the parameters below associated with the wave. Be aware that listening to Mono data in Stereo format, or listening to 8-bit data in 16-bit format will sound really really weird!

Adjusting the various parameters comes in very handy when loading in waveforms of unknown type (RAW). You can play with the various settings until the wave sounds "right".

Sample Rate

This describes how many times per second to take a *snapshot* of the audio. The human ear can perceive sounds up to about 17,000 cycles per second, or 17 KHz. When choosing a sample rate, frequencies of up to 1/2 the sample rate can be produced effectively. So to reproduce frequencies up to 10KHz, a sample rate of at least 20KHz must be chosen. Choose **Custom** to enter a sample rate not listed.

Channels

Mono waveforms support one channel of audio information. Stereo files take twice the space because there are two channels of information represented, a left and a right channel.

Resolution

This describes the number of bits to use for each sample on each channel. Choosing 8-bit resolution will provide 256 unique "volumes". The PC-Speaker, for example, provides only 4-bits of resolution because it can support 16 unique volume levels. Choosing 16-bit resolution will provide 65,536 unique "volumes", for a much higher signal-to-noise ratio. Much quieter sounds can be reproduced at 16-bit resolution than at 8-bit resolution. Compact disk players have a 16-bit resolution.

If you wish to convert your current sample to a new sample rate, please read about [Converting Sample Rates](#) for the correct procedure.

Note: Certain combinations of sample rate, channels, and resolution may not be available on your system. To see the maximum capabilities of your system, look at the status window when starting the program. you may create and edit files that cannot be played on your system, you just may not be able to play them properly.

Byte Swapping

You can change how Cool-Edit interprets the actual waveform data by swapping high and low order bytes in 16-bit mode, or by adjusting the signed/unsigned interpretation of bytes in 8-bit mode. Swapping the byte ordering comes in very handy when loading in waveforms of unknown type (raw PCM). You can try swapping the bytes if the waveform does not sound correct.

Paste Special

Waves from the clipboard can be looped, or mixed the current waves. They are inserted or overlapped starting at the current insertion point.

Volume

Use the volume slides to paste an amplified version of the clipboard wave into the current waveform. By adjusting the volume slides, single channels may be pasted.

Invert

Choosing Invert will invert the data being pasted before pasting. This is very handy in taking the difference between two samples. For example, after filtering, one can hear the audio that was filtered by Copying the selection, choosing Undo, then Paste Special with Invert checked. After auditioning, the original sample can be pasted back by just choosing Paste.

Lock left/Right

When checked, the volume slide bars are locked, so both left and right volumes can be adjusted at the same time.

Overlap

When overlap is checked, the clipboard wave does **not** replace the currently highlighted selection, but is mixed at the selected volume with the current wave. If the clipboard wave is longer than the amount selected, the wave continues being pasted beyond the selection.

Modulate

When modulate is checked, the clipboard wave is modulated with the current wave. This is like overlapping, except that the values of the source wave and clipboard wave are multiplied by each other sample by sample, instead of added. To quickly modulate by a sine wave, use the Generate Tones function which has a "Modulate by Source" option.

Loop Paste

When checked, the clipboard wave is pasted the number of times entered.

If the format of the waveform data in the clipboard differs from the format it is being pasted into, it will be converted accordingly before pasting occurs.

From Clipboard

When chosen, the audio data to be pasted is the data currently on the clipboard.

From File

When chosen, a file may be chosen with **Select File** to be pasted. This is especially useful when the amount of data you wish to paste is too large for the clipboard. If this is the case, use Save Selection to save out the highlighted selection to a file using a non-compressed file format. Then by using this option the data from the file can be pasted.

3D Echo Chamber

This function will calculate the actual echoes as if the source audio (highlighted selection) and microphones (destination channels for echoed wave) were in a room of any given size and with walls of any given dampening factors. The number of echoes to calculate is adjustable, up to about 5,000 echoes. The more echoes there are to calculate, the longer it will take the function to complete. Practically any "ambiance" setting can be created using this function.

One great use for this function is to convert Mono audio to Stereo with all the right ambiance. Choosing a "left" microphone that is one to two feet away from the "right" microphone will simulate the ears of a listener, and will give the effect of "being there" when listened to with stereo headphones. Be sure to copy the mono audio into a stereo format before performing the echo so you can choose two separate microphone locations. A spatial stereo expansion effect can be created by placing the two microphone locations far apart, further apart in the settings than you will be playing them through speakers in real life. For example, if your stereo speakers are 6 feet apart, try placing the left and right microphones 20 or 30 feet apart in the settings.

To give more control over the environment, dampening factors can be applied to any of the 4 walls, floor, and ceiling. If a wall has a dampening factor of 1.0, it is totally reflective (like cement). If a wall has a very low dampening factor, like 0.05, it will absorb most of the sound (like carpeting or sound proofing panels). You can also lower the dampening factor of some of the walls to simulate the fact that other objects in the room are absorbing some of the audio.

Always place the microphone(s) sufficiently far from the source. If the microphone and source are too close together, you will just hear the source and no echoes since it is analogous to placing your ear right next to the sound source where you hear the sound only (which is very loud) and nothing else.

Room Size

The length, width, and height of the room can be entered in units of feet (sorry, no metrics this time... There are approximately 0.3 meters per foot for those who need to convert). When entering source and microphone locations, they must lie between zero and the room's width for the "Distance from Left" parameter, and zero and length for the "Distance from Back" parameter. Room sizes can be as large as memory will allow.

Intensity

The volume of the echoes is determined by the volume of the first (direct) audio. The direct sound that reaches the microphone from the source will be at the same amplitude as the original audio being echoed. Thus, in a room of any size, if all 6 dampening factors are set to zero, there will be no change when echoing. Every echo adds to the amplitude of the finished audio, so the intensity should be set to less than 100%. In fact, the more echoes there are, the lower this value should be set to prevent clipping. In general, about 1% less for every echo, down to a reasonable limit of about 5% for 100 echoes or more (since after a while, the echoes are so quiet, they don't add to the final amplitude of the audio).

Echoes

This is the number of actual echoes to produce. To get a nice reverb and ambiance effect, at least 30 echoes should be generated. The more echoes that are generated, the truer the result will sound. You must sacrifice the quality you desire with the time you are willing to wait for the final product. Generate about 15 echoes or so to test the chamber size and general room sound, then increase that dramatically for the final production. Up to 4,000 echoes can be generated for the very very patient.

Damping Factors

Use the damping factors to set the type of room in which the audio is being played. The factors can simulate wall coverings, floor coverings, and other objects in the room that absorb sound. Granted, in real life, various objects absorb different frequencies of audio. In this simulation, all frequencies are

reflected equally. The effects of speaker placement enhancing or canceling certain frequencies, though, is still accurate. The fact that cement reflects high frequencies better than low ones is not accounted for, but great effects can still be achieved, and these effects are much more realistic than the basic Echo function. A damping factor of 1.0 is the greatest, simulating total reflectivity. A factor of 0.0 is the lowest, for total sound absorption by the reflecting surface. If enough people want the walls to absorb various frequencies, and not just all frequencies, I can add the parameters if enough people want it.

Source Signal Placement

The source (highlighted audio before running this function) can be placed anywhere in the room. The audio is simulated as a point source of audio, not directional. This means the audio will radiate outwards in all directions from the source, and not more in one direction than another. Directional audio would require a handful of new parameters and if this is desired, I can add it in if enough people say they want it. The distance the source is placed to any of the walls will affect the frequencies that are enhanced. In other words, source signal placement is crucial to the ambiance effect that is gained with this function. With stereo source, each channel can be placed independent of each other.

Microphone Placement

There can be up to two virtual microphones. Each microphone represents a destination audio track. The audio placed back into the waveform (the result of the echoing) is exactly what the microphone would hear if it were in the room at the location specified. Stereo signals have two pick up microphones while mono signals have only one, since there is only one channel in which to place the result. Placing the microphones in a stereo setting one foot apart will simulate the ears, and when listened to with stereo headphones, will sound as if you were actually in the room (if enough echoes are generated). The brain will be able to pick out the directions of each echo, as well as the fact that the delays of the echoes will give the brain cues as to the size of the room. Placing the microphones very far apart and listening with headphones will give a very large "aural" or "Spacy" feeling to the audio, like it is all around you and inside you. Don't place the microphones too close to the source, otherwise the relative volume of the echoes will be so low that they will not be able to be heard.

Mix Left and Right into Single Source

When working with stereo audio, there are actually two source signals, one for each channel that can be placed independently. This takes twice as many calculations as a single audio source, so this option allows you to mix the left and right into a single point source for faster calculations.

Experiment with various settings to get the reverb you desire. Some presets are available, but a big part depends on the type of source audio that is being echoed.

Distortion

Use this function to map any sample value to any new sample value. Effects such as blown car speaker, muffled microphone, overdriven amp, cranked car stereo and many more can be created.

The horizontal axis represents the input sample value in dB, while the vertical axis represents the output sample value in dB.

Have fun making your audio sound really really BAD! (Of course, it's great for adding fuzz to guitar licks to get that heavy metal sound).

Normalize

Normalize will amplify the highlighted selection so it is the desired percentage of maximum. Optionally, a DC Bias can be set. Setting a DC Bias to zero will ensure that the waveform is centered on the zero voltage line.

Use this normalize function if you are recording a script in which you want to normalize a waveform to a specific percentage of maximum. After normalizing to a specified level, pressing the F3 key will automatically run Normalize again on another waveform for very fast normalization of waves.

Convert Sample Type

This function converts the sample type "in place" by directly converting the types of the samples in the temporary file that represents the current waveform. Various levels of quality can be chosen, as well as general volume levels when converting between mono and stereo formats. The **High** quality settings are slower, but the quality far surpasses that of the quicker **Low** quality setting in sample rate conversion.

When downsampling (converting to a lower sample rate), the **High Quality** setting performs a brick wall low pass filter at just below the Nyquist (1/2 the lower sample rate) frequency. Optional **emphasis** can be performed when downsampling to try and regain some of the high frequency energy that was lost because of the brick wall filter. In fact, an emphasis of about 3dB per sample-rate-halve or sample-rate-double is recommended due to the loss of high frequency incurred by the compression algorithm. For example, when downsampling 44.1 KHz to 22 KHz, use a 3dB emphasis minimum. If downsampling 48 KHz to 11 KHz, use a 6dB emphasis minimum. The frequencies just below the Nyquist are amplified the number of decibels entered, and quickly, but very smoothly, taper off to no amplification by the lowest frequencies. Choose higher levels of emphasis than the recommended minimum when downsampling to brighten the highs, which should help compensate for the loss in high frequency due to the lower sample rate. Using high quality will very cleanly convert between non-standard sample rates as well (e.g. 44.1 KHz to 16 KHz, or 32 KHz to 11 KHz) with minimal aliasing effects in the very high frequencies.

The **Low Quality** setting just does a straight linear interpolation of data points which introduces some high frequency noise when upsampling, or aliasing when downsampling.

When converting from Mono to Stereo, you can choose the amplification levels for both channels independently, with 100% for both channels being the default. You can choose a value of -100% for one of the channels to get an "inverse mono" effect, where the left channel is the inverse of the right. When converting from Stereo to Mono, you can choose the amplification values for each channel before they are combined. Values of 50% for both channels is the default, meaning that the resultant mono waveform has $(l+r)/2$ signal, or the average of the two channels. You can even choose a negative value for one of the channels to perform a vocal cut effect on some audio.

When converting from 16-bit resolution to 8-bit, you can optionally add some dither in the audio to make sounds still audible that are quieter than the limit that 8-bit audio provides. To do this, a small amount of noise is added to the signal, but quieter audio can be heard in the noise. If dither is not checked, quiet audio will just fade in and out, with a more disruptive choppiness sound that resembles rain falling, or static. Whether or not dithering is used depends on the audio being converted, and your preferences.

Noise Reduction

Background noise and general broad band noise can be dramatically reduced with minimal reduction in signal quality. The amount of noise reduction depends upon the type of background noise, and the allowable loss in the quality of the signal that is to be kept. In general, increases in Signal to Noise ratios of 5dB to 20dB can be achieved (noise is reduced 21dB and signal 1dB for example).

Two steps are required to remove noise. First, the noise level must be set so the filter knows what type of "noise" to remove. To do this, highlight a section of the waveform that has no important signal in it, and only has background noise, then press **Set Noise Level**. The statistical information about the background noise is then gathered, and you are set to remove all noise of this type from your waveform.

Once the statistical data is gathered, the second step is to highlight the section you want to remove the noise from, and choose the level of reduction you desire. A level of zero will remove the least amount of noise, and nearly no signal loss will occur. Typically the noise will be reduced about 3dB at this level. A level of 100 will remove the maximum amount of noise, lowering the noise level by about 20dB. If the signal you are trying to keep gets too distorted at this level, use lower values until a balance is reached between noise reduction and allowable signal distortion. Values any higher than 100 will guarantee loss of the signal that you want to retain, but this may be desired if reducing noise is more important than retaining original signal.

Distortion effects may manifest themselves a "hollow" or "underwater/burbly" sounding signal, dull sounding impacts, "rolly" high end, or a "computerish" mechanical sound. These effects, if heard at all, will fall off if the noise reduction level is reduced. The amount and type distortion depends on the type of noise that is being filtered.

Besides reducing the noise level, the type of noise that is present after reduction is entirely different than the type of noise beforehand. For example, if you are trying to get rid some "tape hiss" from a waveform, the tape hiss sound will completely disappear, and in it's place about 15dB quieter will be completely different type of noise. This noise will contain all frequencies in different combinations, thus it cannot be reduced much further without noticeable signal loss. The new noise has a "burbly" or "bubbly" quality to it, and if amplified, sounds very harmonic--like those 1960's computers in old science fiction films. Since this is so much quieter than the original noise though, it is very acceptable.

Great effects can be generated by setting the noise level to some valid signal component in the waveform, and not the background noise. Whatever frequencies are present in the highlighted selection when **Set Noise Level** is chosen will be removed when the reduction level is set to 100.

Use this function to remove tape hiss, microphone background noise, 60 cycle hum, or any noise that is constant throughout the duration of your waveform. You can even reduce the noise incurred by the sound board's circuitry during recording--just record a second of silence before whatever you want to record and tell the noise reducer to remove the sound of that silence for another 10dB dynamic range.

Save Profile

Once the noise level is set, you can save the noise profile in a *.fft file. This file will contain information on sample type, FFT size, and three sets of FFT coefficients, one for the lowest amount of noise found, one for the highest amount, and one for the power average.

Load Profile

Loads any previously saved noise profile. You can load any *.fft file that Cool Edit has saved. Eventually the spectrum analyzer and Filter functions will support *.fft files. A noise profile is only compatible if it is being used on a sample of the same type when the profile was saved. In other words, a 44K stereo 8-bit sample is not compatible with a 22 KHz mono 16-bit profile. Also, since noise profiles are so specific to the recording environment of waveform in question, even if the sample types are compatible, a profile for one type of noise will not work on another type. Even if the audio samples were

recorded with the same microphone, if the recording environment is different, the type of background noise could be different.

FFT and STATISTICAL SAMPLING settings

These settings will affect the noise reduction quality, and the type of distortion heard when reducing the noise. Try different settings to get the best noise reduction while keeping the intended signal in tact. The **FFT Size** parameter causes the most drastic changes in quality. Good settings for the size range from 4096 to 12000. The **FFT Precision Factor** affects distortions in amplitude. With values of 3 or less, the FFT is performed in giant blocks that are not very continuous between the blocks. This means after each block is processed, there can be a drop or spike in volume at the interval between blocks. Values of 5 and up work best. On past about 10, there is no noticeable change in quality - just the time it takes to compute. I like to use 5 or 7 (and I think odd numbers are best for symmetric properties). The **Number of Samples** can be chosen for the gathering of statistical data. This number describes how many snapshots of noise to take in the highlighted interval when Set Noise Level is pressed. The larger this number, the more accurate the statistical data is. A value of 64 is plenty. You will notice that using very small numbers of statistical samples will greatly affect the quality of the various noise reduction levels. With more samples, a noise reduction level of 100 will most likely cut out more noise, but also cut out more original signal too. But, with more samples, a low noise reduction level will also cut out more noise, but most likely *not* disrupt the intended signal. If the selection used for learning the noise level is too small, then the Set Noise Level button will not activate. It is possible to make a larger section of noise by using Copy and Paste for reasonable results with very short noise samplings.

Special Notes

Noise reduction works best on 16-bit samples, although it will work perfectly on 8-bit samples. Because of the nature of 8-bit audio, it is impossible to get the noise level to less than about -45dB if even that. Noise at -45dB is very audible, as owners of 8-bit sound cards can attest. Converting to 16-bit first, then reducing the noise will produce a sample with much less noise than can be done in 8-bit alone.

The noise reduction works best if the original signal is centered. To center a signal, highlight it and choose "Center Wave" from the Amplify function. Centering the wave adjusts the DC offset to zero. If the wave is not centered, audible clicking may be heard in really quiet situations. Since centering takes out all frequencies below about 16Hz, it is completely safe to do without any ill side effects.

About Carrier Waves

A carrier wave is needed to transport the brainwave frequencies. Because the carrier wave is not what you hear through the headphones directly, you do not need to buy super high-end headphones (5Hz-25KHz) to reproduce the effects. In other words, your headphones do not need to be able to reproduce a 5Hz signal if you are generating a 5Hz theta-frequency brainwave file. The brain *does* however respond better to the lower frequencies, so the better the headphones you buy, the more dramatic the results will be. The best headphones are the kind that cover the entire ear, so outside noise does not get in. Plus, these headphones have much higher response to low frequencies.

Carrier waves must have some correlation between the left and right channels, no matter how slight. So mono (total correlation), inverse (total negative correlation), and spatial (natural recordings that have some of the same sounds coming in both channels) will work great.

The best sounds to use as carriers are sounds that are spread across the entire frequency range, or at least most of the lower frequency range. Good examples are ocean, waterfall (most any recordings from nature), and noise generated by this program. Experiment with mono (both left and right channels the same), inverted (like mono, but the left channel is the inverse of the right, obtained by using the Channel Mixer), and spatial stereo (spatially encoded sounds in nature, recorded with microphones about 9 inches apart to simulate separation between the ears). But don't let this stop you from digitizing your favorite music, and using it as a carrier, or converting your favorite to a mono or inverted wave.

To generate a carrier wave, you can do three things:

Record a sample Once recorded, use the Channel Mixer to create a mono, or inversed wave. Or just leave it the way it was recorded. You may find changes in effectiveness of the brainwave files depending on how you use the Channel Mixer. Keep in mind that this function only operates on stereo waves, so when "mono" is mentioned, it means that the exact same signal is present on both channels--the left channel and right channel are the same.

Generate Tones You may use the Generate Tones function to find a pleasing, relaxing tone for the background (but I find "noise" sounds more relaxing). The way tones work the best is if the left channel's tone frequency is 5-6 Hz different from the right channel's tone. This creates a beat pattern equal to the frequency difference, which the brain responds to somewhat (this is the property that many theta-inducers rely on). To do this, generate one tone with left volume at 40, and right volume at zero. Then generate the second tone with the left and right volumes reversed. Finally, Paste Special (with overlap) one tone on top of the other. Use low frequency tones, like 50Hz to 120Hz for best results. These tones, by themselves, will help coerce the mind into the state associated with the difference between the frequencies. For example, for a theta state of 6Hz, use a 70Hz and a 76Hz tone. Combining this tones sample with an existing brainwave file, by overlap pasting at a quiet volume (20%) is even more effective.

Generate Noise Use the Generate Noise function (pink and brown work best) in any of the modes: mono, inverse, or spatial stereo (independent channels noise will **not** work as a carrier for brainwave frequencies at all, since there is no correlation between the left and right channels). I find that using pink noise in spatial stereo, and running it through the Quick Filter to get rid off some of the "edge" if any works the best. I have also found Inverse to work quite well too, but the brainwave "effect" is more pronounced, and can be distracting, and some sound boards have trouble reproducing sound that is inversed between channels.

Once you have found a pleasing sound, about 10 seconds or so of a monotonous sound (tones, river, waterfall, noise...) you're ready to start. If a monotonous sound is used, more disk space can be saved because we will use the play list to repeat portions. If a music sample were used, it is quite noticeable that the same 10-second piece is being played over and over and over again.

If you're curious you can also [spatially locate a mono sound to the left or right?](#) Do this if you wish to

have the illusion that a particular sound is coming from one side or the other. The function works by pasting a mono sound sample into a stereo waveform, and using the Digital Delay function. Having a quiet "ping" (generated by using the sine wave tone generator with the bell curve envelope) play spatially on the left, then on the right at about 5 second intervals is very relaxing.

Encoding Brainwave Information

There are two types of brainwave files that you can create: A **flat file**, and a **cued file**. The flat file takes more memory, and plays straight through from beginning to end, while the cued file is actually contains pieces of the entire audio program, that when played in the proper order become the brainwave file. The cued file takes less memory, and can very quickly be modified at any time by re-arranging the audio pieces. The average length of a cued file is about 3-4 minutes for a program that can last as long as desired. The flat file is a standard wave file, which means to create a long program, you must have enough space for it. The only advantage to using a flat file is if you are waving music, since music cannot be split into pieces and re-arranged, otherwise it would sound discontinuous. Creating brainwave files using the flat file method will be discussed first, since it is more straightforward

Flat Brainwave File Generation Create a file the length you wish to make your relaxation program using the carrier wave(s) of your choice. Either record music, or use the pink noise generator and copy and paste (or Paste Special) to the desired length. If you are using a monotonous sound, you would be better off using the cued file method. Lengths of good relaxation programs vary from 15 to 30 minutes, and beyond. This means you must have enough hard drive space for the entire file. Since the temporary file takes up hard drive space as well, the maximum size of file you can create, and be able to save, will be one that takes up half of the initial free hard drive space.

Use the Wave function to encode the brainwave patterns into the carrier wave by highlighting a section of the wave, or the whole thing, and choosing Transform:Wave, or click the wave icon. With the wave transformation, you have complete control over the brainwave frequency being encoded, the strength of the signal, and the positioning of the signal left or right. Over the selection highlighted, the intensity, and position remain constant, but the frequency can be varied using the graphical input control. See the section on Authoring Brainwave Files to learn what settings to use for the Wave function, and how to build effective files.

Once the entire file has been waved to your satisfaction, you can save the file if you wish, and play it using the Play button. An interesting side effect is that different sounds are heard if you listen to one channel, listen to both channels with one ear, or listen to each channel with each ear.

Cued Brainwave File Generation These files contain many short snippets of brainwave encodings at different frequencies. Each snippet is cued using the Cue List, and a Play List is generated by adding entries from the Cue List, and looping them if necessary. To listen to a cued brainwave file, you must use the Play button in the Play List dialog box.

First you must figure out how you want to divide up the brainwave program (your 20-30 minute masterpiece) into components. For example, you may want to have patterns of 5Hz, 7Hz, and 9Hz at different points in the program. In this case, you will need at least three pieces for your creation. The actual file will just be 10 seconds of carrier wave at 5Hz, followed by 10 seconds at 7hz, followed by 10 seconds at 9Hz. All the pieces are placed in the cue list by highlighting the piece, and choosing **Add**. It is best to add the piece to the cue list once it is created, or pasted at the end of the current waveform. To create the final program, the pieces are added to the Play List in the order you wish to listen to them. Each piece can be looped if needed. So a 20 minute program can be generated from 3 10-second pieces by adding the cues to the play list and looping.

First you need to create 10 to 20 seconds of carrier wave, and save in a special file in case you need the carrier wave again later. Highlight the wave, and Edit:Copy. When you need another copy of the initial carrier wave, you need only to Paste it.

Add the first carrier wave snippet to the Cue List by pressing the **Add** button in the Cue List dialog. Give the cue for this snippet a name that reflects the waveform transformation you will be using, for example, "6Hz to 5Hz drop".

Choose the **Transform:Wave** function to encode the proper patterns into the carrier wave. Look at the section on Authoring Brainwave Files to learn what settings to choose.

Click past the end of the wave file (make sure the rightmost part of the file is in view), and choose Paste to insert another copy of the carrier wave. Once you do this, you can add the newly inserted selection to the cue list, and give it a name. Repeat the step above for creating a brainwave encoding over the carrier wave you just inserted. Do this as many times as needed until you have all the pieces you need to build the final brainwave file.

Once all the pieces have been generated, add them in the order you like to the play list. To make pieces last longer (if the beginning and ending of the piece are at the same brainwave frequency), increase the number of loops for that entry in the play list.

When Played from the play list, the pieces will be played in the order shown, and looped if necessary.

To get familiar with the cue list, and play list, open one of your favorite wave files, and highlight sections then add them to the cue list. After you have a few selections in the cue list, add them to the play list, and choose a loop count of greater than one for some of them. Choose Play from the play list, and listen to what you've just created.

Authoring Brainwave Files

After learning about carrier waves, and encoding procedures, all you need to know is what frequencies to use, and when to use them during the course of the listening session. Once you know what frequencies to use, and at what intensity, you can generate the completed file using either of the methods above.

Effective brainwave files have some sort of encoding going on the entire length of the session. For the first 3 minutes or so of the session, the listener will not be in a "relaxed" state, and will not respond greatly to the frequencies being presented. During this *warm-up* period, gradually decreasing from about 12Hz down to 8Hz works nicely. After about 4 minutes, the listener's brainwave patterns will start to synchronize with the patterns in the headphones, and the serious brainwave programming can begin.

Frequencies of 8-10Hz correspond to an alpha state -- light relaxation, like a quick afternoon siesta. Frequencies of 6-7Hz correspond to a theta state -- meditation. 4-5Hz correspond to deep relaxation. You can create a session that is constant, in one of these states, or create a session that dynamically flows from one to the other. When going down in frequency, give the listener about one minute to "catch up", and stay in sync with the wave. Going up in frequency does not require the listener to catch up. In other words, if you go from 6Hz down to 4Hz over a 20 second time span, and hold at 4Hz, the listener may not be at 4Hz for another minute. When going from 4Hz to 8Hz in 20 seconds, the listener will be at 8Hz at the end of the 20 seconds. It appears to take extra time when going down in frequency, but no extra time when going up. This basically holds true for the first 20-30 minutes of a session. After that, the opposite tends to occur. It is easier to go lower than go higher. This means that to bring a listener from 4Hz (where she has been for the last 30 minutes) up to 12 Hz, it should be done over a 5 minute period or so. One nice "trick" to do is to keep the listener at around 4-5 Hz for a while, then about once every 2 minutes, go up to 8Hz and back over a 20 second span. This will *alert* the listener slightly, and make them aware for a few seconds of what they are thinking. This is great for getting creative insights and the like. It acts as a sort of *window* to the subconscious, allowing one to remember what is going on. It's kind of like remembering dreams: you do it better if you are awoken in the middle of one.

Another effective method of producing relaxation files is to overlap them. That is, have portions that are one frequency, and slightly spatially located to one side overlapped with a slightly differing frequency spatially located slightly to the other side. This gives the listener the chance to *decide* which frequency to be at, and gives them more freedom over the experience. For example, a session could go from 8Hz to 4Hz over 10 minutes overlapped with 7Hz to 5Hz over the same 10 minutes.

For nice *super-relaxing* effects, generate panning waves (frequencies of 0.05 to 0.2) over your session after encoding the initial brainwave patterns. For example, if you are generating a brainwave file out of 20-second pieces, after generating the main brainwave frequency over the 20 second period, generate a panning wave of 0.05 or 0.1 (which means a period of 20 or 10 seconds) with an intensity of about 50 or so. This will make the sound appear to shift left and right to the listener over a 20 or 10 second period. Now, overlapping a 24-second piece panned at 0.125 (8 second period) at 5Hz with a 0.167 (6 second period) at 6Hz will combine the practices of multiple frequencies with panning for an extremely super-natural effect!

Once you get started creating a few files, and see what the different frequency ranges do, you will become familiar with the different effects and how to generate just the effects you want.

Gamma frequencies of 200Hz or more seem help in relaxation, and do something I'm sure, but I don't know what... yet. This is an area you can experiment with. When generating frequencies above 40Hz or so, it is best to keep the intensity very low, like 7 or 8. The higher the frequency, the lower the intensity has to be, otherwise the encoding will overwrite itself and the signal will be lost.

Sample Theta File - Step-by-Step

- 1 Create a new blank file with **File:New**. Choose a **Stereo** file, either 8 or 16 bit and a 11025, 22050, or 44100 sampling rate. The final file size will be one of the following sizes listed below depending on your choice:

		11025	22050	44100
8-bit	2.6M	5.2M	10.5M	
16-bit	5.2M	10.5M	21.2M	

You must make sure you have enough memory for a file of this size, plus an additional meg for working space. If you plan on saving the file when you are done, you must have at least **twice** this amount of hard drive space available, since a temporary file is used instead of memory while working on the wave.

- 2 Choose **Generate:Noise**. Choose **Pink Spatial Stereo** for **15** seconds at an intensity of **3**. This is usually the longest portion of the generation of brainwave files. Because of this, it is advised that you save this piece of *noise* so that in generating future files, you can just load in this pre-calculated noise as a starting point.
- 3 Choose **Edit:Copy**. From now on, we will paste the noise in when we need it!
- 4 Make sure the noise is highlighted. If it is not, select all by double-clicking on the waveform until it is highlighted.
- 5 Choose **Add** in the Cue list, and give the entry a **Label** of **10Hz to 8Hz**, and a **Description** of **Warm-Up**.
- 6 Choose **Transform:Wave** to bring up the brainwave dialog box. Enter **10** for the **Highest Frequency**, and **8** for the **Lowest Frequency**, and an **Intensity** of **35**. On the graph above, click the leftmost dot, and drag it to the top of the graph. Click the rightmost dot, and drag it to the bottom of the graph. This will product a frequency encoded at 10Hz at the beginning, and glide down to 8Hz by the end. Choose **OK** to generate the encoding. This shouldn't take nearly as long as it did to generate the noise.
- 7 Click the mouse at the rightmost portion of the wave (just beyond the *black* waveform display area). When you do this, the yellow cursor arrows should be all the way to the right of the wave. You must always add new pattern blocks at the **end** of the current waveform.
- 8 Choose **Edit:Paste** to insert another copy of the original noise that we had copied originally.
- 9 Create the following pattern blocks as before (following the steps 5 to 8) , except with the following values for the cue list and waveform transformation:

Label	Description	Hi Freq.	Lo Freq.	Intensity	
8 Hz	Alpha	8	8		37
8 to 6Hz	Glide Down	8	6	38	
6Hz	High Theta	6	6	40	
6 to 5Hz	Deeper Theta	6	5	45	
5Hz	Theta	5	5	50	
5-8-5	Spike	8	5	50	
		<i>(Graph should look like an upside-down "V")</i>			
5 to 12Hz	Awake	12	5	40	
		<i>(Graph should go from left=lowest to right=highest)</i>			

10 Once all the blocks are generated, and in the cue list, Add the pieces to the play list by selecting the wave portion in the cue list and clicking **Add** in the play list. Select the pieces listed below in the order given. After doing so, select each item in the play list, and change the **Loops** for each so the final play list looks like this:

- (1) 10 to 8Hz
- (3) 8Hz
- (1) 8 to 6Hz
- (7) 6Hz
- (1) 6 to 5Hz
- (18) 5Hz
- (1) 5-8-5
- (12) 5Hz
- (1) 5-8-5
- (12) 5Hz
- (1) 5-8-5
- (12) 5Hz
- (1) 5-8-5
- (12) 5Hz
- (1) 5 to 12Hz

When you choose **-Play-** from the play list, the sequence will be played in the order given, looping the number of times specified. This list gives a 21 minute theta session, with bursts into alpha at four points.

11 If you wish to save this piece, and have enough hard drive space, you can do it now. The wave is complete. Enjoy.

About Brainwave Files

The wave option works like many meditation tapes and light/sound devices on the market, which range in price from \$200 to \$500. There's even a board available with plug in glasses (which have blinking lights) for your PC for \$495. I think the files created using the 'Wave' transformation are even more powerful, and are definitely more pleasing to the ears. Most other devices and tapes have a "humming" sound or some other tones to induce the right brainwave frequencies. This program allows you to use ANY sound to encode the frequencies with. The most effective I have found are by using the [noise generator](#), which creates pleasing waterfall like sounds. This function only works on **stereo** waveforms, and the effects work if only if listened to with **stereo headphones**.

There are four major (perhaps more) [Brainwave Patterns](#) ranging from deep sleep to acute awareness. Listening to sounds that have been waved for periods of 5 minutes or more will produce the desired state of awareness in the listener. Sessions of 25 minutes or so work really well!

Many Uses

About Carrier Waves

Encoding Brainwave Information

Authoring Brainwave Files

Step-By-Step Theta File

How to Use Brainwave Synchronization Files

Articles on Theta meditation

References

DISCLAIMER

By using this program, you agree that the author of the program will not be responsible for any damage as a result, direct or indirect, of using this program.

WARNING

Sounds generated by the wave function are not suitable for epileptics or persons undergoing psychiatric treatment since it *does* alter ones state of consciousness.

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Many Uses / Effects of Using

Immediate Relaxation and Stress Relief

Choose between 5hz and 10 hz for different levels of relaxation.

Meditation

Choose between 4hz and 7hz, either cycle between a few, or stay at a particular frequency for different results.

Improved Sleeping Patterns

Any of the Alpha and Theta frequencies (8Hz to 4Hz) for 30-45 minute sessions.

Treatment of Insomnia

Choose between 4hz and 6hz for starters (1/2 hour), then go into frequencies below 3.5hz (1/2 hour), settling on about 1.5hz to 2.5hz for sleep.

Improved and Lasting Sense of Well Being

Try Theta (4Hz to 7Hz) for 45 minutes, daily.

Creative Visualization

About 6hz for a while, then up to 10hz works well.

Alleviation of Migraines and Headaches

Experiment with Alpha and Theta combinations.

Reduction of Depression Symptoms

Again, Alpha and Theta combinations, mostly theta(?)

Self Hypnosis

Choose about 8hz to 10hz while playing any self-hypnosis tape, or guided meditation.

Super Learning

Choose about 7hz to 9hz while playing any learning tapes, like foreign language tapes, etc. to increase comprehension.

Subliminal Programming

Choose 5hz to 7hz while playing your favorite subliminal tapes, or make your own by recording some affirmations, and mix paste them from the clipboard at barely audible volumes.

Improve ESP / Intuition

Theta frequencies help in this area, 4hz to 7hz.

Reaching Higher States of Consciousness

Theta again, with daily half hour sessions.

Quick Refresher on long days

Low Alpha 8hz to 10hz for about 15 minutes works well.

Increased Immune System

Relaxing to Alpha and Theta combinations daily.

Using Alpha Synchronization (8Hz to 12Hz), expect similar increases in neuro-chemical levels:

11% increase in Norepinephrin levels

21% increase in Serotonin levels

25% increase in Beta-Endorphin levels

How To Use

Once you have created your brainwave file (15 minute files on up work best), loop play them for a longer listening time. Sessions of 15 minutes or more work best. It is best to listen to the sessions lying down in a quiet place where you will not be disturbed. If there is no place like this near your PC, it may be a good idea to record the session on tape and listen to it where you can be comfortable and relaxed. When you're fully comfortable, start the session, close your eyes, and let the magical sounds from Cool Edit do the work. Remember, this only works if you listen to the sounds with stereo headphones.

You may notice helicopter, or "washing" type noises moving around in your head. These sounds are actually created inside your head, and are not coming directly out of either channel from the sound board. It is this noise that is doing the work of helping your brainwaves get synchronized to the patterns you have chosen. When I have mixed two different (but similar in frequency ranges) brainwave files together, I have noticed a jet airplane noise moving slowly from left to right in the background. Some people don't hear these artifacts at all, while others hear them extremely well.

Another side effect is that of a wandering mind. When I use frequencies under 8hz, I find myself thinking of the strangest things. You may find that you are not thinking of anything in particular, and your thoughts become very interesting. The feeling is also "warm" and "happy" for some people. Others start recalling their favorite memories as a child, even some they thought they had forgotten forever!

After a session of 15 minutes or more, you may feel quite refreshed, light, airy, clear-headed, etc. I always find myself feeling very good afterwards. Some claim that doing this for 30 minutes a day can result in subtle but great changes in your life. ESP experiences increase, and you may be able to reach new levels of awareness in your everyday life.

Unsolicited Testimonials

Excerpts from Articles on Theta Meditation

Any results gained from using Light/Sound devices such as the Voyager, the Brain Machine, MasterMind, the MindsEye, etc. can be gained from listening to these files as well. Some of these excerpts are from results using these machines with flickering light goggles and tones.

This is my own testimonial. I find that within minutes of listening to a Theta pattern, my thoughts start drifting to really weird stuff that doesn't make any sense at all. It is really quite funny some of the things I find myself thinking of. Then I start to feel really "good" and relaxed--at peace if you will.

It has been stated in several places that a half hour a day of the brain being in the theta state can replace up to 4 hours of sleep. People find they need less sleep at night.

Thirty patients had sessions in Theta (5 hz) and experienced relaxation states of 80-100% after five minutes as well as improved pain relief. Eight patients had blood tests before and after the sessions and showed improved beta-endorphin levels of 10-50%.

Using a first-generation prototype light/sound device, one doctor noted, "these devices produce a distinct relaxation state. Programming the device between 3 and 7 hz, it takes about 10 to 15 minutes for the patients to enter--effortlessly--a state of hypnosis. They terminate the sessions relaxed and with a feeling of well-being." Also, "the device has a calming effect on the nervous or anxious patients. In a majority of cases, the patients feel relaxed, and calm during a period of three to four days after the session. It happens that the subjects have a reminiscence of childhood experiences, particularly when in Theta. They related their experiences which we incorporated into our psychotherapeutic program."

"The machine works like a tranquilizer and the effect lasts for several days. Using the machines in Theta frequency, clients are very receptive to suggestions on behavioral aspects such as reducing tobacco, alcohol and food consumptions." Many patients "were more creative during the sessions."

"By inducing hemispheric coherence the machine can contribute to improved intellectual functioning of the brain. Like children spending most of their time in Theta, the machine allows a reduction in learning time. With adults a return into Theta allows them to rediscover childhood experiences. The machine is like a 'lost and found office' for the subconscious."

D.J. Anderson, B.Sc., M.B., "The treatment of Migraine with Variable Frequency Photo-Stimulation," in HEADACHE, March 1989, pp 154-155:

D.J. Anderson used photo-stimulating goggles with variable frequency using red LEDs in order to stimulate the optic nerve, through closed eyes, right and left with frequencies between 0.5 and 50 hz. The study included seven patients who suffered a total of more than 50 migraines during the observation period. Forty-nine of these migraines were relieved (either by reducing the average duration or by increasing the frequency interval in between migraine crisis) and 36 other migraines could be stopped while using the goggles.

The more these sounds are used, the easier it becomes to produce and maintain Alpha/Theta rhythms. As these states of higher awareness become infused into normal brain activity, the result can lead to what some have called a fifth state of consciousness, or an "awakened mind." In this state of illumination and bliss one sees the world as distinctly as before but with a new mind that perceives the universe with new meaning. It's this experience of illumination that is the seed for all breakthrough scientific theories, literary ideas, revolutionary inventions, and artistic masterpieces. The technology used here induces these states by forcing your brain to focus your mental energies inward... tapping your own vast reserve of creative genius and eventually unfolding "an enlightened state of awareness."

An unusual side benefit of listening to these sounds is a surprising need for less sleep. Some users are able to reduce their sleep requirement by as much as 3 - 4 hours each night, rising each morning feeling

refreshed as if they had slept a full 8 hours. The reason? It's believed the theta-sounds replace the need for extensive dreaming which is the main purpose of sleep. Another interesting side effect, many users report a dramatic increase in sex drive. No one knows exactly why, but it may be linked to changes in brain chemistry. But, perhaps the most unusual side effect is the reported increase in psychic functioning, including episodes of precognition, out-of-body experiences, and spontaneous channeling events.

When you finish each session your entire body becomes charged with a new energy and vitality. Fears and anxieties are gone. You are renewed, more alert, and mentally you feel on top of the world.

What causes the euphoria and peak experiences? The neuroscientists say the 'high' you experience is caused by a release of endorphins in the brain. A hundred times more powerful than morphine it makes you feel like you're soaring with eagles.

Zen meditators have been found to alter Alpha/Theta frequency according to their depth of meditation, reports Japan's leading neurophysiologist, Dr. Tomio Hirai. He has correlated brain-wave patterns with certain stages of meditation. And according to Dr. Hirai, "Meditation is not merely a state between mental stability and sleep, but a condition in which the mind operates at the optimum. In this condition the person is relaxed but ready to accept and respond positively to any stimulus that may reach him."

Research now confirms that brainwave rhythms correspond to certain states of consciousness, and this suggests that individuals capable of altering their brainwave patterns can have significant control over other mental and physiological functioning. As Elmer and Alyce Green of the famous Menninger Institute first reported in the mid-70's, "...simply causing your brain to generate theta activity for a few minutes each day seems to have enormous benefits, including boosting the immune system, enhancing creativity, and triggering integrative experiences leading to feelings of psychological well-being."

Biofeedback researchers have found that people who enter the "theta state," expand their states of consciousness, acquire super-receptivity to new information, and demonstrate a greater ability to "rescript" material on a subconscious level. Even more astonishing are the findings of a study conducted on a group of chronic alcoholics at a University in Colorado. After 13 weeks the group that learned to generate theta and alpha brainwaves, showed a far greater recover rate, and a complete transformation of personality. There is a remarkable body of evidence...

Another Note: I did not do a lot of proper citing, I just wanted to present some information that is available "out there".

The major brainwave patterns are:

Beta > 13 hz Normal state of alertness, stress and anxiety.

Alpha 8-12 hz Light relaxation, "superlearning", positive thinking.

Theta 4-7 hz Deep relaxation, meditation, increased memory and focus.

Delta 1-3 hz Deep sleep, lucid dreaming, increased immune functions.

Gamma 200+ hz Not sure exactly what these do...

Selecting **Loop/Play Toggle** from the **Options** menu will toggle the play button between the **Loop** and **Play** states.

Level Meters

Selecting **Level Meters** from the **Options** menu will start monitoring the recording source, which is useful for setting the recording levels before recording. To stop monitoring, press the **Stop** console button. The meter may also be started and stopped by double-clicking on the meter display area. If your audio system software crashes when doing this, try increasing the STACKS in CONFIG.SYS to STACKS=12,512.

The levels displayed represent the peak amplitude in decibels, where a level of 0dB is the absolute maximum before clipping occurs. If clipping does occur, the clip indicator to the right of the meter will light up. Just click on the clipping indicator to clear it at any time. When displaying stereo audio, the top meter represents the left channel, and the bottom the right. Yellow peak indicators will *stick* for 1-1/2 seconds to easily read the peak amplitude.

Clicking the Right Mouse Button on the level meter display will bring up a configuration menu. From the configuration menu, you may do the following:

Start/Stop Meter

Start or Stop monitoring of the input source. When monitoring is active, the meters will respond directly to the audio input.

Clear Clip Indicators

The box(es) at the right will light up red if audio is clipping. Click on the box or choose this option to reset the indicator. It is best to record audio as loud as possible without clipping. It is sometimes easier to keep the loudest point somewhere between -2dB and 0dB when setting the recording levels on your sound board's mixer.

Adjust for DC

Many sound boards record audio with a slight DC offset, which means that the center of the waveform being recorded is not at the exact center of the waveform display, but a little above or below it. This can dramatically throw the level meters off since the amount the waveform is displaced could be interpreted as a constant sound that loud. To compensate, make sure this menu item is checked. The recording meters will dynamically adjust to the DC offset, and display the true amplitude of the signal in decibels.

Show Valleys

Just as the yellow indicators show peak levels, if Show Valleys is chosen, valley levels (minimum amplitudes) will be marked as well. This gives a good indication of the dynamic range of the audio. If the valley indicators are close to the peak indicators, the dynamic range is low. If they are spread far apart, the dynamic range is high (the difference between the quietest sounds and loudest sounds is greater).

90dB Range to 30dB Range

This is the range that the meter covers. When recording 8-bit audio, there is no need for anything greater than a 45dB range, since 8-bit audio can not really record anything below a volume level of -45dB. Use a lower range to see the loud portions more clearly. Use a higher range to see the quieter portions for very high dynamic range audio. Note: You may find that when you think your sound board is recording pure silence, you will see the meters fluctuating between points around -87dB up to -60dB instead of going all the way down. This is because of noise in the sound board. Some sound boards have higher signal-to-noise ratios than others. Generally, the higher quality the sound board, the lower the meters will go down during *pure* silence. To quickly see how noisy your own sound card is, choose File->New and create a new 44.1KHz 16-bit file. Then start the level meters, and choose the 90dB Range. This test only works for 16-bit sound cards, since 8-bit sound cards have a maximum dynamic range of around 45dB.

Dynamic / Static Peaks

Choosing Dynamic Peaks will cause the yellow peak level indicators to reset to a new peak level after 1-1/2 seconds. In Static mode, the peaks never reset. Use Dynamic mode to easily see visually the peak amplitude *right now*. As audio gets quieter, the peak indicators will start backing off as well. Use Static mode to retain the maximum amplitude of the signal since monitoring, playing, or recording began. The peak can still be reset manually at any time by clearing the clip indicators (clicking on the clip indicator at the right). Static mode is great for finding out how loud a song will get before recording it. Just start the meters and start playing the song. When the song is over, the peak indicators will show how loud the loudest part of the song was.

Use the mouse to click on the starting point of the wave you wish to select, then drag the mouse to the ending point. When doing this, the portion that is being selected will be inverted (you'll see a white bar indicating the selected portion). Use the Shift button in combination with the mouse to extend a selection.

Double-clicking in the viewing field will select the entire viewing field.

After selecting a range, **Zoom In** will expand the selection to full viewing area

If zoomed in, zooming out will display more of the wave in the viewing area to the left and to the right.

Clicking on **Full View** will display the entire waveform in the viewing field.

The green bar above the viewing field will shrink or grow depending on whether the view is zoomed in or not. The location of the green bar depicts the relative location in the waveform the viewing area is displaying. Clicking on the green bar and dragging left or right will scroll the wave being viewed left or right. Clicking in the black area on the left or right of the green bar will scroll the bar left or right exactly one page. Double-clicking will bring up the samples data entry box.

Beginning and Ending samples are displayed in the boxes at the right side of the window. Beneath is a time display showing the amount of time selected, or currently being viewed. Double clicking on any of these displays will toggle the mode between Samples and Time.

Clicking the **Play** button will play the selected range, or the entire portion of the wave in the viewing field if no range is selected. If the button reads **Loop**, then the selection will be repeated until the **Stop** button is pressed.

Clicking the **Stop** button will stop the currently playing wave from playing.

If your system supports recording, pressing this will start recording. Press **Stop** when done recording. Recording starts at the insertion point and overwrites any data thereafter if any.

The point of insertion for **Paste** and **Paste Special** is indicated by the yellow arrows above and below the viewing area. The exact sample is listed in the **Beg:** box.

Technical Notes

Note 1:

When a file is opened using COOL, a temporary file (~COL?????.TMP) in your temporary directory is created. This is where the actual wave data is going to be played from, recorded to, or edited. This means that the largest file that COOL will work with is limited to the amount of free space on the hard drive containing the temporary directory (pointed to by the TEMP environment variable). If you wish to save the file, you must have enough free space on the drive you are saving to. Remember that the temporary file for COOL is taking up space on the TEMP drive, which means you cannot save a file onto your TEMP drive that is larger than one half the amount of free space you had on your TEMP drive before you started. If COOL crashes, there may be a temp file (~COL?????.TMP) in your temporary directory that you should delete.

One advantage to using this method is virtually unlimited waveform sizes, since it depends only on the size of your hard drive. If you have a 500 meg hard drive, you can record one hour of CD quality music. The drawback, of course, is that the temporary file can get extremely large, which may not allow you to save your masterpiece on the same drive. Of course, if you have a second hard drive with enough space, there will be no problem saving the wave.

Note 2:

If you are using a compressed, or double-spaced hard drive, there are some special considerations. Recording high data rate waves (such as 44.1K/Stereo/16-bit) may severely slow down your system. If your system appears to lock up, press the Stop button, then try reducing the play/record buffer size in the Settings menu. Also be aware that the initial recording time left that is displayed when recording can be slightly off, since the amount of space audio data takes up on a compressed drive varies with the type of audio being recorded! Cool will try and estimate the time left every few seconds, so if you see a jump in time left (either a favorable jump, or a not so favorable decrease) it is because Cool is trying the best it can to estimate the remaining time.

Note 3:

For fine tuning your system to take full advantage of both your sound card, and the COOL program, please see the [Settings](#) section.

If you wish to have COOL use a separate directory for the temporary file (other than that pointed to by the TEMP environment string), then add the line "TempOverride = E:\newdrive" to the [Size] section in cool.ini, where E:\newdrive can be any drive, and any directory you want to use for temporary files.

Note 4:

For more accuracy in using any transformation, use 16-bit samples while working with waves. If your board only supports 8-bit samples, you can have them converted on the fly at playback time by choosing the "convert 16 to 8 bit" mode in Settings. To convert an 8-bit sample to 16-bit, open the waveform, copy the entire wave, then say "New" and choose 16-bit, and paste the waveform. Alternatively, you can open an 8-bit file as 16-bit by using File:Open As, or you can use the Edit->Convert Sample Type option.

Note 5:

All temporary files begin with the tilde (~) symbol. The main Cool Edit temporary file containing the currently active waveform begins with ~COL. Undo information is saved in a file starting with ~NDO. Some other functions create files starting with ~CTE and ~CTM. All temporary files have the .TMP filename extension. The rest of the filename is chosen at random when the file is created. If there are no copies of Cool Edit running, none of these files should be present. If you find them, you can safely delete them, as long as Cool Edit is not running. These files can be left behind in extreme circumstances when Cool Edit crashes, or Windows unexpectedly quits while Cool Edit is running.

Note 6:

Cool Edit uses the Windows convention of defaulting to the directory specified in the program manager's

Properties for the program's icon. This is actually a very good method, as you can make several icons and have each point to a separate default working directory. To have Cool Edit remember the last directory you were in would violate this convention.

