Brought to you by Syntrillium Software Corporation
P.O. Box 62255, Phoenix, AZ 85082-2255, USA
URL: http://www.syntrillium.com
Sales: +1-602-941-4327
Fax: +1-602-941-8170
Toll-free sales: 1-888-941-7100 (US and Canada only)

Cool Edit™ 96 is a full-featured digital audio editor for Windows 95 and Windows NT. Click on one of the topics below to find out more.

Overview:

- **What is *Cool Edit* and what can it do?**
- **What's new in *Cool Edit 96*?**

Using Cool Edit*:*

- **Common procedures**
- **Navigating the main screen**
- **Keyboard and mouse shortcuts**
- **Troubleshooting**

How to register your copy:

- **Instructions**
- **Registration form**
- **Other programs from Syntrillium Software**

Cool Features and how to use them:

- **File menu**
- **Edit menu**
- **View menu**
- **Transform menu**
- **Generate menu**
- **Analyze menu**
- **Options menu**
- **Help menu**

Miscellaneous topics:

- **Custom toolbar**
- **File filters**
- **Effects modules**
- **Technical notes**
- **Bibliography**
- **License Agreement**

Cool Edit™ was created by David Johnston.

or by obtaining a registration number by any means, you are agreeing to this license. Registration numbers can only be given out by Syntrillium Software Corporation, or an agent acting on behalf of Syntrillium Software Corporation.

Warranty Disclaimer

Syntrillium Software Corporation makes no warranty of any kind, expressed or implied, including any warranties of fitness for a particular purpose. In no event will the company be liable for any incidental or consequential damages arising from the use of, or inability to use, this program.

What is *Cool Edit 96* and what can it do?

***Cool Edit 96* is a digital audio editor for Windows 95 and Windows NT.** You can use it to record and play files in a wide variety of audio formats, edit files and mix them together, and convert files from one format to another.

But *Cool Edit* is much more than just an editor, recorder, and player: it's like a complete recording studio inside your computer. You can work on multiple files simultaneously with the New Instance feature. You can create sounds from scratch with Generate Tones, Generate Noise, and Generate DTMF Signals. You can "touch up" your audio with features like FFT Filter, Quick Filter, Amplify, Compress, Envelope, Stretch, Channel Mixer, and Noise Reduction. You can add fantastic effects to files like Reverb, Delay, Echo, 3D Echo Chamber, Flanging, and Distortion. You can even adjust the pitch or tempo of your audio with the Stretch function. When you find just the effect you want, you can save it as a Preset for later recall, and you can use *Cool Edit*'s Scripts and Batch Processing features to automate your work.

***Cool Edit* also offers powerful data analysis features.** The Spectral View gives you a multi-color map of the sounds in your files. You can generate a Frequency Analysis to see which frequencies predominate at a point or in a region of your waveform. The Statistics feature gives you information on peak amplitude, minimum and maximum RMS power levels, and more.

***Cool Edit* offers two truly unique features.** Want to hear your dog sing opera? You can do it with the Making Music feature. You can also create Brainwave files to induce states of deep sleep, theta meditation, or alpha relaxation.

Sounds complicated? Don't worry; *Cool Edit* is also easy to use. If you make a mistake, the multi-level Undo feature can back you out. You can quickly learn how to navigate the main screen, and *Cool Edit* offers a variety of keyboard and mouse shortcuts and an editable toolbar. The Cue and Play lists make it easy to find, name, and play segments within a file, and the built-in CD player controls give you easy access to your computer's CD player. If you get into trouble or can't figure out how to do something, just look at the Troubleshooting or Common Procedures Help topics, or send an email to support@syntrillium.com.

Above all, *Cool Edit* is fun! Once you've played with it for a while, you'll see how it got its name.

File menu

The File menu displays file-handling options. Click on one of the items below for more information.

File Menu Options:

- **New**
- **New Instance**
- **Open**
- **Open As...**
- **Open Append...**
- **Re-Open**
- **Close**
- **Save**
- **Save As**
- **Save Selection**
- **Flush Virtual File**
- **1…8 *filename* list**
- **Exit**

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

If you're having trouble with some aspect of *Cool Edit*, you may find a solution below. If you can't find the information you need here, check the Hardware Configurations or Common Procedures Help topics. If these topics still don't help, email support@syntrillium.com, and we'll try to find a solution for you.

Q: Why does *Cool Edit* create files with the extension ".pk" alongside my audio files?
A: These are "peak files". They enable *Cool Edit* to load, save, and redraw files more quickly than it could do without them. You can safely delete these files, and *Cool Edit* will re-create them. If you like, you can also disable creation of peak files entirely in **/Options/Settings/System**.

Q: I am using *Cool Edit* to master .WAV files for use with CD-R equipment, and there is an annoying click after each track. Why is that?
A: Some CD-ROM Recording equipment does not read the RIFF .WAV file correctly, and interprets some of the information chunks as audio. The RIFF specification is available from Microsoft for these companies. In the meantime, you can force *Cool Edit* not to write any extra information by not using the Cue or Play lists (these are saved in the .WAV file as the standard), and by clearing out all information in the **/View/Info** dialog. Uncheck the *Fill * fields automatically* checkbox to prevent *Cool Edit* from automatically filling in the date and software package fields in the future.

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, you might encounter problems starting and stopping audio too quickly.

Q: I've already registered *Cool Edit*, but the features I want to use are still unavailable ("greyed") in the menus and toolbar. What's wrong?
A: Probably nothing is wrong. You must first select all or a portion of a waveform before you can use features like Reverb, Noise Reduction, and others. Try loading a file and selecting some or all of it with the mouse. Once youve done this, you should see those features "light up" in the menus and toolbar.

Q: I'm trying to use *Cool Edit* for the first time, and I can't use /File/Open or /File/Save, and there are only a few features in the Transform menu. Have I done something wrong?
A: You may not have installed *Cool Edit* properly. If you have an automated unZipping or other automatic archive extraction application installed on your computer, and if you simply double-clicked on COOL95.EXE in order to run it and you don't see options like Reverb in the Transform menu, then probably *Cool Edit* is not fully installed, and it can't find its own program files. You must first install (and unzip!) *Cool Edit* into a new directory on your hard drive before you can run it.

Q: Can *Cool Edit* read and write MPEG files?
A: Yes, but you must download the separate MPEG filters from Syntrillium's web site at http://www.syntrillium.com to enable *Cool Edit* to support MPEG.

Q: Can *Cool Edit* convert files, either singly or in batches?
A: Yes! To convert a single file, use **/File/Open** to open the file in *Cool Edit*, select **/File/Save As**, and specify the file type at the bottom of the Save As dialog. To convert multiple files, see Batch File Conversion.

Q: I keep running out memory. How can I free up more RAM?
A: If you are editing very large files (several hundred megabytes or more), try increasing the Peaks Cache in **/Options/Settings/System** to 1024 or even 1536 or 2048 to use less RAM. You can also try reducing the Wave Cache size in **/Options/Settings/System** and/or increasing the virtual memory settings for the system.

Q: I can't record in *Cool Edit*. When I try, I get a flat line, and no sound when I play. What's

wrong?
A: Probably you have the record level set too low for the audio signal you're trying to record. See Adjusting Recording and Playback Levels to find out how to remedy this.

Q: When I record (or play), I hear skipping and dropouts. What can I do to eliminate this?
A: You probably have a too-small buffer size in the Settings dialog. A minimal buffer size is about 2 seconds for fast machines, about 8 for slow ones. Also try decreasing or increasing the number of buffers. Please note that some audio drivers have problems with too many buffers. 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). We have 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 your sound card and your main board, video board, or other installed boards. If the problem persists, see below for a list of suggestions on solving it:

1) Try reducing (or perhaps increasing) the Play/Record buffer size under **/Options/Settings**.

2) Try reducing to as little as 3 (or increasing up to 12) the number of buffers under Settings.

3) Add or update the line in CONFIG.SYS to read "STACKS=16,512".

4) Add the following lines to the SYSTEM.INI file (16meg systems use 2048, and 8meg systems use 1024):

   [vcache]
   MinFileCache=4096
   MaxFileCache=4096

5) Check to make sure that if your hard drive (the one pointed to by the Temp Directory setting in **/Options/Settings**) requires a DMA or IRQ setting, it does not somehow conflict with that of your sound card. This is common for digital I/O cards and SCSI drives.

6) If your sound card has a choice for "Single Mode DMA", do not enable this item.

7) Set the hard disk priority to "Network Server" and caching to 32K lookahead (not the default 64K) in Control Panel-System-Performance-File System.

8) Go to Control Panel/System/Performance/Virtual Memory and choose "Let me specify my own virtual memory settings" and choose a nice fast hard disk (assuming you have one) with at least 30 MB or so free. Then enter 20 for Minimum and 20 for Maximum. If you need more virtual memory, go as high as 60 for Min and Max.

Q: When I try to record or play, I get "RecordVoc Error MMsystem032: The specified format cannot be translated or supported. Use the Capabilities function to view supported formats." What does this mean?
A: This indicates you're trying to record or play a file in a format not supported by your sound card or its drivers. See **/Options/Select Wave Device** to find out what your card can do. Many 16-bit cards, for example, can't handle 48khz, so if you try to play a 48khz file with such a card in *Cool Edit*, you will see this error. Similarly, this error will come up if you try to play a 16-bit file on an old 8-bit sound card. If this is the case, try checking "Play 16-bit files as 8-bit" in **/Options/Select Wave Device**. This will convert any 16-bit audio data to 8-bit as the file is played. NOTE: You can see a chart of your card's capabilities in **/Options/Select Wave Device** in *Cool Edit*.

Q: The program crashes right away if I try to Play or Record anything!
A: The real-time VU meters may be incompatible with some sound cards or their drivers. Try disabling

them by right-clicking on the VU meter at the bottom of the window, and unchecking the "Show on Play and Record" option.

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 cannot load normal wave (*.WAV) files. Why?
A: Do you have the file WAVEPCM.FLT in the same directory as COOL95.EXE? The *.FLT files are required for any file loading and saving. Also, no programs can be running at the same time called "WAVEPCM". If you are running another program whose filename is WAVEPCM.EXE or WAVEPCM.DLL, then rename *Cool Edit*'s WAVEPCM.FLT to something different like WV.FLT.

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 **/Options/Select Wave Device**. 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 will take a very long time. A coprocessor will speed up these operations 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 Edit* 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: Why are some functions not selectable?
A: If you have the unregistered version of *Cool Edit*, you chose which functions you wanted to use for the editing session. The others will not be selectable until you choose them in 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-click on the time (or samples Start and End) window to 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: *Cool Edit* 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 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. We are 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, but *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 if you 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 the ones distributed with your current version of *Cool Edit* into the same directory as COOL95.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.flt 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.flt to wav.flt).

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. As a rule of thumb, you should have as much free space on the temporary files drive as twice the size of the largest file with which you will be working.

Q: How can I save files in TrueSpeech® format?
A: First make sure your file is in 8000Hz/Mono/16-bit format. If it isn't, use **/Edit/Convert Sample Type** to convert it to that format. Then use **/File/Save**, select "ACM Waveform" as your file format, click on "Options", and choose "DSP Group TrueSpeech(TM)" as the file format.

Q: Why do I hear repeating sounds on playback?
A: This may occur for one of two reasons. **First**, after a long operation (say, Stretching an 800MB file) the result is a file that plays OK up to a point but then begins repeating a sound over and over again. Example: if the entire file is a voice saying "It's a small world after all", then the file comes out as "It's a small world after after after after". (Note that it usually doesn't happen to small files, though.) Most likely cause: *Cool Edit* ran out of hard drive space when performing the function. Solution: Free up more disk space and/or turn OFF Undo in **/Options/Settings**. **Second**, if a sound always or nearly always repeats on playback, the most likely cause is an IRQ or DMA conflict between the sound card driver and some other device. Solution: check the IRQs and DMAs being used in config.sys, autoexec.bat, and the System control panel in Windows to resolve the conflict. NOTE: this problem will ordinarily manifest itself with any playback software, including the Windows Sound Recorder (Start - Programs - Accessories - Multimedia - Sound Recorder).

Q: *Cool Edit* 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 a letter to Syntrillium Software or email support@syntrillium.com explaining in detail the bug they've found, and we will respond by either finding a fix in the program or explaining why the program does that.

Q: Why do I hear crackling noises when I enable the "auto-scroll" feature?
A: Some video cards may use all of the CPU's computing time to process their data when you scroll (thus slowing the audio down). This can cause crackling noises or other artifacts on playback. Disable auto-scroll to get around the problem.

Batch file conversion

If you need to convert multiple files from one format (like .WAV) to another (like .AU), you can use *Cool Edit*'s Scripting feature to accomplish this.   Here's how to do it:

1) Copy the text between the asterisks below (*****) to a new text file and save it as BATCONV.SCP. This creates a null script that you can run on batches of files.

2) Launch *Cool Edit*.

3) Choose "Cool Scripts/Batch Process" from the Options menu.

4) Click on the "Open/New" button, and choose BATCONV.SCP.

5) Choose "Batch Conversion", and click on the "Batch Run" button.

6) Select the files you want to convert in the Source Files box.

7) Select the destination directory, output format, and output filename template.

8) Click on Begin Processing to start the conversion.

```
*****
Collection: Batch Conversion

Title: Batch Conversion

Description: Null script for performing batch conversions

Mode: 2

End:
*****
```

Hardware configurations

Here we offer some recommendations for how to best use *Cool Edit* with particular hardware configurations.   You may find these suggestions useful if you are having trouble using *Cool Edit* with your system.

Digital Audio Labs Card D and Card D+:   You may notice skipping or "drop-out" problems in recording or playback with *Cool Edit*.   If so, try reducing the "playback/record" buffer from the default 4 seconds to 2 seconds.

Turtle Beach Tahiti: Use 4 seconds, using 2 buffers in **/Options/Settings** to avoid skipping or popping problems in playback or recording.

Other programs from Syntrillium Software

You can find information on Syntrillium's other products at http://www.syntrillium.com. You can also download the latest versions of our shareware products from there:

Kaleidoscope™ is a screen saver for Windows. It turns your display into just what its name implies: a kaleidoscope, with endless variations of color, movement, and variation. Dancing, spiraling figures fill the screen with every color of the rainbow, creating hypnotic effects that soothe the mind and, of course, prevent screen burn-in. You can choose from among 35 preset kaleidoscope styles, change styles with more than 20 user controls, or create your own presets when you find just the patterns and colors you want.

...but what makes *Kaleidoscope* truly unique is that it responds to music! Just pop your favorite music CD into your computer for a truly hypnotic and mind-tingling experience. Watch the movement and colors change- deep colors and graceful movement when the music is quiet and low, and bright, dancing spirals when it's loud and vibrant.

We've all heard chimes playing in the wind. Perhaps there's no more soothing sound; these random but strangely melodic aural cues can transport you instantly to another time and place: your parents' back porch when you were a child, or the inn by the sea that you visited last summer. *Wind Chimes*™ puts these sounds on your computer. It uses your sound card's built-in synthesizer and sophisticated algorithms to replicate the unpredictable patterns of real wind chimes. It can also do much more: select the "Piano Bar" preset, and you're sipping ice tea at a favorite café. Try "Glorious Sunrise" and you're lying on a secluded beach in the early morning. "Acoustic Journey" boldly leads you to new harmonic and rhythmic concepts in music. Best of all, you can create your own ambient sounds-- select chime types varying from pianos to gunshots, wind speeds from a gentle breeze to a raging storm, and many other settings to create just the ambient soundscape you want. You'll never hear the same pattern twice, because *Wind Chimes*, like the wind, is always changing.

Version history

96
Added vertical zoom and vertical ruler.
Enhanced horizontal zoom (down to the sample).
Added auto-update to frequency analysis when FFT Size <=1024.
Added peak files for faster loading, saving, and redraw.
Added RealAudio® (export only) and DiamondWare® DWD (import/export) file format support.
Added click-and-drag rulers to waveform and spectral views.
Added Statistics function.
Now using internal clipboard by default (supports large copy/paste operations).
Improved Stretch.
Improved Noise Reduction.
Added separate controls for left and right channels in Echo function.
Enhanced Distortion (four-quadrant control).
Improved sample-rate conversion.
Added "live update" while recording in waveform view.
Enhanced customizability in **/Options/Settings**.
Revamped menu structure for easier access to DSP and other features.
Added Most Recently Used (MRU) file list to File menu.
Added automated Setup and Uninstall.
Added integration to CakeWalk Pro Audio 5.0 and above.
Changed the WAV header to match the 'standard' 44-byte wave header expected by some CD-ROM burning software and other older wave handling software.
Correction made so that files without the required 'fact' chunk can still be read properly.
Fixed Text output file format (it was acting as if no disk space was left when trying to save).
Certain extra text info was not being parsed properly when reading in AU files; this has been fixed.
File system configuration changed to allow for larger files (in excess of 2 Gigabytes) and new caching scheme to speed up file operations somewhat.
Improved spectral view (may be faster).
New Gaussian windowing option gives finer detail for spectral components at the expense of extra background noise in the plot.
Auto-play on command-line load option added to General Settings.
Made ruler display easier to read for very large files in SMPTE Drop mode.
Extra RIFF information unknown to *Cool Edit* is preserved when saving

95
Cool Edit has switched to using the 32-bit APIs, and compiled for 32-bit Windows to run on Windows 95 or later and NT 3.51 or later. Most functions have increased in speed by 50% to over 3 times as fast by going to 32-bit and doing more optimizing.
Added the new *Explorer* style dialogs for most File Open and Save dialogs. The *Explorer* dialog style allows one to copy, rename, delete, and move files as well as simply open or save them.
Added option to use old style file open dialogs for those of you who do not wish to use the new *Explorer* style dialogs.
Many minor bug fixes that have increased the quality and stability of *Cool Edit*.
Auto-play in the File Open dialog will now play all formats recognized by *Cool Edit*, not just .WAV files.
Fixed bug in saving and recalling Quick Filter presets.
Hitting Cancel on Not enough memory for undo inadvertently disabled all previous undos.
Added custom toolbar configuration dialog under Options. Also added some new toolbar items for other common operations.
Added Display Time Format menu under Edit.
Save Selection works while saving a portion of one channel of a stereo waveform.
Revised Settings dialog, and added special settings for color selection and spectral view.
Spent an extra couple weeks updating *Cool Edit*s copy protection to keep the program from being cracked and hacked into.

1.52
Sample rate conversion quality has improved even more.
Users of the Lite version of *Cool Edit* now have access to the new high quality sample rate conversion.
Added Amiga IFF file format.
Some glitches in working with scripts have been eliminated.
CD titles are automatically read from the Windows 95 CDPLAYER.INI file.
File Open and File Save As dialog boxes remember nonstandard filename extensions, and remember last used Options settings for each format.
If no selection is made before performing a Transform operation, the current view is used.
A new ruler has been added to the waveform display.
Frequency Analysis display has been improved with decibel calibration and solid line view for more accuracy.
Spectral View has been improved by increasing the dynamic range of the display. Even the faintest signals are now plotted.
Waves play automatically if loaded from the command line.
Left and Right channels can be chosen individually by clicking near the top or bottom of the waveform display.
When loop playing, the looping will change to match the portion being selected.
Added option to control the number of play/record buffers for fine tuning recording quality.
Batch file processing now handles about 5,000 files per batch (up from a previous limit of about 200).
Faster sample rate conversion.
Faster Filter function in "locked" mode when not dynamically changing filter over time.
Faster Echo Chamber function by 5 to 20 times or more depending on room configurations.

1.51
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 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' characteristic of Reverb function.
Quick Help no longer disables the Alt key.
Undo Music was corrupt, it's fixed now.
Drive letter only (e.g. "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.
Added Zero Crossing function to adjust cursor position or highlight to zero crossings. Use F4 to quickly snap to zero crossing.
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)
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

There weren't supposed to be any more bugs. But we 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.flt), 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 COOL95.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 most sound cards, 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:
Doug Coulter, Becky Allen, Sara Allen, Bill Wolford, Staffan Berger, Curtis Palmer, Dan Black, Darrell Plank, David Norris, Duane Young, Geoffrey Tilga, Gordon Currie, Harry C. Koons, James Blackwell, Jarrod Beckman, Mike Rozak, Neil St. Laurent, Randy Omel, Richard Pepin, Ron McCabe, Ronald Mendoza, Stan Eker, Ted Tatman, Todd Frye, Will Urban, and Mike Harvey.
Other *Cool Edit* Fans who made suggestions and found bugs here and there.

Other Acknowledgments
All format types have been reproduced from publicly available documentation. Syntrillium Software Corporation makes no claims that they are in any way correct, so they may not be compatible with other interpretations of the same format. It is up to you to first test to make sure that when saving audio files in a particular format, the file generated is compatible with other systems that read the same format.

Syntrillium Software Corporation
David Johnston Bob Ellison Lise Lambert Matt Bieber Ondra Witherspoon

Alexandra von der Gablentz.

Common procedures (how to...)

If you are asking "How do I...?" you may find the answer here.

HOW DO I:

Adjusting recording and playback levels

Cool Edit doesn't control the recording (gain) or playback (volume) level directly, but you can adjust it with the mixer that came with your card or with the mixer built into Windows 95. You may need to do this if your recordings are too quiet or if you can't hear them when you play them in *Cool Edit*. Here's how to do it with the Windows 95 Volume Control applet:

1) Click on Start - Programs - Accessories - Multimedia - Volume Control

2) To adjust the Play (output) level, make sure Select is checked for the source you want to use and move the slider up to the level you want.

3) To adjust the Record (input) level, select Properties from the Options menu in the mixer and click on the Recording button in the "Adjust volume for..." box. Then check the source you want to use and adjust the slider.

If you want to monitor the audio while recording, you may need to look for a "monitor while recording" setting for your sound card. This can sometimes be found by pressing the Advanced button (if available) in the Recording Control dialog.

NOTE: The "Record" (input) and "Play" (output) levels are separate settings. One controls the input level of the audio source, and the other controls the volume of the sound going to your speakers.

Opening wave 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 Edit* as you 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 and choose **/Edit/Paste**, 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 *Cool Edit 96*, the best method for converting sample rates is to use the Convert Sample Type function (**Edit** menu). This function incorporates method 3 below and performs it automatically.

There are various methods for converting sample rates:

1) Simply copy the waveform you wish to convert, open a new instance of *Cool Edit*, and say New. Choose the new sample rate, bit rate, and so on. Then Paste. When *Cool Edit* 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.

2) A slightly cleaner method is to use the Stretch function (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), 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 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.

3) The best method for downsampling or upsampling without sacrificing quality (in fact, upsampled waves sound better than the original at a lower sample rate on some sound boards) is to filter out all frequencies that cannot be represented in the new sample rate *before* doing the sample rate conversion. And the sample rate conversion itself should be phase corrected so the sample sounds as if it were sampled at the destination sample rate to begin with. The Convert Sample Type function uses this method.

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 its 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/Mix Paste** 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, 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/Mix Paste**. 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/Mix Paste**. 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 want the sound source to 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 Effects/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/Special/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 upon which to base your effect. The basic method is to create a few seconds of tones or noise, and then use the transformation functions to manipulate the wave and create the desired effect. With these two sound sources, noise and tone, you can create practically any effect.

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 are limited only to your imagination!

Function presets

Many of the functions have presets available for easy recalling of your favorite settings. You can add new presets at any time. All preset information is saved in the COOL.INI, which is usually in your Windows directory.

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

Whenever you have settings you would like to keep, enter a name for your settings, and press the Add button. Note that nothing will prevent you from adding two presets with the same name. T avoid confusion, make sure that you give your presets different names.

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 under the same name.

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

If you find 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 its 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, we apologize. New versions sometimes necessitate changing 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, our apologies if this happened to you.

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, you can download the *Cool Edit* File Filter API from Syntrillium's web site at http://www.syntrillium.com. This application and the file filters were written in C (not C++) using Microsoft Visual C/C++ 4.0.

If you have written file filters in the past, and you find they do not work for *Cool Edit 96*, its probably because they are 16-bit DLLs, and *Cool Edit* requires 32-bit DLLs. Try recompiling and fixing up for 32-bit, and change the VALIDLIBRARY define to 1152.

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, you can download the *Cool Edit* Transform and Generate Effects API from Syntrillium's web site at http://www.syntrillium.com. This application and the effects modules were written in C (not C++) using Microsoft Visual C/C++ 4.0.

Undo

If the Undo function (**Edit** menu) is enabled, (see Settings), you can back up 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 is specified in the Settings dialog.

When *Cool Edit* is saving Undo information, you can press the "Skip" button to abort the undo saving process. Of course, Undo will not be active afterwards, so be careful! You will be warned of this in a dialog box each time, unless you choose "Cancel" to never see the dialog box again.

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 (**File** menu) closes 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 previously used.

Scrolling through a 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, because the portion being viewed with respect to the entire wave is smaller.

You can click and drag the green bar at any time to scroll the portion being viewed left or right. 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 above the waveform display brings up the Viewing Samples text entry box. You can enter exact starting and ending points in this box.

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 want to 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 can also create more new instances and open waves into them.
- Position the cursor where you want to place the start of the copied waveform, and choose **/Edit/Mix Paste**. Make sure the Overlap box is checked. Choose the mixing volume, and press OK.
- Do this for each wave you want to mix.

Edit menu

The Edit menu displays all the waveform editing options.

Edit Menu Options:

- **Undo**
- **Enable Undo**
- **Repeat Last Command**
- **Copy**
- **Windows Clipboard**
- **Cut**
- **Paste**
- **Mix Paste**
- **Delete Selection**
- **Trim**
- **Select Entire Wave**
- **Zero Cross Adjust**
- **Adjust Sample Rate...**
- **Convert Sample Type**

Transform menu

Much of *Cool Edit*'s power is in the Transform menu, because it contains the "effects" (DSP functions) that can make your audio sound, well, really *cool*. Click on one of the items below for information on that effect.

General:
Invert
Reverse
Silence

Amplitude:
Amplify
Channel Mixer
Dynamics*
Envelope*
Normalize

Delay Effects:
Delay*
Echo*
Echo Chamber*
Flanger*
Reverb*

Filters:
FFT Filter*
Quick Filter*

Noise Reduction*

Special:
Brainwave Synchronizer
Distortion*
Music

Time/Pitch:
Stretch

*NOTE: This function is not available in the Lite version.

Generate menu

Cool Edit doesn't just edit sound; it can create it as well. Click on one of the items below to find out how to add tones, noise, or silence to your waveform.

Generate Silence
Generate DTMF Signals*
Generate Noise
Generate Tones

*NOTE: This function is not available in the Lite version.

Options menu

The Options menu displays various options for customizing *Cool Edit* and accessing special features.   To find out more, click on one of the items below.

Loop Mode
Monitor VU Level
Scripts and Batch Processing...
Select Wave Device
Custom Toolbar Settings…
Settings...

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.

Navigating in *Cool Edit*

Cool Edit offers a wide variety of features and controls to make it easy to find, view, and edit your files. See below for more information on how to move around and speed up your work. You may also want to take a look at the Keyboard and Mouse Shortcuts and Cue and Play list topics for more information on those subjects.

Toolbar:

Commonly used functions are represented as buttons in the Toolbar below the menu. Hold the mouse over any of these buttons to see a Quick Help message describing the function in more detail.

Rulers:

The **vertical ruler** to the right of the waveform display indicates the amplitude of the wave, and the **horizontal ruler** below the waveform shows the time dimension. You can configure each ruler or zoom to a new position by right-clicking on the ruler. To quickly change the scale for either ruler, double-click on it. To move the display right, left, up, or down, click and drag on the appropriate ruler. To zoom to a particular portion of the ruler, right-click and drag over that portion.

When you have zoomed in on a portion of a wave, you can click on and drag the **View Indicator** above the waveform OR the ruler below the waveform to move the view to a new location. To quickly jump to a new position, use **/View/Viewing Range** or double-click on the green bar above the waveform and enter the starting and ending sample values that you want to view.

Selecting all or portions of a wave:

Click on one side of the portion you want to select and drag left or right to select it.

Right-click on the waveform (or use the cursor keys) to extend the current selection.

Double-click on the waveform to select the entire visible wave.

Click near the top of the left (upper) channel or near the bottom of the right (lower) channel to select that channel only. The mouse cursor acquires an "L" or an "R" when you do this.

Zooming:

Choose **Zoom** to expand the selected portion horizontally to full width. **Zoom In** zooms to the middle of the current view, regardless of the location of the selected portion. **Zoom Out** enlarges the view of the waveform, while **Full** displays the entire waveform in the workspace.

Use **Ctrl+Up** and **Ctrl+Down** to zoom in and out vertically. Click on Full to reset vertical zoom to normal.

You can move selection currently being viewed by clicking and dragging the green bar in above the waveform. To make the viewed selection larger or smaller, right-click and drag on the green bar.

Note: when you have viewed in far enough, *Cool Edit* shows the **individual samples** as small squares on the waveform line (the line itself represents an analog interpretation of the digital data). You can make fine adjustments to individual samples by clicking on them and dragging them up or down. Double-click on the sample values to see the current sample value or to edit the value directly (by entering a new value).

Playing and recording wave files:

The wave Play and Record buttons are located in the lower left area of the main window. Click on **Play** to play the portion of the wave that is currently being viewed, or the portion that is highlighted. If Loop mode is enabled, the Play button changes to Loop.

Click on **Pause** to 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.

Click on **Record** to start recording at the current insertion point. Any waveform data after that will be recorded over.

Click on **Stop** to end waveform playback or recording.

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.

Markers:

The **Yellow Arrows** above and below the waveform indicate the current cursor position (the point of insertion).

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

The **Blue Brackets**, when present, represent the Cue List entries for a selection range.
To select a Cue range, click on one of the markers or between the two blue brackets.

VU (Record Level) Meters:

The **Record Level Meter** below the Play/Record buttons displays the current peak amplitude of the audio being monitored, recorded, or played in real time. To check the record level, double-click on the level meter. Right-click on the meter to configure it.

Other indicators:

The **Time Display** boxes in the lower right area of the main window show the current Starting and Ending points of the current selection, or portion being viewed. To toggle the time display between seconds, samples, and frames, double-click on the appropriate Time Display box.

The **Wave Format** window displays the format of the wave in Channels, Sample Rate, and Bits Per Sample. To change the sample type, use **/Edit/Convert Sample Type** or double-click on the Wave Format box.

If an audio CD is in the CD-ROM drive, pressing the CD button will display the CD Player Controls at the bottom of the window.

Shortcuts

Cool Edit offers a wide variety of keyboard shortcuts to make editing easy and fast. See the list below to find out how to use the keyboard and mouse to speed up your editing.

KEYBOARD

Selection controls:

Ctrl+B	Select Both Channels
Ctrl+L	Select Left Channel
Ctrl+R	Select Right Channel
Ctrl+S	Select Entire View
Ctrl+A	Select Entire Waveform
Left	Adjust *left side* of highlight one pixel to the left
Right	Adjust *left side* of highlight one pixel to the right
Shift+Left	Adjust *right side* of highlight one pixel to the left
Shift+Right	Adjust *right side* of highlight one pixel to the right
F4	Adjust both left and right sides of selection to zero crossings
Escape	Unselect (if any selection made, it is unselected) and reset cursor to start

Editing controls:

Ctrl+C	Copy selection to internal clipboard
Ctrl+T	Trim to selection
Ctrl+V	Paste from internal clipboard (or Windows clipboard if internal clipboard is empty)
Shift+Insert	Paste from internal clipboard (or Windows clipboard if internal clipboard is empty)
Alt+Backspace	Undo
Delete	Delete selection
Shift+Delete	Cut selection to internal clipboard
Ctrl+Insert	Copy to internal clipboard
Ctrl+X	Cut waveform to internal clipboard
Ctrl+M	Mix paste
Ctrl+Z	Undo

Play and Record controls:

Space	Toggle Play / Stop
Ctrl+Space	Toggle Record / Pause
Alt+S	Stop (like Space when already Playing)
Shift+Space	Toggle "Play from Cursor To End" / Pause

View and Zoom controls:

Ctrl+Down	Vertical Zoom Out
End	Jump view to end of wave (doesn't affect cursor)
Ctrl+End	Zoom into right side of wave selection or cursor
Home	Jump view to start of wave (doesn't affect cursor)
Ctrl+Home	Zoom into left side of wave selection or cursor
Ctrl+Right	Zoom "In" (zoom to center of view)
Ctrl+Left	Zoom Out
Ctrl+Up	Vertical Zoom In
Page Down	Scroll forward one screenfull (doesn't affect cursor)
Page Up	Scroll backward one screenfull (doesn't affect cursor)

Miscellaneous:

Alt+I	Waveform Info box
F2	Repeat last command (bring up dialog if applicable)
F3	Repeat last command with last parameters (no dialog)
F8	Add current cursor location or selection to cue list. If playing, add current play

	location to cue list
Enter	If CD window is up, accept text changes made in CD title window
Tab	Go to next song if CD window up
Alt+Z	Bring up Frequency Analysis dialog

MOUSE

Left click and drag on waveform to highlight and select a range of samples
Click and drag near the top or bottom of a stereo wavform to select a single channel
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 the waveform ruler to change the ruler format.
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
Right click on vertical ruler (right side of waveform) to zoom vertically or select vertical scale
Right click on horizontal ruler (below waveform) to zoom in or out or select time scale
Double-click on an individual sample to see the current sample value or to edit the value directly

Opening a waveform

Cool Edit supports a wide variety of data types. When you load a file for editing, *Cool Edit* converts the waveform type to its own internal temporary file type for faster editing and larger edit file size. You can then save the wave in any of the supported formats you want.

When the **Auto Play** box is checked, *Cool Edit* will play any wave you click on (provided that your system has a driver that understands the format of the wave being highlighted). You can also play individual files when Auto Play is un-checked by selecting them and clicking on the **Play** button, and you can stop Auto Play for the current file by clicking on the **Stop** button (which replaces Play when Auto Play is playing).

When the **Show Info** box is checked, *Cool Edit* basic information about the audio file, such as the file format, uncompressed file size, and running time.

When **Don't ask for further details** is checked, *Cool Edit* will not prompt you for more information about the file format after you select a file to open. For example, if "Don't ask" is checked and you double-click on a raw (headerless) PCM file, *Cool Edit* will not prompt you for the sample rate, bit resolution, A-law/mu-law compression, or other information about the file. Instead, *Cool Edit* will use the last settings specified when you opened a headerless file.

See Wave Formats for descriptions of the various formats.

See Open As for information on forcing *Cool Edit* to open a file as a specified format. To append another waveform to the end of the current one, see Open Append. You can open several files at once (appended one after another) by selecting them with the Shift (for contiguous selection) or Ctrl (for non-contiguous selection) key pressed. When you open multiple files simultaneously, *Cool Edit* converts all file types to that of the first file that is opened.

Appending a waveform

You can append any waveform to the end of the current waveform with **/File/Open Append**. If the waveform being appended is of a different type, it will be quickly converted as it is being copied. You will get best results if you only append files of the same sample rate, because in this case no interpolation is required. See Open for more information on the Auto Play and Show Info options.

See Wave Formats for descriptions of the various formats.

Wave file formats

The following is a list of the various file formats that *Cool Edit* currently supports. Note that if you want to load from or save to a format that is not listed here, you may be able to use an ACM Waveform driver to do so. To do this, use **/File/Open As** or **/File/Save As**, click on **Options**, and try to find the format you want to use. When exporting to an ACM format, you may first need to use **/Edit/Convert Sample Type** to convert the file to a format supported by the ACM driver.

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 forma 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 we have found none that actually work properly. In building this format, we followed the specification to the letter without making any assumptions. In the past, we 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 we 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.

We 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. We 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. We have found that the preceding index tables provided the best quality.

CCITT mu-Law and A-Law waveforms (.WAV)
These formats compress 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 that 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, .SND)
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 tansferred the wave to the PC from a Mac, you can name the file with the .AIF extension, and load it

using this file filter. If you want to save 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. The 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, .SND)
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 Mono 16-bit audio. This format has no header, so any file format with the extension VOX will be assumed to be in this format. When opening VOX files, you will be prompted for sample rate unless "Don't Ask" is checked.

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.

Amiga 8SVX (.IFF, .SVX) (*.*)
The Amiga 8SVX format is an 8-bit mono format, which can also be compressed to a 4-bit Fibonacci delta encoded format.

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. Some formats that come standard with Windows 95 like GSM 6.10, and DSP Group TrueSpeech are supported through this file format. If you own a Sound Blaster card, for example, the Creative ADPCM file format will also be available. If a format above is not available for your specific sound card, ask the sound card provider for ACM drivers that support their formats. See Using ACM for more information.

DiamondWare DWD (.DWD)
This is the audio format used by DiamondWare's Sound Toolkit. This is a programmer's library that lets you quickly and easily add high quality interactive audio to games and multimedia applications. See http://www.dw.com for more information.

RealAudio 3.0 (.RA)
Use this format to create files for Progressive Networks's streaming audio servers and players. Click on the Options button in **/File/Save** to select the specific sub-format you want. NOTE: *Cool Edit* can only export to RealAudio format; it cannot import .RA files. *Cool Edit*'s RealAudio filters support RealAudio 3.0.

MPEG Layer I & II (.MP2)
The MPEG file filters for *Cool Edit* are available free of charge from the Syntrillium home page at http://www.syntrillium.com. The Motion Picture Experts Group audio file format gives the highest compression with the least amount of quality loss, but is also the slowest. In most cases, it takes longer to load or save an MPEG file than it does to play it. Layer 3 MPEG gives even higher quality, but is not yet available for *Cool Edit*. Only the sample rates of 48000, 44100, and 32000 are supported. Any other sample rates will be converted before being saved to this format. You can, however, fool the codec into using other rates by choosing **/Edit/Adjust Sample Rate** and specifying one of the valid rates before saving. Then, after re-loading one of these files, go back and re-adjust the sample rate back to the correct one.

The ISO MPEG standard is available from the International Standards Institute, but here is a brief overview of the compression options:

Compression: Layer I compression sounds worse at the same compression ratios as Layer II, but may be compatible with more older MPEG players that do not support Layer II. Unless you are striving for compatibility with an old MPEG decoder, use Layer II.

Psychoacoustics: This refers to the way a person perceives audio, and the model used determines which frequencies are assumed to mask other frequencies. That is, if a certain sound cannot be heard over another, that sound is not saved. In tests, it seems that Model 2 (AT&T) sounds best, but you may wish to try each model and compare since it may also be heavily dependent on the type of audio being compressed.

De-emphasis for decoder: If the audio has been pre-emphasized, you should choose the type of de-emphasis you would like the decoder to perform. *Cool Edit* has no pre-emphasis functions, so in general this option should be set to None.

Data Rate/Compression Ratio: The higher the compression ratio you choose, the lower the quality of the sample will be, but the less space it will take on your hard drive as well. Any rates marked with a double-asterisk (**) are not standard, and there may be many decoders that will not be able to decode the file.

Cool Edit will always be able to decode these non-standard rates. At the highest quality, the compression ratio is still about 4:1, making the MP2 file 1/4th the size of a non-compressed wave file.

Enable Error Protection: If checked, some extra redundancy error correction bits are placed in the file. If some bits are in error, they will be corrected. The decoder will still be able to play the file correctly even if the file is somewhat damaged.

Copyrighted Material: If the material you are saving is copyrighted, you should check this box.

Original Material: If the material is original (not a copy), check this box.

Joint Stereo: For stereo files, choosing Joint Stereo will combine the lower frequency bands and encode them as if they were mono with some extra information so the decoder can still place the audio into the proper channels. Normally, one cannot tell the difference between audio saved with Joint Stereo or without, except for the fact that with Joint Stereo in effect, it may sound better, because there will be more room to save information about the higher frequencies.

Choosing a directory

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

Opening a waveform as any format

The **Open As** function (**File** menu) 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, 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. *Cool Edit* will open the waveform and convert it.

Re-opening a waveform

Use **/File/Re-Open** to reload the previously loaded waveform. *Cool Edit* will discard any changes that were made to the edited waveform since the last time you saved the file.

Editing individual channels

You may sometimes want to edit only one channel of a stereo file. Normally, both **/View/Left Channel** and **/View/Right Channel** are checked, meaning both channels are being edited simultaneously. To edit only one channel, uncheck the channel you want to preserve so that 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.

You can also select an individual channel by clicking near the top (for the left channel) or bottom (for the right) of the waveform. When you do this, the mouse cursor acquires and "L" or an "R".

Viewing Range

Select **/View/Viewing Range** or double-click on the green/black samples portion bar to bring up the viewing range window.

Enter the leftmost and rightmost samples that you want 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.

Display Time Format

Select **/View/Display Time Format** and choose either Decimal, SMPTE Drop, Samples, or Custom to change the format of the Begin, End, and Time windows. Double-click on the waveform ruler or the Begin time window to change the format of the ruler. To change the Custom time format, select **/View/Display Time/Format/Define Custom**.

New Instance

... Or "Give me another window"

Select **/File/New Instance** to 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 and don't want to disturb the original wave. *Cool Edit* supports mixing between waveforms in different windows; just use Copy and Paste to insert or **/Edit/Mix Paste** to overlap and mix.

If you tend to use a large number of *Cool Edit* windows open at once, you may want to decrease the wave cache so that each instance of *Cool Edit* will use less memory on your system. See **/Options/Settings/System** for information on the wave cache.

Frequency Analysis

Choose **/Analyze/Frequency Analysis** to perform a frequency analysis on the current position in your file. This brings up an analysis dialog that contains a graph of the frequencies present at the insertion point (yellow arrow cursor) or at the center of a selection. *Cool Edit* performs a Fast-Fourier-Transform to determine the frequencies present, and it interpolates and displays the most prominent frequency. You can see the frequency and amplitude of a given point on the plot by moving the mouse over the graph area.

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

FFT Size - Higher FFT sizes will give more accurate results in terms of frequency (such as the overall Frequency estimate), but higher sizes are also much slower.

FFT Window - Different window types will give different frequency graphs. The Triangular window gives a more precise frequency estimate, but is also the *noisiest*, which means other frequencies will be shown as present, even though they may be much lower in volume. At the other extreme, the Blackmann-Harris window has a more broad frequency band which isnt as precise, but the sidelobes are very low, making it easier to pick out the major frequency components.

Linear View - When checked, the horizontal scale of the plot is linear; when not checked, it is logarithmic.

Range - You can specify a range from 24 to 240dB for the vertical scale of the plot.

Click **Scan** to scan the highlighted selection and show all frequencies present in that selection.

Continuous Frequency Analysis - when you set the FFT size to 1024 or lower, the frequency analysis window updates while you play your file. You can also generate a step-by-step animation by clicking on the main waveform window and then holding down on the Right Arrow key. As the cursor scrolls across the display, *Cool Edit* displays the spectral information in the analysis window.

To gain higher resolution and see more detail in the lower frequencies, use Convert Sample Type to downsample the waveform to a lower sample rate. The highest frequency value displayed will be one-half the new sample rate.

If stereo data is being viewed, the left channel will be shown in Cyan and the right will be shown in Magenta.

Check the **Linear View** box to display the plot with a linear horizontal frequency scale. Un-check it to display the plot on a logarithmic scale.

Waveform and Spectral View

Cool Edit offers two modes for viewing waveforms: Waveform View and Spectral View (**View** menu). **Spectral View** enables you to display waveforms by their frequency components. This is a handy function for analyzing your audio data to see which frequencies are most prevalent throughout your data.

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

In Spectral View, the more abundant a frequency, the brighter the display color. 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 or higher color displays, there will be more gradations between the colors. See Settings for information on various settings for spectral view. The frequency resolution, window type, colors, and energy plot can all be fine tuned.

Lower frequencies are displayed near the bottom of the display, and higher frequencies are displayed near the middle or 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. 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.

To gain higher resolution and see more detail in the lower frequencies, use Convert Sample Type to downsample the waveform to a lower sample rate. The highest frequency value displayed will be one half the new sample rate.

Select Wave Device

If you have multiple sound cards, or multiple output devices (such as a sound card *and* the PC speaker), use **/Options/Select Wave Device** to choose the input and output devices you want to use.

If your system is equipped with MIDI devices, you can 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.

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

FFT Filter

Use **/Transform/Filters/FFT Filter** to remove 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.

You can enter **Graph Values** in dB or percentage (depending on logarithmic or passive mode) 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 when separate initial and final settings are used. 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. Because 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 **FFT Size** 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.

When **Log Scale** is checked, the graph is logarithmically displayed in frequency, which is closer to how the ear hears sound. To do finer editing in low frequencies, leave Log Scale checked. For detailed high frequency work, or work with evenly spaced intervals in frequency, uncheck it.

The **Points** parameter

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, 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.

The noise level of the filter is lower than that of 16-bit samples, so there should effectively be no extra noise induced by using this filter depending on the Window being used. A Blackman window, for example, will have the stop band noise below the -96dB mark.

This function supports Presets.

For best results, filter using 16-bit samples. If your source is 8-bit, *Cool Edit* will automatically convert to 16-bit to do the filtering, and convert back to 8-bit. But if you are doing multiple edits, you should use /Edit/Convert Sample Type to convert to the 16-bit format. You can convert back to 8-bit when you have completed processing.

FFT transition settings

In the FFT Filter, you can 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, and the right 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, and 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

You can use **/Transform/Special/Music** to put your clippings to music, or to 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. Selecting music a second time will therefore 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 them 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 to the end of the song. The music is played through channels 1 and 13 for Extended and Base level compatibility. You can choose the instrument 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 we 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

Use **/File/Save** to save the file currently being edited. Waveforms can be saved in many formats. Having many format options gives you the option 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

/File/Save Selection is identical to Save except that the highlighted selection is saved, and not the entire waveform.

Immediately saving a waveform

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

If you do not want to save header fields such as the copyright, author, and others (see Info), un-check the "Save extra non-audio information" box.

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

Select **/Edit/Repeat Last Command** or press **F2** to repeat the last function that modified waveform data.

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

Settings

You can customize *Cool Edit*'s colors, use of memory and hard disk space, spectral view, behavior when pasting, and miscellaneous other settings in **/Options/Settings**. See below for information on the individual settings available.

General:

Highlight after paste: After doing any Paste operation, you can choose to 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.

Use old style file open/save dialogs: If checked, the File Open and File Save dialogs will be similar to those for 16-bit Windows applications, instead of the new Explorer style. You may want to use these dialogs if you are used to them and dont want the extra features of Explorer dialogs such as New Folder, Delete, Move, List View with file sizes and dates, etc. One advantage to the older style is that directories are listed separate from files.

Auto-play on command-line load: If checked, then when you load a file from the command line when you launch *Cool Edit*, *Cool Edit* will play the file automatically upon loading the file. For example, if you choose Start - Run and enter "c:\cool\cool96.exe thisfile.wav" as your command line, then *Cool Edit* will play thisfile.wav immediately upon loading it.

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.

Live update during record: If enabled, the waveform display will update in real time as audio is being recorded. Otherwise the display is turned off during record.

Auto-scroll during Play: If enabled, the waveform display scrolls to keep up with the playback when you have zoomed in on a portion of a wave and then play past the viewed portion (by pressing Shift+Play, for example).

Show center line on top: If enabled, the center line for each channel is displayed on top of the waveform display itself. If disabled, the waveform is displayed on top of the center line.

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.

Custom Time Code Display: Double-clicking on the time boxes or waveform ruler will change their display format. When in the custom Hours:Minutes:Seconds:Frames format, you can customize the number of frames per second that will be displayed. Some common settings are 30 (for SMPTE non-drop), 25 and 50 for PAL, and 75 for CD-ROM mastering.

System:

Total Buffer Size: This is 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).

Number of Buffers Using: This number of buffers may affect recording quality in that some audio drivers may not be able to handle a large number of buffers accurately. If you experience any stuttering or missing (chopped out) audio, try reducing the number of buffers. Also experiment with the total buffer size, as reducing the number of buffers will increase the size of each buffer, since the total of all buffers will be roughly equally to the time specified by the Total Buffer Size setting.

Wave Cache: By default, *Cool Edit* maintains its own audio data buffer, and each instance of *Cool Edit* reserves the amount of memory specified in this field, so if you need to use several sessions of *Cool Edit* simultaneously, reduce the size to 512K or so, or click Use System's Cache (to use the Windows cache rather than *Cool Edit*'s. Recommended cache sizes are from 1024kb to 2048kb. Cache sizes greater than about 2048kb will tend not to increase speed notably for most processing, and sizes below about 1024kb will tend to slow processing down.

Peaks Cache: This determines the number of samples per block to be used when storing peak (.pk) files. Peak files enable fast loading of .WAV files and fast displaying of large files. If you are working with very large files (several hundred megabytes or more in size), you should consider increasing the Peaks Cache to 1024 or even 1536 or 2048 to use less RAM, if RAM is critical on your system. Larger peak file block sizes will reduce the RAM requirement for large files at the expense of slightly slower drawing at some zoom levels.

Use System's Cache: Choose this to let Windows handle all disk caching. Keep in mind that *Cool Edit* usually handles caching better than Windows can. But, this option reserves the least amount of memory, so may be desired for systems with low RAM (less than 16MB).

Asynchronous Access: It is best to leave this option checked. It enables multiple file read and write operations to go on at the same time, and if the system supports asynchronous hard drive access, things will run smoother and faster. Windows 95 does not support this, but other (and future) operating systems do (and will), and leaving the option checked will not adversely affect operation on Windows 95.

Save Peak Cache Files: If enabled, .WAV files will have peak files saved with them with the extension ".pk" following the original file name. Un-check this box if you do not want *Cool Edit* to save peak files on your hard drive.

Temp Directories: more than one temp directory can be specified, and *Cool Edit* will see the combined effect of the temp directories as a single large temp directory. For best results, select separate physical hard drives for the primary and secondary Temp Directories. You can specify an amount to leave "free" for headroom purposes on both the primary and secondary directories.

Enable Undo: Check this box to enable the Undo feature. If you are working with large files and/or have limited hard disk space to work with, you may want to disable Undo.

Undo Levels: This number specifies how many levels of undo *Cool Edit* will save. Note that this is a minimum figure; *Cool Edit* will create more undo levels if there is enough space available. It will also remove undo levels as necessary, and when *Cool Edit* must remove undo levels beyond your minimum setting, it will warn you about it and given an option to cancel the operation.

Purge Undo: Click on this button to purge all Undo levels below the specified minimum undo levels and free the hard disk space used by them. For example, if you have 5 levels of Undo set, Purge will free all levels below level 5, so you will have at most only 5 levels of Undo after the purge. To purge **all** undo levels, uncheck "Enable Undo" and the entire undo history will be removed when you do the next operation on the waveform. You can also remove all undo levels by entering 0 for the number of undo levels and then pressing Purge to expunge the entire undo history.

Colors:

You can select nearly any color used in waveform displays, and you can save your favorite color schemes as Presets. To set colors, click on the appropriate Display Element, and then click on the Color button to change the color for that element. *Cool Edit* displays the current scheme in the Example window. When you have found the scheme you want, click on Save As to store the scheme as a Preset. To delete a color scheme, select that scheme and click on Delete.

Spectral:

Windowing Function: This is the function that will be used to window the data before being displayed. In general, it is best to keep this at Blackmann or Blackmann-Harris. The windows are listed in order from those with the narrowest frequency band but the most noise to those with the widest frequency band but the least extra noise.

Resolution: This setting specifies the number of frequency bands to be viewed, and the logarithmic and linear energy plots have adjustable range and scaling values.

Energy Plot: Choose *Logarithmic* to see every nuance of the audio. The color chosen will be based on the decibel amplitude of the energy present at the particular time and frequency. The quietest signals will be displayed in some color. Choose *Linear* to color the display based on absolute amplitude percentage. Linear is sometimes useful to see the general overview of a signal without getting bogged down by detail at much quieter levels.

Reverse Color Spectrum Direction: If checked, *Cool Edit* "reverses" the palette of the Spectral view color scheme.

Data:

Dither Transform Results: When enabled, a higher dynamic range is possible when processing audio, and the results are cleaner, with less distortions and negative artifacts. *Cool Edit* uses higher than 16-bit arithmetic for most processing, but the results must be converted back to 16-bit when complete. If this option is disabled, the results are truncated to 16 bits, thus losing the more subtle information. If enabled, more information about the audio is retained by adding dither. The drawback is that a small amount of white noise is added at the quietest volume level, and each operation you perform on the data will add a little more of this noise. However, the trade-off between using dither (thus adding noise) and truncating the data (thus creating artifacts and correlated quantization noise) generally favor using dither, so it is best to leave this option enabled. With dithering, you get almost 24-bit sample performance in only 16-bits, as the dynamic range is increased by about another 10dB, allowing singals as quiet as -105dB

Smooth All Edit Boundaries: If checked, some functions, such as Amplify, will automatically crossfade at the starting and ending boundaries of the selection being edited to smooth abrupt transitions at those endpoints. You can specify the crossfade duration in milliseconds.

Auto-Convert Settings for Paste: When pasting different sample formats, *Cool Edit* uses these settings when auto-converting the clipboard to the current sample format. Valid settings range from 30 to 1000.

Cue List

Select **/View/Cue List** to access *Cool Edit*'s Cue List feature. 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. 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. You may enter a maximum of 96 cues.

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. Can also be used as a comment.

Goto (double clicking)
Double clicking a cue item selects that item in the waveform.

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 'ltxt' 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 want *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. One great use for markers is to highlight a wave from the zero crossings. Go to the start of wave portion you want 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. Zoom out, go to the end of the wave portion, and once again zoom in to find the ending zero crossing. Then, 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. Voilà! You can choose "Zoom In" 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 want 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

Select **/View/Play List** to access the Play List feature. This 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. The maximum size of a play list is 64 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
This determines the number of times 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

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

Trim

/Edit/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 only one channel is selected, your edits (copy, paste, etc. and so on) are applied to only one channel, completely independent of the other.

Digital Delay

With **/Transform/Delay Effects/Delay**, you can 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 be the delayed signal, keep it as 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), 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

With **/Transform/Time\Pitch/Stretch**, you can change the tempo or pitch of a wave. You can even change the tempo without affecting the pitch, or vice versa. This is especially useful for adjusting the pitch or of two waveforms to match each other, or for slowing down a recording to decipher the individual notes being played. You can use a *Constant Stretch* to change the pitch and/or tempo of the entire selection by a constant percentage, or a *Gliding Stretch* to stretch the selection in linear fashion from one ratio to another. This gives the effect of slowing down and speeding up, or raising and lowering pitch.

Ratio, Length: You can specify the Ratio and Length with the slider or by entering exact values in the Ratio and Length boxes. Entering one automatically changes the other. If the initial and final lengths are different, then the actual final length will be exactly (initial + final) / 2 when in Preserve Pitch mode.

Transpose: To shift the pitch by a specified number of half-steps (sharps or flats) on the musical scale, select a value in this box. The numerical value for transposing musically is automatically entered into the Ratio box. For example, to make your sound as if it were one half-step higher (if played on a keyboard, and black keys included) choose 1# for 1 sharp. The 'b' values will lower the pitch by one half-step at a time.

Precision: High Precision is ideal for voice, while Medium Precision is generally just as good for complex music. Select Low Precision for faster processing.

Stretching Mode:

Time Stretch: Use this setting to change the tempo while preserving the pitch. Lower percentages will slow down the tempo, while higher ones will increase the tempo. The pitch remains the same throughout.

Pitch Shift: Use this setting to change the pitch while preserving the tempo. 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.

Resample: Use this setting when you do not need to preserve the pitch or the tempo. 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.

Pitch and Time Settings:

Splicing Frequency: This setting determines the size of a 'chunk' of audio data. Splicing frequencies will create 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.

Overlapping: This 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.

Choose appropriate defaults: Check this box if you want *Cool Edit* to select the default Pitch and Time Settings for you.

This function supports Presets.

Compact disc player

You can control a your computer's CD player with a *Cool Edit*'s set of control buttons. To enable these buttons, choose **/View/CD Player**. The controls appear at the bottom main window, below the VU meters and the waveform play/record buttons. You can name your CD and the individual songs as well (for display the next time you use the same CD) by typing the name(s) into the box to the right of the CD Player buttons.

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), the title reflects the title of the current song. As you enter song titles, you can 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 64kb in size, the next file is used for storing new data. This keeps the size of each INI file below 64kb, 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, you can enter thousands and thousands of CDs completely without fear of losing your favorite tracks.

[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: Marks the location currently being played.

[Goto Mark]
Arrow to Red X: Goes to the location that was marked earlier.

[Eject]
CD Ejecting: Spits 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.

View Info (wave information)

Use **/View/Info** to see the fields stored in the header of the file currently being edited. *Cool Edit* can store text information in Windows .WAV files in addition to the audio data itself. Most other wave editors should preserve some or all of the fields you see here. If the "Fill * Fields Automatically" box is checked, 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 its 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, because it will be displayed in OLE objects and the like.

Original Artist: The one who created the sound initially. Examples: Beatles, Patrick Henry, Fred Flintstone.

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

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

Key Words: In the future, sounds may be searched for by key words. Separate key words with a semicolon followed by a space. Example: Violin; Hayden; Johann Strauss.

Digitization Source: Where was the sound digitized from? A tape deck, CD, DAT drive, or maybe directly from a microphone? You may want to describe the board used here too, like SoundBlaster, or DWA.

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 it. Separate names with a semicolon and a space. When a new person edits the file, he or she can add his or her name to the list. Examples: John Cravitz; Fred Millstone.

Digitizer: Who is the technician that did the actual digitizing? Put that person's name right here.

Source Supplier: The name of the person or organization that supplied the original source material. 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

Comments: This is for making any comments you want. Feel free to include any special effects or enhancements you made to any pre-existing waves so that the editing history can be tracked. Please do not use any line returns. End each sentence with a period. Example: It took 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. 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.

Bitmap: Any DIB or BMP file can be inserted, but a 32 X 32 16-color is 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.

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, as they are easier to display by other OLE aware applications. OLE applications may represent your wave file using the bitmap you select.

Cut

Cool Edit 96 can use two different clipboards: the standard Windows clipboard, or its own internal one. The internal clipboard is faster and can handle larger copy and paste operations, but it cannot copy to or paste from other applications, like the Windows clipboard. When you choose **/Edit/Cut**, *Cool Edit* copies the data to its own internal clipboard. If you want to copy data from *Cool Edit* to another application, you should use **/Edit/Windows Clipboard/Copy** to make sure you are using the standard Windows clipboard.

Use Ctrl+X or Shift+Delete to quickly cut the selected wave.

Envelope

Select **/Transform/Amplitude/Envelope** to access the Envelope feature, which gives you fine control over which parts of your wave are amplified and by how much. The top of the graph represents 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.

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

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

This function supports Presets.

Generating Silence

Use **/Generate/Silence** and enter the number of seconds of silence desired to generate that amount of silence at the insertion point.

Pretty simple, eh?

Silence

Choosing **/Transform/Silence** to silence out (that is, remove all audio from) the selection.

Just about as simple as Generate Silence, huh?

Generating DTMF Signals

/Generate/DTMF Signals 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 want 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 defined length.

Tone Time: All tones will last for as long as 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 this character is typed in the Dial String, it will be interpreted as a pause, and silence will be inserted for the duration of the Pause Time.

DTMF Signals: Select this to generate DTMF (normal push-button telephone type) signals using combinations of the standard frequencies 697Hz, 770Hz, 852Hz, 941Hz and 1209Hz, 1336Hz, 1477Hz, 1633Hz.

MF Signals (CCITT R1): Select this to generate MF (internal to telephone networks) signals using paired combinations of the standard frequencies 700Hz, 900Hz, 1100Hz, 1300Hz, 1500Hz, and 1700Hz.

Custom: Select this option to specify custom frequencies for each pair. Click on Reset to DTMF to reset all custom frequencies.

Amplitude: This sets 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 entered!

Generating Tones

With the Tones feature (**Generate** menu), *Cool Edit* can generate a pure sine, square, sawtooth, or other wave type at any frequency supported by the current sample rate (divide the sample rate of the file by two to determine the highest frequency supported). It can also generate harmonics for the base frequency, modulate the tone by a specified frequency, and automatically transition from one frequency to another. Generating tones is a great way to provide a base sound to create spectacular special sound effects.

To generate tones based on one constant set of parameters, check "Lock to these settings only". To generate tones that transition from one set of parameters to another, un-check that box and specify the parameters you want in the Initial and Final Settings tabs.

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

Modulate By: Enter the variation in frequency you want 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.

Frequency Components and Multipliers
You can choose up to 5 overtones, mix them at any proportion, and change the multiplier used to gain the overtone's frequency. If Lock is not checked, all of these change over the duration of the signal from initial to final settings, creating a "swoop" effect. 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 of this slider's frequency with respect to the others. You can generate lots of really great effects with just these 5 overtones. Just experiment and have fun!

dB Volume: Use these sliders to select the intensity (amplitude) of your tones in each channel.

Phasing:

Start Phase: This is the starting location in the "sine wave" cycle that will be produced. If you start at 0 degrees phase, sine waves will start at the baseline. If you start at 90 degrees phase, the sine wave will start at full amplitude (making a noticeable click as well). If you are working in great detail with sine waves (or other types of waves) and need to have the phase "just so" then this option allows you to adjust that.

Phase Difference: 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.

Change Rate: The stereo phase difference can be changed dynamically over time at the given rate. For example, if 1 Hz is entered, the phase difference will cycle through 360 degrees each second.

General:

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

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

Source Modulation: If a selection is 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.

This function supports Presets.

Experiment with all the settings for various wild effects.

... and have fun!

Paste

Cool Edit 96 can use two different clipboards: the standard Windows clipboard, or its own internal one. The internal clipboard is faster and can handle larger copy and paste operations, but it cannot copy to or paste from other applications, like the Windows clipboard. When you choose Paste from the Edit menu, *Cool Edit* pastes the data from its internal clipboard if there is audio data there and uses the Windows clipboard only if the internal clipboard is empty. If you want to paste data from another application, you should use **/Edit/Windows Clipboard/Paste** to make sure you are using the standard Windows clipboard.

/Edit/Paste inserts the wave from the internal 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.

Deleting a selection

Once a range is Selected, it can be removed with **/Edit/Delete**. The deleted portion is *not* copied to the clipboard. It is gone forever.

Quick Filter

The 8-band Quick Filter (**/Transform/Filters** menu) allows one to customize to suit most filtering needs. This "equalizer" works pretty much the same as a standard audio equalizer does, except that the bands are not the same as you might 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 (although of course it increases 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: Adjust these to increase or decrease the frequency component specified beneath the bar.

Volume Bars: Use these controls to adjust the final volume after equalizing. Check the **Lock Vol** checkbox to 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 ordinarily will, result in clipping if the volume adjustment is not turned down.

For serious filtering, see Filter.

Copy

Cool Edit 96 can use two different clipboards: the standard Windows clipboard, or its own internal one. The internal clipboard is faster and can handle larger copy and paste operations, but it cannot copy to or paste from other applications, like the Windows clipboard. When you choose **/Edit/Copy**, *Cool Edit* uses its own internal clipboard. If you want to copy data from *Cool Edit* to another application, you should use **/Edit/Windows Clipboard/Copy** to make sure you are using the standard Windows clipboard.

Before copying, the portion that is to be copied must first be selected.

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

Amplify

Select **/Transform/Amplitude/Amplify** to 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. The sample 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...

Use **/Transform/Delay Effects/Echo** to create continuous echoing and reverb effects.   Each successive echo decays in amplitude by the falloff ratio.   To create the effect of a single echo, use the Delay function instead.

Note that the Echo feature supports separate Ratio, Delay, and Initial Echo Volume settings for the left and right channels.   You can create a wide variety of interesting by choosing different settings for the two channels.   To make the echo "bounce" from one channel to the other, check **Echo Bounce**.   To create identical echo effects for both channels, check **Lock Left/Right**.

Falloff Ratio
Each successive echo will be a certain percentage less than the previous one.   Choosing a falloff ratio of zero results in no echo at all, while choosing a ratio of 100 produces an echo that 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 un-highlighted area, stopping at the right-hand edge of the wave visible in the window.   If the window is zoomed in, the echoing will stop before the end of the file, because it will stop at the right side of the portion on screen.   For example, by using this option, a single word 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 want 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

Use **/Transform/Delay Effects/Reverb** create high-quality reverberation effects. This function can reproduce the effects of a specific environment everything from your coat closet to a grand amphitheater. 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 add some reverb, even as little as 300ms, to open up the sound so it sounds like 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 amphitheaters 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% of the total Reverb length works well. But interesting effects can be achieved by using longer attack times with shorter Reverb lengths for very subtle Reverb. Or, you can couple very short attack times 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, you can choose the exact time it takes for the highest frequencies to be completely cut out. Faster Absorption times simulate rooms that are occupied and have furniture and carpeting like night clubs or theaters. 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.

We have built a few presets to get you started. But for best results, experiment with the different parameters and you'll find just the Reverb you are looking for. The "Large Occupied Hall" gives a very nice live theater 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! Use **/Transform/Delay Effects/Flanger** to get to *Cool Edit*'s flanger. 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 a 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 vica 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

To select the entire wave, choose **/Edit/Select Entire Wave** or press Ctrl+A. This selects the entire waveform, from zero to the end of the wave. It makes no difference whether the view is zoomed in or not. Double-clicking on the waveform selects the visible portion of the wave, while choosing **Select Entire Wave** selects the entire waveform independent of the portion being viewed.

Brainwave Synchronizer

With the Brainwave Synchronizer (**/Transform/Special** menu), you can modify stereo files to produce sounds that when listened to with stereo headphones can 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.

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/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 (**Transform** menu) gives you total control over the left and right channels. The default values 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

/Transform/Invert simply inverts the samples, so all positive offsets are negative and all negative offsets are positive. On stereo waveforms, both channels are inverted.

Dynamics (Compressor/Expander/Limiter/Noise Gate)

The Dynamics function (**/Transform** menu) 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 invertable, 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/<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*.

Generating Noise

Use **/Generate/Noise** to create 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.

Scripts

Scripts are similar to Macros. With *Cool Edit*'s Scripts feature (**Options** menu), you can record commonly-used procedures and run them again on other files or even on groups of files. For example, if you commonly need to boost the bass, add reverb, and export files to a new format, you can record these steps in a script and then run the script to perform those steps automatically. Multiple scripts can be kept in one script file, and identified by name.

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. 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). 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. We 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 mar '?' 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 save 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.

Custom Toolbar Settings

You can arrange the toolbar in any order by changing the button ordering in the listing shown in the Custom Toolbar Settings dialog (**Options** menu). You may highlight more than one item at a time and move them up or down the list. Items at the top of the list will appear at the left of the toolbar. Press the *Zap* button (looks like a lightening bolt) to move the highlighted button(s) to the end of the list. Put all your most used functions near the top, and zap all the functions that you never use!

Check **Enable Toolbar Help** to make the help balloons appear when you hold the mouse over a toolbar button.

The File Save icon can be interpreted as a Save As or a Save Now (without any dialog asking for a filename if the file is already named) by making the appropriate choice after **File Save**.

Internal to the COOL.INI file, the toolbar format is slightly different than with previous versions of *Cool Edit*. Your previous settings should have been retained as closely as possible, but because the older version assigned priorities to the buttons (so buttons in the middle did not display until the window was wide enough), new buttons may appear. Just zap the buttons you do not use.

About the author

David Johnston has been working with sound technology since 1991, and has been toying with electronic synthesizers since the 70's. He started programming while in high school on a Commodore PET computer. In 1982, he started his programming career on an Atari 800; he has since written software for Apple]['s, Commodore 64's, Macintoshes, PC-compatible computers, and numerous mainframes and workstations.

David attended Eastern Montana College for two years and then moved to the Seattle area, where in 1991 he acquired a Bachelor's Degree of Science in Computer Science and Engineering from the University of Washington. He worked as a software engineer for several years at Microsoft, both during and after finishing college. In his spare time, he wrote programs for the challenge and fun of it. Two of those applications, *Cool Edit* and *Kaleidoscope*, went on to become some of the most popular shareware programs available.

In 1995, David Johnston and Bob Ellison co-founded Syntrillium Software Corporation to enhance and market *Cool Edit*, *Kaleidoscope*, and other products. David now devotes all his programming energies toward creating the most powerful and easy-to-use software products around, with special emphasis on applications that work with sound. In 1996, he created *Wind Chimes*, which simulates the sound of real wind chimes on the computer. Click here to find out more about Syntrillium's products.

Cool Edit 96 Registration Information

You can register *Cool Edit* by phone, by email or fax, by regular mail, or through the CompuServe ShareWare Registration service. For immediate registration, call Syntrillium directly with your credit card information ready. Within 2 business days of receiving your order, Syntrillium will email your registration number 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 email Syntrillium Software at **sales@syntrillium.com**.

1) **PHONE:** Call Syntrillium with your credit card information ready to register instantly over the phone between the hours of 9:00 and 5:00, Mountain Standard Time:

   1-888-941-7100 (toll-free from the USA or Canada)

   +1-602-941-4327 (outside the USA and Canada)

2) **EMAIL:** Go to **/Help/Instant Registration Form** and fill out the information in the Instant Registration Form dialog. Include your credit card information, click on Copy to Clipboard to copy your completed registration form to the clipboard, and then go to your email application, create a new message to send to sales@syntrillium.com, and use **/Edit/Paste** to paste the registration information to your email message. Note: the Instant Registration Form automatically scrambles your credit card number for increased security.

   See registration form for a copy of the full registration form that you can copy and paste to a text editor or word processor for printing or editing.

3) **FAX:** Go to **/Help/Instant Registration Form** and fill out the information in the Instant Registration Form dialog. Include your credit card information or click on Check/Money Order Enclosed, and then click on Print to print the registration form.

   See registration form for a copy of the full registration form that you can copy and paste to a text editor or word processor for printing or editing.

4) **MAIL:** Use **/Help/Instant Registration Form** or the blank registration form with your credit card information or a money order or check drawn in US dollars from a US bank to:

   Syntrillium Software Corporation
   P.O. Box 62255
   Phoenix, AZ 85082-2255
   USA

5) **CompuServe:** Use GO SWREG, select Register ShareWare, answer the questions that CompuServe asks, search for keyword "Syntrillium", and select "Display Selected Titles".

Click here to view and copy or print the registration form

Note to users in Benelux and Germany: you can also order *Cool Edit* from Syntrillium's agent in the Netherlands, CopyCats Software & Services. **Click here** to view and copy or print the CopyCats registration form.

Cool Edit 96 Registration Form

Please print clearly!

Name: ____
Address: ____
____
____
____
City: _______________ State/Prov: ________ Zip:________
Country: ________________________

Phone #: ____
FAX #: ____
Email address: ____

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.

[ ] **$25** **Lite Version** You will receive a registration number turning 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, Filter, Flange, Noise Reduction, Quick Filter, Reverb, and Generate DTMF Tones.

You can pay with a US check or money order or with an international money order drawn in US dollars. We also accept Visa, Mastercard, Discover, or American Express; just fill in your card number and expiration date below.

Visa #: ________/________/________/________ Exp Date (required): _________
MC #: ________/________/________/________ Exp Date (required): _________
Discover #: ________/________/________/________ Exp Date (required): _________
AmEx #: ________/________________/____________ Exp Date (required): _________

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

Where did you find *Cool Edit*?

Syntrillium's WWW site___ Other WWW site___ (if so, which: ____________________)
CompuServe___ America Online___ Magazine Cover CD___
Friend___ Product bundle___ (if so, which: _____________________)
Other:____

Brand of PC: ______________________________ Operating system: ______________________
CPU: ________ Sound Card: _________________ RAM: _______ Hard Drive size: _______

What do you use *Cool Edit* for? ____

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

Sound Card: ____

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

Syntrillium Software Corporation
P.O. Box 62255
Phoenix, AZ 85082-2255
USA

Phone (sales only): +1-602-941-4327
Toll-free sales: 1-888-941-7100 (US/Canada only)
Fax: +1-602-941-8170
Email: sales@syntrillium.com

URL: http://www.syntrillium.com

Cool Edit 96 Upgrade Form

Please print clearly!

Why upgrade? You may want to upgrade from the Lite or Basic version of *Cool Edit* to the Basic registration level. Currently registered users of *Cool Edit* versions 1.50 and below may receive an upgraded registration number for *Cool Edit 96* free of charge**. If you currently are registered for *Cool Edit* Lite and would like to upgrade to the Basic version, then check the appropriate box below.

**Note: if you are currently using *Cool Edit* 1.51 or above, you do not need a new registration number to use *Cool Edit 96*.

Name: ____
Address: ____
____
____
____
City: _______________ State/Prov: ________ Zip:________
Country: ________________________

Phone #: ____
FAX #: ____
Email address: ____

Visa #: ________/________/________/________ Exp Date (required): _________
MC #: ________/________/________/________ Exp Date (required): _________
Discover #: ________/________/________/________ Exp Date (required): _________
AmEx #: ________/________________/____________ Exp Date (required): _________

Choose the upgrade you would like

[ ] $0 **Updated Registration Number:** You will receive an updated registration number for use with *Cool Edit 96*.

[ ] $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.

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

Syntrillium Software Corporation
P.O. Box 62255
Phoenix, AZ 85082-2255
USA

Phone (sales only): +1-602-941-4327
Toll-free sales: 1-888-941-7100 (US/Canada only)
Fax: +1-602-941-8170
Email: sales@syntrillium.com
URL: http://www.syntrillium.com

New wave

Use **/File/New** to create a 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. To reproduce frequencies up to 10Khz, a sample rate of at least 20Khz must be chosen. You may enterany sample rate directly, or choose a common sample rate 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 a 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. Although you can create and edit those files, your system may not be capable of playing them properly.

Adjust Sample Rate

Select **/Edit/Adjust Sample Rate** to change how *Cool Edit* interprets the actual waveform data according to the parameters specified below.

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, you maust choose a sample rate of at least 20Khz.   Choose **Custom** to enter an unlisted sample rate.

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

Resolution
This describes the number of bits to use for each sample on each channel.   Choosing an 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 a 16-bit resolution will provide 65,536 unique "volumes", for a much higher signal-to-noise ratio.   Much quieter sounds can be reproduced at a 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.

NOTES:

1) 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.   Although you can create and edit those files, your system may not be capable of playing them properly.

2) Adjust Sample Rate allows *only* changing of Sample Rate.   To change the interpretation of the data as mono/stereo or a certain bit resolution, re-open the waveform as Raw PCM and choose the appropriate interpretation.

Mix Paste

Waves from the clipboard can be looped or mixed with the current wave via **/Edit/Mix Paste**. 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 **Mix Paste** 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.

Crossfade (*n* milliseconds)
Use this option for smoother pasting. When crossfading is enabled, *Cool Edit* fades in the first *n*-milliseconds and fades out the last *n*-milliseconds of pasted data.

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 you can paste the data from the file by using this option.

3D Echo Chamber

The 3D Echo Chamber (**/Transform/Delay Effects** menu) calculates 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 25,000 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, use about 30% for 100 echoes, 5% for 1000 echoes, etc.

Echoes
This is the number of actual echoes to produce. To get a nice Reverb and ambiance effect, at least 300 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 100 echoes or so to test the chamber size and general room sound, then increase that dramatically for the final production. Up to 25,000 echoes can be generated, perhaps even more depending on the size of the room and size of memory.

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.

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. 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

Cool Edit's Distortion feature (**/Transform/Special** menu) can produce the familiar sounds of a cranked electric guitar or even effects such as a blown car speaker, muffled microphone, or an overdriven amp. Use this function to map any sample value to any new sample value.

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

The distortion function can also be four-quadrant, so separate distortion graphs can be given for positive and negative waveform cycles. To use four-quadrant distortion, un-check Symmetric and set specific values for the Positive and Negative distortion.

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

/Transform/Normalize amplifies the highlighted selection to within the specified percentage of the maximum (or optimum) level.   You can also specify a DC Bias or adjust for a DC Bias offset coming from your sound card or recording environment.   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, press the F3 key to automatically run Normalize again on another waveform for very fast normalization of waves.

Converting sample types

Use **/Edit/Convert Sample Type** to change your waveform from one sample type (such as 44KHz/16-bit/stereo) to another. 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. You can select the level of quality you want and specify general volume levels when converting between mono and stereo formats. Higher quality settings take longer to process, but at the highest setting the resultant waveform is identical to having sampled the material at the new rate to begin with.

High quality settings should be used for greater downsampling ratios. When upsampling, the Low quality setting sounds nearly the same as the high quality setting. The difference lies in a larger phase shift in the higher frequencies, but since the phase shift is completely linear, it is very difficult to notice. Downsampling at even the lowest quality setting will not have any undesired noisy artifacts. Instead, it may just sound a little more muffled because of more high end filtering.

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 fal ling, or static. Whether or not dithering is used depends on the audio being converted, and your preferences.

Quality settings - Higher values retain more high frequencies while still preventing aliasing of higher frequencies to lower ones. Since the filter slope is much steeper with higher quality settings, the chance of ringing in these high frequencies is greater (frequencies just below the Nyquist may be boosted abnormally high). With settings too low, many high frequencies may be attenuated too much, leading to muffled sounding audio. Usually values between about 100 and 400 do a great job for most conversion needs.

Pre/Post Filter - To prevent any chance of aliasing, the pre-filter on downsampling, or post-filter on upsampling will remove all frequencies above the Nyquist thus keeping them from aliasing to lower audible frequencies. In general, for best results this option should be enabled.

Dither Amount - When converting to a lower bit resolution, adding some dither (random noise) at a very low level will actually improve the perceived quality of the audio. Quieter signals can be *encoded* in the random noise, allowing sounds that would have been too soft to be heard still audible after conversion. Generally, about 0.2 to 0.7 bits of dither give the best results without adding too much noise. If dithering is disabled, *Cool Edit* simply truncates the data, which can give a *crackly* effect that fades in and out on very quiet audio. With about 0.2 bits of dithering or more, a soft constant hiss is heard in the background instead.

This function supports Presets.

Noise Reduction

Cool Edit's Noise Reduction feature (**/Transform** menu) can remove background noise and general broad band noise 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).

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.

See How to use Noise Reduction for quick help on using this feature, or read below for more detailed information.

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, and then press **Get Noise Profile from Selection**. *Cool Edit* then gathers the statistical information about the background noise, and you are set to remove all noise of this type from your waveform.

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 you reach a balance 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 the original signal.

Distortion effects may manifest themselves as a "hollow" or "underwater/burbley" 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. Adjust **Smoothing Amount** and **Transition Width** up or down to minimize these artifacts.

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 its 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. Because this is so much quieter than the original noise though, it can be quite acceptable.

You can generate great effects 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 **Get Noise Profile from Selection** is chosen will be removed when the reduction level is set to 100.

Load Profile
Click on Load Profile to loads any previously saved noise profile. You can load any *.fft file that *Cool Edit* has saved. 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 44KHz, 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.

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.

Number of Statistical Snapshots in Profile
This number describes how many snapshots of noise to take in the highlighted interval when Get Noise Profile From Selection 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. 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 Get Noise Profile From Selection 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.

FFT Size
This setting 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. FFT sizes up to 24,000 can be used, but tend to introduce reverberation. Sizes around 8192 seem to work best.

Precision Factor
This setting 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 that 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. Beyond values of about 10, there is no noticeable change in quality-- just the time it takes to compute. We suggest using 5 or 7 (odd numbers are best for properties of symmetry).

Special Notes
Noise reduction works best on 16-bit samples, although it also works perfectly well on 8-bit samples. Because of the nature of 8-bit audio, however, it is impossible to reduce the noise level more 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 and 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. Because 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 we 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, **Mix Paste** (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). We 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. We 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 Mix Paste) 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/Brainwave Synchronizer**, 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 use **/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/Brainwave Synchronizer** 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 awaken 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 to help in relaxation. 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/Brainwave Synchronizer** 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
		(Graph should look like an upside-down "V")		
5 to 12Hz	Awake	12	5	40
		(Graph should go from left=lowest to right=highest)		

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. 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 we 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 Syntrillium Software Corporation and 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.

References on brainwave synchronization

Adams, H. B. (1965). A case utilizing sensory deprivation procedures. In L. P. Ullman & L. Krasner (Eds.), *Case Studies in Behavior Modification.* New York: Holt, Rinehart & Winston.

Adrian, E. D. & Yamagiwa, K. (1935). "The origin of the Berger Rhythm." *Brain*, 58, 323-351.

Atwater, F. H. (1988). "The Monroe Institute's Hemisync process: A Theoretical Perspective." Faber, Va: Monroe Institute.

Bandler, R. (1985). "Using Your Brain--For a Change." Moab, UT: Real People Press.

Barber, T. X. (1957). "Experiments in hypnosis." *Scientific American*, 196, 54-61.

Bremer, F. (1958a). "Physiology of the corpus callosum." Proceedings of the Association of Research on Nervous Disorders, 36, 424-448.

Bermer, F. (1958b). "Cerebral and cerebellar potentials." *Physiological Review*, 38, 357-388.

Brackopp, G. W. (1984). Review of research on Multi-Modal sensory stimulation with clinical implications and research proposals. Unpublished manuscript--see Hutchison (1986).

Budzynski, T. (1973). "Some applications of biofeedBack-produced twilight states." In D. Shapiro, et al (Eds.), *BiofeedBack and Self-Control*: 1972. Chicago: Aldine-Atherton.

Budzynski, T. H. (1976). "BiofeedBack and the twilight states of consciousness." In G. E. Schwartz and D. Shapiro (Eds.), *Consciousness and Self-Regulation*, Vol. 1, New York: Plenum Press.

Budzynski, T. H. (1977). "Tuning in on the twilight zone." *Psychology Today*, August.

Budzynski, T. H. (1979). "Brain lateralization and biofeedBack." In B. Shapin & T. Coly (Eds.), *Brain/Mind and Parapsychology.* New York: Parapsychology Foundation.

Budzynski, T. H. (1981). "Brain lateralization and rescripting." *Somatics*, 3, 1-10.

Budzynski, T. H. (1986). "Clinical applications of non-drug-induced states." In B. Wolman & M. Ullman (Eds.), *HandBook of States of Consciousness.* New York: Van Nostrand-Reinhold.

Budzynski, T. H. (1990) "Hemispheric asymmetry and REST." In Suefeld, P. Turner, J. W., Jr. & Fine, T. H. (Eds.), *Restricted Environmental Stimulation*, New York: Springer-Verlag.

Cade, C. M. & Coxhead, N. (1979) "The Awakened Mind: BiofeedBack and the Development of Higher States of Consciousness." New York: Delacorte Press.

Cheek, D. (1976). "Short-term hypnotherapy for fragility using exploration of early life attitudes." *The American Journal of Clinical Hypnosis*, 18, 75-82.

Davidson, R. J., Ekman, P., Saron, C. D., Senulis, J. A., & Friesen, W. V. (1990). "Approach-withdrawal and cerebral asymmetry: Emotional expression and brain physiology." *Journal of Personality and Social Psychology*, 58, 330-341.

Deikman, A. (1969). "De-automatization and the mystic experience." In C. T. Tart (Ed.), *Altered States of Consciousness.* New York: John Wiley & Sons.

Deikman, A. (1971). "Bimodal consciousness." *Archives of General Psychiatry*, 25, 481-489.

Donker, D. N. J., Nijo, L., Storm Van Leeuwen, W. & Wienke, G. (1978). "Interhemispheric relationships of responses to sine wave modulated light in normal subjects and patients." *Electroencephalography and Clinical Neurophysiology*, 44, 479-489.

Evans, F. J., Gustafson, L. A., O'Connell, D. N., Orne, M. T. & Shor, R. E. (1966). "Response during sleep with intervening waking amnesia." *Science*, 152, 666-667.

Evans, F. J., Gustafson, L. A., O'Connell, D. N., Orne, M. T. & Shor, R. E. (1970). "Verbally-induced behavioral response during sleep." *Journal of Nervous and Mental Disease*, 1, 1-26.

Evans, C. & Richardson, P. H. (1988) "Improved recovery and reduced postoperative stay after therapeutic suggestions during gneeral anaesthetic." *Lancet*, 2, 491.

Felipe, A. (1965). "Attitude change during interrupted sleep." Unpublished doctoral dissertation. Yale University.

Foster, D. S. (1990) "EEG and subjective correlates of alpha frequency binaural beats stimulation combined with alpha biofeedBack." Ann Arbor, MI: UMI, Order No. 9025506.

Foulkes, D. & Vogel, G. (1964). "Mental activity at sleep-onset." *Journal of Abnormal Psychology*, 70, 231-243.

Glicksohn, J. (1986). "Photic driving and altered states of consciousness: An exploratory study." *Imagination, Cognition and Personality*, 6, 167-182.

Green, E. E., Green, A. M. (1971). "On the meaning of the transpersonal: Some metaphysical perspectives." *Journal of Transpersonal Psychology*, 3, 27-46.

Green, E. E., & Green, A. M. (1986). "BiofeedBack and States of Consciousness." In B. B. Wolman & M. Ullman (Eds.). *HandBook of States of Consciousness.* New York: Van Nostrand Reinhold.

Harding, G. F. & Dimitrakoudi, M. (1977). "The visual evoked potential in photosensitive epilepsy." In J. E. Desmedt (Ed.), *Visual Evoked Potentials in Man: New Developments*. Oxford: Clarendon.

Henriques, J. B. & Davidson, R. J. (1990). "Regional brain electrical asymmetries discriminate between previously depressed and healthy control subjects." *Journal of Abnormal Psychology*, 99, 22-31.

Hoovey, Z. B., Heinemann, U. & Creutzfeldt, O. D. (1972). "Inter-hemispheric 'synchrony' of alpha waves." *Electroencephalography and Clinical Neurophysiology*, 32, 337-347.

Hutchison, M. (1986). *Megabrain*. New York: Beech Tree Books. William Morrow.

Hutchison, M. (1990). "Special issue on sound/light." *Megabrain Report:* Vol 1, No. 2.

Iamblichus. "The epistle of Porphyry to the Egyptian Anebo." In *Iamblichus on the Mysteries of the Egyptians, Chaldeans, and Assyrians*. Trans. by Taylor, T. London: B. Dobell, and Reeves & Turner, 1895.

Janet, P. (1889). *L'Automatisme Psychologique.* Paris: Alcan.

Koestler, A. (1981). *The Act of Creation*. London: Pan Books.

Kooi, K. A. (1971). *Fundamentals of Electroencephalography.* New York: Harper & Row.

Kubie, L. (1943). "The use of induced hypnagogic reveries in the recovery of repressed amnesic data."

Bull. Menninger Clinic, 7, 172-182.

Lankton, S. R., & Lankton, C. H. (1983). *The Answer Within: A Clinical Framework of Ericksonian Hypnotherapy.* New York: Bruner/Mazel.

Leman, K. & Carlson, R. (1989). *Unlocking the Secrets of Your Childhood Memories.* Nashville: Thomas Nelson.

Lilly, J. C. (1972)). *Programming and Metaprogramming in the Human Biocomputer.* New York: Julian.

Lubar, J. F. (1989). "Electroencephalographic biofeedBack and neurological applications." In J. V. Basmajian (Ed.), *BiofeedBack: Principles and Practice*, New York: Williams & Wilkins.

Mavromatis, A. Hypnagogia: *The Unique State of Consciousness Between Wakefulness and Sleep.* New York: Routledge & Kegan Paul, 1987.

Miller, E. E. (1987). *Software for the Mind: How to program Your Mind for Optimum Health and Performance.* Berkeley, CA: Celestial Arts.

Moscu, K. I. & Vranceanu, M. (1970). "Quelques resultats concernant l'action differentielle des mots affectogenes et nonaffectogenes pendant le somneil naturel." In M. Bertini (Ed.), *Psicofisiologia del Sonno e del Sogno.* Milan: Editrice Vita e Pensiero.

Moses, R. A. (1970). *Adler's Physiology of the Eye: Clinical Applications.* St. Louis: Mosby.

Nemiah, J. C. (1984). *The unconscious and psychopathology.* In S., & Meichenbaum, D. New York: John WIley & Sons, pp. 49-87.

Oster, G. (1973). "Auditory beats in the brain." *Scientific American*, 229, 94-102.

Peniston, E. G. & Kulkowski, P. J. (1989). "Alpha-Theta brainwave training and B-endorphin levels in alcoholics." *Alcoholism*, 13, 271-279.

Richardson, A. & McAndres, F. (1990) "The effects of photic stimulation and private self-consciousness on the complexity of visual imagination imagery." *British Journal of Psychology*, 81, 381-394.

Rossi, E. L. (1986). *The Psychobiology of Mind-Body Healing.* New York: W. W. Norton.

Rubin, F. (1968). (Ed.), *Current Research in Hypnopaedia.* London: MacDonald.

Rubin, F. (1970). "Learning and sleep." *Nature*, 226, 447.

Schacter, D. L. (1977). "EEG theta waves and psychological phenomena: A review and analysis." Psychology, 5, 47-82.

Schultz, J. & Luthe, W. (1959). *Autogenic Training: A Psychophysiological Approach in Psychotherapy.* New York: Grune & Stratton.

Sittenfeld, P., Budzynski, T. & Stoyva, J. (1976). "Differential shaping of EEG Theta rhythms." *BiofeedBack and Self-Regulation*, 1, 31-45.

Stoyva, J. M. (1973), "BiofeedBack techniques and the conditions for hallucinatory activity" In McGulgan, F. J. and Schoonover, R. (Eds), *The Psychophysiology of Thinking.* New York: Academic Press.

Svyandoshch, A. (1968). "The assimilation and memorization of speech during natural sleep." In F.

Rubin (Ed.), *Current Research in Hypnopaedia.* London: MacDonald.

Swedenborg, E. *Rational Psychology.* Philadelphia: Swedenborg Scientific Association, 1950.

Tomarken, A. J., Davidson, R. J., & Henriques, J. B. (1990). "Resting frontal brain asymmetry predicts affective responses to films." *Journal of Personality and Social Psychology*, 59, 791-801.

Townsend, R. E. (1973). "A device for generation and presentation of modulated light stimuli." *Electroencephalography and Clinical Neurophysiology*, 34, 97-99.

Tucker, D. M. (1981). "Lateral brain function, emotion, and conceptualization." *Psychological Bulletin*, 89, 19-46.

Van der Tweel, L. H. & Verduyn Lunel, H. F. E. (1965). "Human visual responses to sinusoidally modulated light." *Electroencephalography and Clinical Neurology*, 18, 587-598.

Van Dusen, W. (1975). *The Presence of Other Worlds.* London: Wildwood House.

Walter, V. J. & Walter, W. G. (1949). "The central effects of rhythmic sensory stimulation." *Electroencephalography and Clinical Neurophysiology*, 1, 57-86.

Wickramasekera, I. E. (1988). *Clinical Behavioral Medicine: Some Concepts and Procedures.* New York: Plenum Press.

Technical bibliography

Embree, Paul M. and Kimble, Bruce (1991). C Language Algorithms for Digital SIgnal Processing. Prentice Hall, Inc. ISBN 0-13-133406-9 (TK5102.5.E46)

Everest, F. Alton (1994). The Master HandBook Of Acoustics (3rd ed). McGraw Hill, Inc. ISBN 0-8306-4437-7 (QC225.15E93)

ICMC Proceedings from 1988 on up

IEEE Transactions on Acoustics, Speech, and Digital Signal Processing

Kuc, Roman (1988). Introduction to Digital Signal Processing. McGraw Hill, Inc. ISBN 0-07-035570-3 (TK5102.5K823)

Mitra, Sanjit K. and Kaiser, James F. (1993). HandBook for Digital Signal Processing. John Wiley & Sons, Inc. ISBN 0-471-61995-7 (TK5102.5.H318)

Morgan, Don (1994). Practical DSP Modeling, Techniques, and Programming in C. John Wiley & Sons, Inc. ISBN 0-471-00434-0 (TK5102.9.M67)

Roads Curtis (1989). The Music Machine (Selected Readings from Computer Music Journal. The MIT Press. ISBN 0-262-18131-2 (ML1093.M88)

Rorabaugh, C. Britton (1993) Digital Filter Designer's HandBook : Featuring C Routines. McGraw-Hill, Inc. ISBN 0-07-911166-1 (TK7872.F5R68)

Vaidyanathan, P. P. (1992) Multirate Systems and Filter Banks. Prentice-Hall, Inc. ISBN 0-13-605718-7 (TK5102.5.V24)

Watkinson, John (1994). The Art of Digital Audio (2nd ed). Focal Press. ISBN 0-240-51320-7

Ziemer, Rodger E. (1989) Signals and Systems: Continuous and Discrete. Macmillan Publishing Company. ISBN 0-02-431630-X (QA402.Z53)

Many uses / effects of using brainwaves

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 brainwaves

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 we have mixed two different (but similar in frequency ranges) brainwave files together, we 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 you use frequencies under 8hz, you may find yourself 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. 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 on brainwaves

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 a testimonial from David Johnston, author of *Cool Edit*: 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: This collection does not include a great deal of proper citing; it is just intended 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...

The Loop Mode Toggle from the Options menu toggles the play button between Loop and Play states. When Loop mode is on, the Play button loops the current selection (or the entire file if nothing is selected) continuously; when it is off, it plays the selection (or file) once and then stops.

Level meters (VU meters)

Select **/Options/Monitor VU Level** to start monitoring the recording source. This feature can be 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, or the equivalent for Windows 95.

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 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 to 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. If the option to Adjust for DC offset is enabled, false clip reading may occur since the baseline is being adjusted. Disable the DC offset adjustment to have the clip indicators only light up when absolute clipping occurs.

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.

Show on Play and Record
Enables the meters while playing and recording waves in *Cool Edit*.

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. Note: The clip indicators will always light if clipping occurs, but if Adjust for DC is enabled, the indicators may light up when the audio has a DC offset.

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 higer 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 **Mix Paste** 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 and/or the number of buffers 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 unexpectedely 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.

License Agreement

IMPORTANT - READ CAREFULLY: This Syntrillium End-user License Agreement ("EULA") is a legal agreement between you (either an individual or a single entity) and Syntrillium Software Corporation (SYNTRILLIUM) for one (1) license to use *Cool Edit*. *Cool Edit* includes computer software and associated media and printed materials, and may include "online" or electronic documentation ("SOFTWARE PRODUCT" or "SOFTWARE"). By installing, copying, or otherwise using the SOFTWARE PRODUCT, you agree to be bound by the terms of this EULA. If you do not agree to the terms of this EULA, promptly return the unused SOFTWARE PRODUCT to the place from which you obtained it for a full refund.

SOFTWARE PRODUCT LICENSE

The SOFTWARE PRODUCT is protected by copyright laws and international copyright treaties, as well as other intellectual property laws and treaties. The SOFTWARE PRODUCT is licensed, not sold.

1. GRANT OF LICENSE. This EULA grants you the following rights:

- Software. You may install and use one copy of the SOFTWARE PRODUCT, or in its place, any prior version for the same operating system, on a single computer. The primary user of the computer on which the SOFTWARE PRODUCT is installed may make a second copy for his or her exclusive use on either a home or portable computer.

- Storage/Network Use. You may also store or install a copy of the SOFTWARE PRODUCT on a storage device, such as a network server, used only to install or run the SOFTWARE PRODUCT on your other computers over an internal network. However, you must acquire and dedicate a license for each separate computer on which the SOFTWARE PRODUCT is installed or run from the storage device. A license for the SOFTWARE PRODUCT may not be shared or used concurrently on different computers.

- Multi-User and Site License. If you have acquired a multi-user or site license for the SOFTWARE, you may make the number of additional copies of the computer software portion of the SOFTWARE PRODUCT authorized in your license agreement with SYNTRILLIUM, and you may use each copy in the manner specified above. You are also entitled to make a corresponding number of secondary copies for home or portable computer use as specified above.

2. DESCRIPTION OF OTHER RIGHTS AND LIMITATIONS.

- Limitations on Reverse Engineering, Decompilation, and Disassembly. You may not reverse engineer, decompile, or disassemble the SOFTWARE PRODUCT, except and only to the extent that such activity is expressly permitted by applicable law notwithstanding this limitation.

- Separation of Components. The SOFTWARE PRODUCT is licensed as a single product. Its component parts may not be separated for use on more than one computer.

- Rental. You may not rent or lease the SOFTWARE PRODUCT.

- Software Transfer. You may permanently transfer all of your rights under this EULA, provided you retain no copies, you transfer all of the SOFTWARE PRODUCT (including all component parts, the media and printed materials, any upgrades, this EULA and, if applicable, the Certificate of Authenticity), and the recipient agrees to the terms of this EULA. If the SOFTWARE PRODUCT is an upgrade. any transfer must include all prior versions of the SOFTWARE PRODUCT.

- Termination. Without prejudice to any other rights, SYNTRILLIUM may terminate this EULA if you fail to comply with the terms and conditions of this EULA. In such event, you must destroy all copies of the SOFTWARE PRODUCT and all of its component parts.

3. UPGRADES. If the SOFTWARE PRODUCT is an upgrade from another product, whether from SYNTRILLIUM or another supplier, you may use or transfer the SOFTWARE PRODUCT only in conjunction with that upgraded product, unless you destroy the upgraded product. If the SOFTWARE PRODUCT is an upgrade of a SYNTRILLIUM product, you now may use that upgraded product only in accordance with this EULA. If the SOFTWARE PRODUCT is an upgrade of a component of a package of software programs that you licensed as a single product, the SOFTWARE PRODUCT may be used and transferred only as part of that single product package and may not be separated for use on more than one computer.

4. COPYRIGHT. All title and copyrights in and to the SOFTWARE PRODUCT (including but not limited to any images, photographs, animations, video, audio, music, text, and "applets" incorporated into the SOFTWARE PRODUCT), the accompanying printed materials, and any copies of the SOFTWARE PRODUCT are owned by SYNTRILLIUM or its suppliers. The SOFTWARE PRODUCT is protected by copyright laws and international treaty provisions. Therefore, you must treat the SOFTWARE PRODUCT like any other copyrighted material except that you may either (a) make one copy of the SOFTWARE PRODUCT solely for backup or archival purposes or (b) install the SOFTWARE PRODUCT on a single computer provided you keep the original solely for backup or archival purposes. You may not copy the printed materials accompanying the SOFTWARE PRODUCT.

5. DUAL-MEDIA SOFTWARE. You may receive the SOFTWARE PRODUCT in more than one medium. Regardless of the type or size of medium you receive, you may use only one medium that is appropriate for your single computer. You may not use or install the other medium on another computer. You may not loan, rent, lease, or otherwise transfer the other medium to another user, except as part of the permanent transfer (as provided above) of the SOFTWARE PRODUCT.

6. US GOVERNMENT RESTRICTED RIGHTS. The SOFTWARE PRODUCT and documentation are provided with RESTRICTED RIGHTS. Use, duplication. or disclosure by the Government is subject to restrictions as set forth in subparagraph (e)(I)(ii) of the Rights in Technical Data and Computer Software clause at DFARS 252.227-7013 or subparagraphs (c)(1) and (2) of the Commercial Computer SoftwareRestricted Rights at 48 CFR 52.227-19, as applicable. Manufacturer is Syntrillium Software Corporation/P.O. Box 62255/Phoenix, AZ 85082-2255.

MISCELLANEOUS

This EULA is governed by the laws of the State of Arizona. If this product was acquired outside the United States, then local law may apply.

Should you have any questions concerning this EULA, or if you desire to contact SYNTRILLIUM for any reason, please contact the SYNTRILLIUM subsidiary serving your country or write: Syntrillium Software, P.O. Box 62255, Phoenix, AZ 85082-2255.

LIMITED WARRANTY

LIMITED WARRANTY. SYNTRILLIUM warrants that (1) the SOFTWARE PRODUCT will perform substantially in accordance with the accompanying written materials for a period of ninety (90) days from the date of receipt, and (2) any hardware accompanying the SOFTWARE PRODUCT will be free from defects in materials under normal use and service for a period of one (1) year from the date of receipt. Some states and jurisdictions do not allow limitations on duration of an implied warranty, so the above limitation may not apply to you. To the extent allowed by applicable law, implied warranties on the SOFTWARE PRODUCT and hardware, if any, are limited to ninety (90) days and one year, respectively.

CUSTOMER REMEDIES. SYNTRILLIUM's and its suppliers' entire liability and your exclusive remedy shall be, at SYNTRILLIUM's option, either (a) return of the price paid, or (b) repair or replacement of the SOFTWARE PRODUCT or hardware that does not meet SYNTRILLIUM's Limited Warranty, and which is returned to SYNTRILLIUM with a copy of your receipt. This Limited Warranty is void if failure of the SOFTWARE PRODUCT or hardware has resulted from accident, abuse, or misapplication. Any replacement SOFTWARE PRODUCT or hardware will be warranted for the remainder of the original warranty period or thirty (30) days, whichever is longer. Outside the United States, neither these remedies nor any product support services offered by SYNTRILLIUM are available without proof of purchase from an authorized international source.

NO OTHER WARRANTIES. To the maximum extent permitted by applicable law, SYNTRILLIUM and its suppliers disclaim all other warranties, either express or implied, including, but not limited to, implied warranties of merchantability and fitness for a particular purpose, with regard to the SOFTWARE PRODUCT, and any accompanying hardware. this limited warranty gives you specific legal rights. You may have others, which vary from state/jurisdiction to state/jurisdiction.

NO LIABILITY FOR CONSEQUENTIAL DAMAGES. To the maximum extent permitted by applicable law, in no event shall SYNTRILLIUM or its suppliers be liable for any special, incidental, indirect, or consequential damages whatsoever (including, without limitation, damages for loss of business profits, business interruption, loss of business information, or any other pecuniary loss) arising out of the use of or inability to use the software product, even if SYNTRILLIUM has been advised of the possibility of such damages. Because some states and jurisdictions do not allow the exclusion or limitation of liability for consequential or incidental damages, the above limitation may not apply to you.

Cool Edit 96 Registration Form for Benelux and Germany

Users in Benelux and Germany who would like to pay in their own currency can order *Cool Edit* from our local agent, CopyCats Software & Services. To do so, fill in this form and send it, along with your payment, to:

CopyCats Software & Services
Postbus 1088
1700 BB Heerhugowaard
Nederland/Niederlande
Tel/Fax: +31 (0)72-5745993
Email 100121.1562@compuserve.com

Please print clearly!

Name: Mr / Mrs____
Company: ____
Address: ____
Postcode: _____________ City: ____________________________
Country: ____________________
Phone: ____________________ Fax:______________________
Email,CIS: ____
VAT ID#: BE / DE |_|_|_|_|_|_|_|_|_|

Note: Belgian and German businesses should fill in their VAT ID# (BTW-nummer / Umsatzsteuer-Identifikationsnummer). If you do, CopyCats will not charge VAT and your order will be 17.5% cheaper (please divide order total by 1.175).

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.

[ ] $25 **Lite Version** You will receive a registration number turning 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, Filter, Flange, Noise Reduction, Quick Filter, Reverb, and Generate DTMF Tones.

Prices (fully inclusive):

Basic:	Dfl. 99,50	DM 89,95	1850 BEF
Lite:	Dfl 49,50	DM 45,00	925 BEF

Please select your method of payment:

[ ] (NL,B,D) Enclosed check, eurocheque, girobetaalkaart, Zahlkarte
[ ] (NL) Overmaking naar Postbank 43.28.577
[ ] (B) Overmaking naar Postcheque 000.1656064.80
[ ] (D) Überweisung auf Postbank Saarbrücken Kto.Nr. 556 33-666 BLZ. 590 100 66
[ ] (NL,B,D) Op rekening / Auf Rechnung (30 day nett / businesses and government orders only)

Where did you find *Cool Edit*? ____

What do you use *Cool Edit* for? ____

Dank u voor uw registratie! - Vielen Dank für Ihre Registration!

What's new in *Cool Edit 96*?

If you've seen *Cool Edit* before, you may be wondering, "So what's the big deal about this new version?" Well, in a few words, we've added more *Speed*, *Power*, and *Features*. In a few more words…

Blinding speed!

Faster file-loading: the first time you load a file, *Cool Edit 96* creates a peak file so that file operations and subsequent loading and saving are **much** faster.

Faster Spectral View: no more waiting for the screen to draw.

Faster re-draw: remember how you used to have to wait for the waveform to redraw when you edited a large file? *Cool Edit 96* redraws waveforms almost instantly!

Higher Quality - We have boosted the quality of practically every function to nearly that of using 24-bit samples, while still editing 16-bit audio. This means that *Cool Edit 96* keeps more audio data after processing for the cleanest results.

Zoom aaaaallll the way in!

Now you can zoom right down the individual sample! You can also click and drag on individual samples to tweak your waveform to your heart's content.

New Vertical Zoom - can't tell the peaks from the valleys? Now you can zoom in **vertically** to see them better. Use Ctrl+[up] and Ctrl+[down].

More format support!

RealAudio® (export only) - create your files in *Cool Edit 96* and save directly to RealAudio (.ra) format! See http://www.realaudio.com for more information on creating audio for your web site with RealAudio.

DiamondWare Digitized Format (import/export) - read and write the DWD file format used by DiamondWare 's Sound ToolKit, a programmer's function-call library for interactive audio. See http://www.dw.com for more information on DiamondWare's products.

Click-and-drag rulers - you can click on and drag both the vertical and horizontal rulers (right side and bottom, respectively) to see just the area you want to edit. You can also right-click on either ruler to adjust the Zoom level and other settings.

New high-precision Stretch - change pitch or tempo with near-perfect results!

Integration with Cakewalk® Pro Audio™ - edit your wave files with *Cool Edit* without ever leaving *Cakewalk!*

Cleaner Noise Reduction - less noise and fewer artifacts.

New automatic crossfading on edits eliminates clicks at the boundaries when, for example, filtering portions of a waveform.

New Statistics function - get information on peak amplitude, RMS power levels, and more!

Separate controls for left and right channels in Echo function for really cool effects.

Four-quadrant Distortion for more control on distortion qualities.

New internal clipboard - now you can make cuts and pastes of any size, without suffering from the limitations of the Windows clipboard. (You can also still use the Windows clipboard if you need to transfer audio data between *Cool Edit* and other applications.)

Continuous <u>Frequency Analysis</u> - when you set the FFT size to 1024 or lower, the frequency analysis window updates while you play your file.

Higher-quality sample rate conversion and noise shaping.

"Live" updating during recording when in waveform view - watch your waveform grow as you record!

More customizability - change nearly any color setting, select primary and alternate temporary file locations, and much more!

New Menu structure - now it's easier to find the features you want.

Most-recently-used file list - load the files you've been working on without having to search for them.

Setup and Uninstall - now it's easier than ever to install and remove *Cool Edit*.

Statistics

The Statistics dialog (**Analyze** menu) gives you the following details about the current waveform. You can jump directly to the exact location in the waveform where a minimum or maximum was found by clicking on appropriate arrow in the dialog.

Minimum/Maximum Sample Value gives the the maximum and minimum sample values in the range. Pressing the [>] button associated with the value will place the cursor at that location.

Peak Amplitude is the absolute maximum sample value given in decibel form. Pressing [>] will jump to the peak amplitude in the waveform.

Possibly Clipped Samples is the count of samples that were at -32768 or 32767 (for 16-bit) which could possible represent clipped samples. Pressing [>] will jump to the first such sample.

DC Offset is the measure of the DC (Direct Current or center of the waveform). Positive values are above the center line (zero volts) and negative values are below.

Minimum/Maximum RMS Power: the Root Mean Squared (closer to what the ear hears) amplitude of the waveform is scanned using a window of the size given below as the RMS Window Width. The [>] buttons will jump to the average quietest and loudest sections of the waveform.

Average RMS Power represents the average power of the entire selection. This is a good measure of the overall loudness of the waveform selection.

RMS Window Width: when calculating the RMS minimum and maximum values, a sense of locality is needed. So the RMS is taken over a window of this size. This means that for the minimum RMS power, for example, is the minimum for any *n*-millisecond window (where *n* is the value entered).

View menu

The View menu displays options relating to changing the display or viewable area.   Click on one of the following items to get Help on that function.

Waveform View
Spectral View
Left Channel
Right Channel
Viewing Range
Display Time Format
Vertical Scale Format
Info...
Cue List
Play List
CD Player

Analyze menu

The Analyze menu contains functions that give you information about the file or selection being viewed.
Click on one of the following items to get Help on that function.

Frequency Analysis
Statistics*

*NOTE: This function is not available in the Lite version.

Help menu

Cool Edit's Help file is the best source of information and advice on all of *Cool Edit*'s features. The Help menu gives you following options:

Contents: Displays the Contents topic.

Quick Reference: Brings up a Search dialog for locating topics in the Help file.

Features: Displays a topic that provides general descriptions of *Cool Edit*'s features and capabilities.

How to register: Displays information on how to register your copy of *Cool Edit*.

Registration Form: Displays information on how to register *Cool Edit* and how to get to the registration form itself.

Instant Registration Form: Brings up *Cool Edit*'s Instant Registration Form dialog, which you can use to print your registration form or to copy it to the clipboard for emailing.

About…: Displays the *Cool Edit* "About" box.

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.

Use this option to copy audio data to or paste it from the standard Windows clipboard. Data on the Windows clipboard is available to other Windows applications, while data on *Cool Edit*'s internal clipboard is not. However, *Cool Edit*'s internal clipboard does not have the memory limitations of the Windows clipboard. See Copy for more information on the difference between the internal clipboard and the Windows clipboard.

Zero Cross Adjust (F4)

The Zero Cross Adjust function (**Edit** menu) moves the begin and end points of the current selection to the nearest places where the waveform crosses the center line (zero amplitude). Zero cross adjustment is important because if you delete or insert a portion of a wave without lining up the amplitudes at the interface between the deleted or inserted portion, the result is often an audible pop or click at that point. Normally, you should use this feature whenever you want to delete a portion of a wave or insert a portion in the middle of a wave. The process is as follows: first select the portion you want to delete (or position the cursor where you want to insert) and then choose **/Edit/Zero Cross Adjust** (or press F4) to adjust the selection.

Recording and playing files with *Cool Edit*

Recording with *Cool Edit* is easy. You can record from a microphone, your computer's CD player, a MIDI source, or any signal you can plug into the microphone or "Line in" ports on the back of your sound card.

To start recording, simply use **/File/New** to open a new file, select the sample rate, bit resolution, and number of channels (stereo or mono) that you want to use, press OK, and click on the Record button in the lower left area of the main window to begin. When you are done recording, click on Stop. To save your recording, see Saving.

Note that you may need to adjust the recording level for your input signal.

Converting files with *Cool Edit*

To convert a single file from one format to another, use **/File/Open** to load the file, and then select **/File/Save As** to save the file, and specify your new target format in the **Save as Type** field.   See Batch Conversion for information on converting multiple files automatically.

Most recently-used file list (File menu)

At the bottom of the File menu, *Cool Edit* lists the eight files you most recently opened. You can quickly open any of these files again by selecting it in the File menu with the mouse or by typing the number associated with the file you want.

How to merge two mono files to a stereo file

In *Cool Edit*, open the first mono file. Select the whole file and use **/Edit/Copy** to put it on the clipboard. Then create a new, stereo file in *Cool Edit* (the file you want to create with both mono files). Select **/View/Left** (or **Right**) **Channel** so that you're editing only one channel, and choose **/Edit/Paste** to paste the mono data into that channel. Choose **/File/New Instance** to open a new *Cool Edit* window, open the other mono file, and copy it to the clipboard. Close that file and then select **/View/Left** (or **Right**) **Channel** again and de-select the other one to edit only the other channel. Paste the file in, and you're done.

How to use Noise Reduction

To quickly reduce noise from a file, select at least one half second of the file (preferably one with ONLY the noise and no foreground audio), and select **/Transform/Noise Reduction**. Click on Get Noise Profile From Selection to sample the noise. Then click on Close, select the entire file (or the portion you want to noise-reduce), go to **/Transform/Noise Reduction** again, and click on OK to perform the noise reduction.

See Noise Reduction for more detailed information on using Noise Reduction.

How to record from a CD

1) Choose **/View/CD Player** to access *Cool Edit*'s CD player controls (you'll see them pop down at the bottom of the main window). Note: You can instead use an external CD player application to control the CD player, but you can only use one such application at a time.

2) Choose **/File/New** to create a new file in *Cool Edit*. Select the sample type, bit resolution, and number of channels (mono or stereo). Note: "CD quality" recording is 44100 Hz, 16 bit, stereo.

3) Click on the red Record button in *Cool Edit*'s wave controls to start recording.

4) Click on the Play button on *Cool Edit*'s CD player controls (or other CD application) to start the CD playing. You can also click on the track number in *Cool Edit*'s CD controls to play a specific track.

5) When the recording is complete, click on the Stop button in both *Cool Edit*'s wave controls and on the Stop button in *Cool Edit*'s (or other CD application's) CD player controls.

Flush Virtual File

By default, when you open a wave file that that *Cool Edit* can interpret directly without decompressing or converting it (such as a Windows WAV), *Cool Edit* opens the file virtually, meaning no wave data is copied to the temp file; rather the data is used directly from within the original wave file. This feature saves disk space and can be very useful when you are working with very large files or have little free disk space. However, there is a trade-off: when the file is open virtually, it is always "open" in *Cool Edit* and cannot be opened, deleted, or moved by another application until you close it in *Cool Edit*. (Note: multiple instances of *Cool Edit* can virtually open the same file without any such problems.)

To "free" the file for use by another application, select **/File/Flush Virtual File**. This involves just copying the file's audio data to *Cool Edit*'s temporary file so that other applications can open, delete, move, or otherwise gain access to the file.

This is the same reason that Saving back to the original file sometimes incurs a "Flushing Virtual File" progress box. If you have changed the length of the file during your editing, or if other non-audio information was altered for the file (**/View/Info** data, Cue List, Play List, etc.), then the entire file must be re-written back out. If only portions of the audio data have changed, then only those changed portions will be written back, and this automatic flushing will not be done.

Installing and uninstalling *Cool Edit*

If for any reason your installation of *Cool Edit* becomes corrupt, just run the main Setup program again to re-install it.   If you decide you do not want to register *Cool Edit*, or if for some other reason you need to remove it from your system, then we at Syntrillium will be very sorry!   However, we have provided an easy "Uninstall" procedure: just click on the Windows Start button, select Settings - Control Panel - Add/Remove Programs, select *Cool Edit 96* in the list of installed software, and click on the Add/Remove button.

Using *Cool Edit* with *Cakewalk Pro Audio*

When you install *Cool Edit*, it automatically registers itself for use with *Cakewalk® Pro Audio™* from Twelve Tone Systems. This means you can use *Cool Edit* as your waveform editor from within *Cakewalk*. *Cool Edit* shows up as an item in *Cakewalk*'s **Tools** menu; when you select a wave file within a track, *CakeWalk* automatically loads that wave into *Cool Edit* for editing.

Note: you must have *Cakewalk Pro Audio* 6.0 or higher to integrate *Cool Edit* with it.

Using ACM

Cool Edit provides support for Microsoft's ACM (Audio Compression Manager) driver, which enables you to load and save files in a variety of formats other than those supported by *Cool Edit*'s own converters, such as DSP Group TrueSpeech and GSM 6.10. Some of these formats come as a standard part of Windows 95, while you may acquire others when you install other software. To save a file in an alternate format using the ACM driver, use **/File/Save As**, select ACM Waveform as your target format, and click on Options. You can select from among various quality levels, and each level will give you different options for formats and attributes.

Please note that the ACM driver you want to use may require that the file be in a specific format before saving. For example, if you want to save a file in the DSP Group TrueSpeech format, you should first use **/Edit/Convert Sample Type** to convert the file to 8KHz/mono/16bit, because that is the only format supported by the TrueSpeech ACM driver. For more information on any particular ACM driver, contact the creator of the format (such as DSP Group for TrueSpeech, or CCITT for the various CCITT formats) or the manufacturer of the hardware that uses the format in question (such as Creative Labs for the SoundBlaster ACM driver).

Instant Registration Form

Use **/Help/Instant Registration Form** to access *Cool Edit*'s easy-to-use form for ordering your license to use *Cool Edit*. Just fill in all applicable fields (especially your full name and email address), select your payment method, and click on Print to print the form for faxing or mailing or Copy to Clipboard so that you can paste it into an email message (which you can send to **sales@syntrillium.com**).

Please note: we do require your full name for registration purposes; this helps us identify you in case you should lose or forget your registration information at any time.

See the Registration Form for information on the differences between the Lite and Basic registration levels.

Sampler Info

Use **/View/Sampler Info** to add or view information used by other programs, such as synthesizer up/downloading software. See below for descriptions of the fields supported.

Target Manufacture ID and Target Product Code default to 0 for this version - not supported yet by any known applications.

Sample Period is filled in with the sample rate (or within 1Hz of it), and is filled automatically. Change this field if you wish the sampler to think the data is at a different rate than it actually is.

MIDI Unity Note: This is the value at which the sampler will assume the current waveform is to be played. No pitch shifting will be done to the sample if played at this not on a synth. A4 would be a 440Hz tone.

Fine Tune: The actual tone can be entered as precise as 1/100th of a cent. Enter the numer of cents above the Note that the tone actually is.

Find using Analysis: If a sampler loop is selected in the list below, the frequency at the center of that loop will be filled in for the Note and Fine Tune fields. If no loops are selected, the center of the entire waveform will be used to gain the current note. This value can be off by a few hundredths of a cent, so manual adjustment after finding the note may be necessary. For example, you may get G#4 at 99.99 cents, which would probably really be A4 and 0 cents.

SMPTE Offset: The SMPTE Format describes the format of the actual offset in the Offset field. This information may be used by a sampler to play the sample at the specified time offset.

Sampler Loops: New loops can be added by first highlighting the area then going to this dialog and pressing New. If no area is highlighted, New should still be pressed to add a new loop. The actual starting point, ending point or length can also be entered directly by hand in the appropriate fields. Samplers can usually play loops forward, backward, or back and forth and back again. Each loop can be looped a different number of times, or infinite (such as a sustain loop would be, and the infinite loop would exit once the synth key is depressed).

Note: Currently this information is only saved in .wav files.

Vertical Scale Format

Choose **/View/Vertical Scale Format** to change the units used in the vertical scale on the right-hand side of the Waveform View. In Waveform View, you can select from among Sample Values, Normalized Values, and Percentage units.

Note: In Spectral View, the vertical scale is always in Hertz (Hz).

Quick Reference