Version 1.1 User Guide

Overview

Cool Edit Pro is a comprehensive multitrack digital audio editor for Windows 95 and Windows NT. To use it, you need only a PC compatible computer running one of these operating systems, one or more Windows-compatible sound cards, and some imagination.

Designed as a complete audio environment, *Cool Edit Pro* puts all of the functionality needed for taking an audio project from conception to completion in one package, eliminating the need for separate applications, or "plug-ins". From recording to mixdown, you'll find everything to do the job in *Cool Edit Pro*, such as session notes for planning and sketching out your session, and the flexibility of *unlimited multiple sound card support,* for complete freedom in routing your audio. Once you've recorded (…up to 64 tracks!), you can apply *Cool Edit Pro*'s unmatched DSP and editing power. Tweak EQ, and other effects in real-time to get things just right. Arrange your tracks with drag and drop, down-to-the-sample or frame precision; snap-to guides, crossfades, and images will help you get it done. When you're ready to mix, there's separate level, pan, mute, solo, and routing for each track.

While *Cool Edit Pro* does provide an all in one audio solution, it also works with the rest of your audio tools as part of your studio. Support for Microsoft's DirectX/ActiveMovie means you can use DSP modules from leading manufacturers like Waves and QTools from within *Cool Edit Pro.* If you're needs go beyond audio to include working with MIDI or video, you'll find *Cool Edit Pro*'s MIDI/SMPTE synchronization provides seamless integration with these mediums.

Though *Cool Edit Pro* is loaded with features, its layout is such that you're not encumbered by excess windows; you won't need a 20-inch monitor to get the job done. Get at just the features you want. You'll find this under-the-hood architecture throughout *Cool Edit Pro.*

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

Cool Features

Cool Edit Pro's combination of digital recording and editing features includes:

- **Multi-track editing and mixing**—up to 64 tracks!
- **30 distinct effects modules**, including Delay and Echo effects, Noise/Hiss/Click-and-Pop Reduction, Reverberation, Pitch and Tempo adjustment, Graphic and Parametric Equalizers, and more
- **Support for more than 16 distinct file formats** (not including sub-formats)
- **Noise, Tone, and DTMF Tone generation**
- **32-bit sample resolution support and full 32-bit internal processing**
- **"Phrase" recognition (auto cue generation)** (will change somewhat)

- **Cue and Play List** support for multiple files and for segments within files
- **Scientific filters, text export capability, and file-statistics** gathering functions for data analysis
- **Scripting and Batch-Processing** capability
- **Waveform and Spectral View options**
- **Multi-track SMPTE/MTC chase locking synchronization**

With *Cool Edit Pro* and any Windows sound card, you have the power of a complete digital recording studio under your fingertips!

Notes on how to navigate your way around the manual

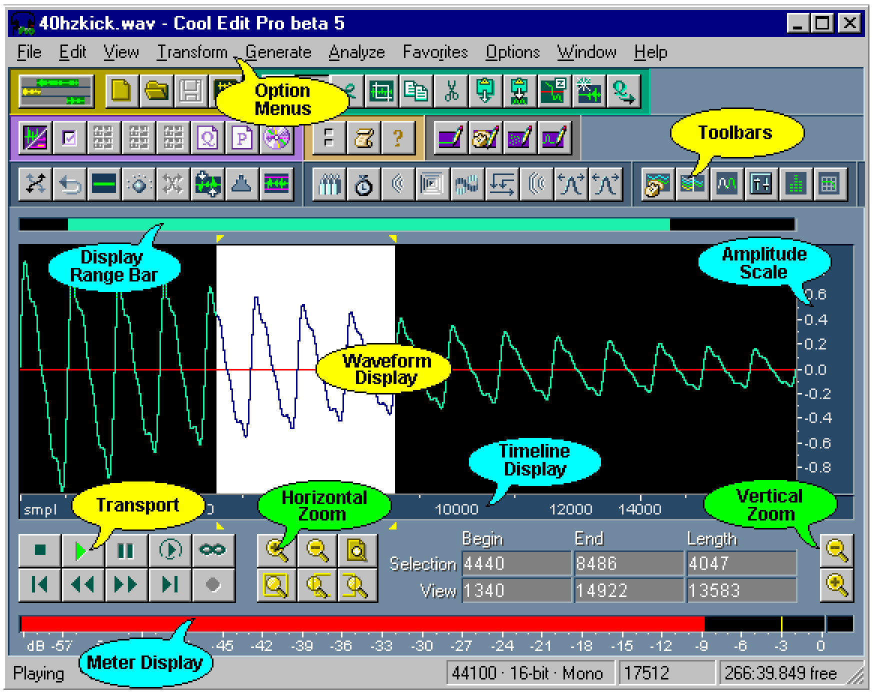
- The icon gives hints on "noteworthy" information.
- The icon alerts you to hints that are just too cool to pass up!
- The icon alerts you to a short simple exercise that can be used to quickly learn about a specific *Cool Edit Pro* option or function.

Installing and uninstalling *Cool Edit Pro*

If for any reason your installation of *Cool Edit Pro* becomes corrupt, simply re-install the program by selecting **Settings - Control Panel - Add/Remove Programs**, choose the **Add/Remove Programs** icon, install the *Cool Edit Pro* CD-ROM into your drive and follow the simple installation instructions. If you need to remove *Cool Edit Pro* from your system, for some reason, 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 Pro* in the list of installed software, and click on the **Add/Remove button**.

Getting to Know *Cool Edit Pro* and Its Main Screens

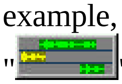
Cool Edit Pro can be thought of as being two programs in one. It can be used as a single-waveform editor that can be used to edit and process mono and stereo waveforms **and** it can also be a multitrack hard disk recording system that can digitally mix numerous audio files (using up to 64 tracks!) to either a single sound card or multiple sound cards, while also providing for real-time level and pan mix capabilities in a non-destructive editing environment.



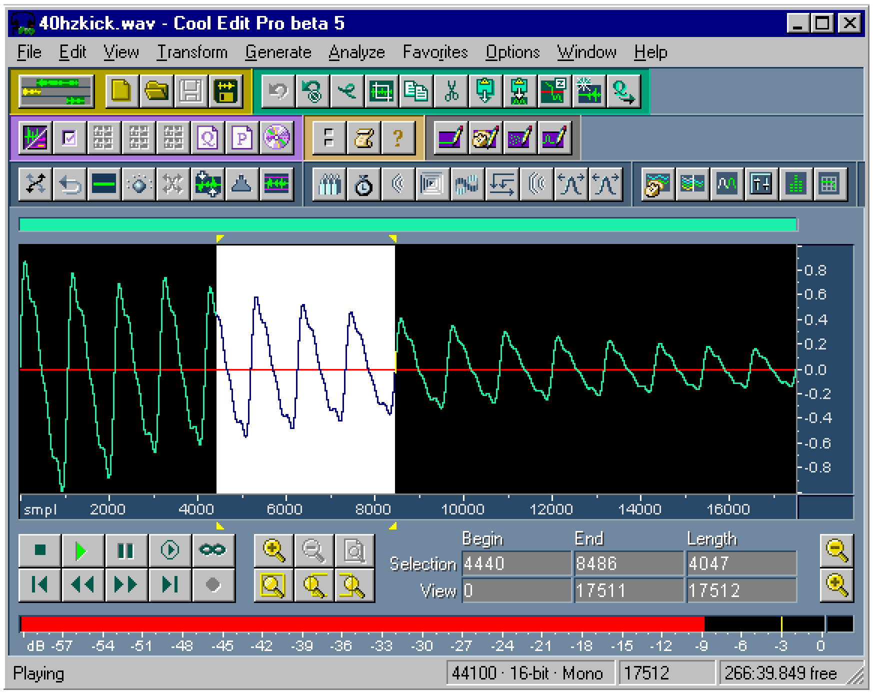
Navigating in Edit View

This chapter is meant to be a brief introduction to the various navigation and functional controls of this deceptively simple-yet-powerful program. Please, take some time out to browse through the various chapters within this on-line manual, and check out the various capabilities that have given this editor its name.

Cool Edit Pro offers two main edit modes: **Edit View** and **Multitrack View**. To choose between these options, select the one you want from the View menu or click-on the Edit Mode icon to toggle between the two operating states. A checkmark will appear next to the view mode that's currently selected.

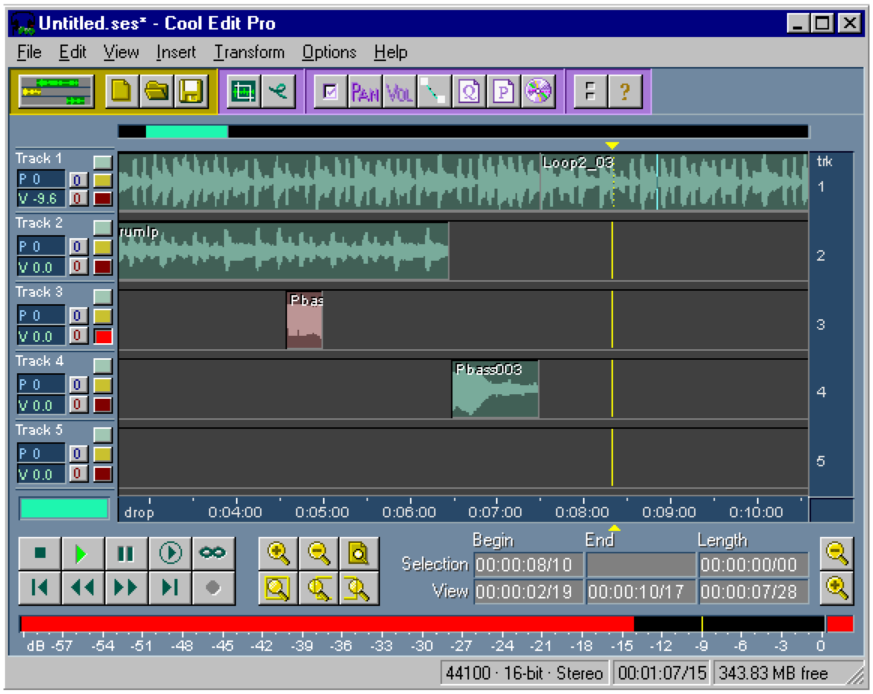
The Edit mode icon that is currently displayed on the screen represents the alternate operating mode that the system will "jump" to when pressed. For example, when working in the Edit View mode, the Multitrack View icon "" will appear. Conversely, when working in the Multitrack View mode, the Edit View icon "" will appear.

Edit View - When in the **Edit View Mode**, *Cool Edit Pro* can be thought of as being a single-waveform editor that can be used to edit and process mono and stereo waveforms. Once edited, the audio files can be saved or played back through any sound card that has been installed within your computer.



Edit View - Main Screen

Multitrack View - When in the **Multitrack View Mode**, *Cool Edit Pro* can be thought of as being a multitrack hard disk recording system that can digitally mix numerous audio files (using up to 64 tracks!) to either a single sound card or multiple sound cards, while also providing for real-time level and pan mix capabilities in a non-destructive editing environment.



Multitrack View - Main Screen

Before you get started, I'd like to clue you in on one of the cooler little tricks that has been implemented into *Cool Edit Pro*: the **Right-Click Button**. Whenever you see a simple function button, window, or waveform action, you might try right-clicking on it. Chances are that you'll be surprised by a useful shortcut menu or a set of handy options that can make you life just a little bit easier. Go ahead, give it a try!

Menus

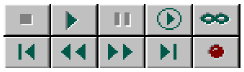
The Edit View's pulldown **Menus** offer quick and easy access to all of *Cool Edit Pro*'s Session and audio file handling, editing and signal processing functions.

Customizable Tool Bars

Many of *Cool Edit Pro*'s most commonly used functions are represented as icons within the Toolbar, giving you instant access to functions at the press of a button! By simply holding the mouse over any of these icons will bring up **Tooltips** that describe the function in simple terms. The toolbar can be arranged in any order by changing button layout within the **/Options/Toolbars** dialog box listing. More than one item within the list may be highlighted at a time.

The same listing can also be quickly invoked by right-clicking on any toolbar.

Transport Controls



The following items are included within *Cool Edit Pro*'s transport controls:

 **Stop:** Stops waveform or session playback.

 **Play:** Plays the portion of the wave or session that is currently being viewed (or that is highlighted) from the current cursor position.

 **Pause:** Temporarily pauses the playback or recording of audio. The button turns into a **Continue** button when audio is paused. When recording, the red record bar will turn yellow to indicate a paused state.

 **Play to End:** Plays the currently-visible waveform or session window then stops.

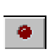
 **Play Looped:** Plays the currently-visible waveform or session window in a continuous loop fashion.

 **Go to Beginning:** Places the playback cursor at the beginning of the waveform.

 **Rewind:** Shuttles the playbar cursor backwards in time. This function supports scrubbing, meaning that the audio file will be played back as it shuttles over the waveform. Right-clicking on the Rewind button will allow you to set the rate at which the cursor will shuttle (ranging from a constant rate of 2X up to 8X, or several revisable shuttle rates.)

 **Fast Forward:** Shuttles the playbar cursor forwards in time. This function supports scrubbing, meaning that the audio file will be played back as it shuttles over the waveform. Right-clicking on the Rewind button will allow you to set the rate at which the cursor will shuttle (ranging from a constant rate of 2X up to 8X, or several revisable shuttle rates.)

 **Go to End:** Places the playback cursor at the end of the waveform.

 **Record:** Starts recording from the current playbar cursor position. Any waveform data after this point will be recorded over.

Recording and playing files with Cool Edit Pro

Recording with *Cool Edit Pro* 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 and then save your recording.

You may need to adjust your input signal to obtain the optimum recording and signal-to-noise levels.

Level Meters

The **VU Record Level Meters** 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, simply double-click on the level meters. You can also right-click on the meters to configure them.

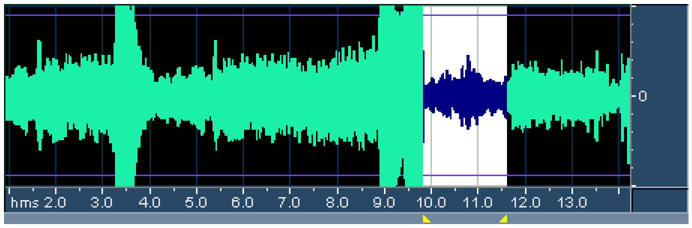
- **Double-Clicking** on the Record Level Meter will start and stop monitoring.
- **Right-Clicking** the display will bring up the Level Meter configuration menu.

Level Meters

The levels displayed represent the peak amplitude in decibels, where a level of 0dB is the absolute maximum before clipping occurs. If clipping does occur, the clip indicator to the right of the meter will light up. Just click on the clipping indicator to clear it at any time. When displaying stereo audio, the top meter represents the left channel, and the bottom the right. Yellow peak indicators will "stick" for 1-1/2 seconds, so that you can easily read the peak amplitude. If the option to **Adjust for DC offset** is enabled, a false clip reading may occur. Disable the DC offset adjustment to have the clip indicators only light up when absolute clipping occurs.

Waveform Display

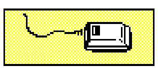
As its name suggests, the **Waveform Display** window is the area where track- and waveform-based data is displayed. It's the heart and soul of *Cool Edit Pro*, where audio files are placed in time, non-destructively edited, looped, mixed with respect to both level and panning parameters, etc.



Waveform Display

Waveform Navigation

Cool Edit Pro 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. The following list describes the mouse movements that are crucial to navigating around the **Waveform View** window.



Defining a range: Left-click on one side of the range portion that you want to select and drag the mouse to the left or right.

Changing Range Boundaries: Hold down the **Shift** key and Left-click on the range boundary that you want to move and drag the boundary to its new location.

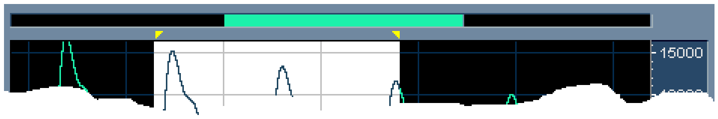
Select All: **Double-clicking** in the viewing field will select the entire viewing field.

Zoom Options: See **Horizontal** and **Vertical Zoom** options below.

Selecting L/R Channels: 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 will acquire an "**L**" or an "**R**" when you do this.

Display Range Bar

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



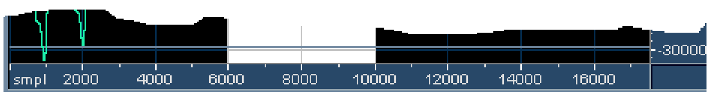
Display Range Bar

Left-click and dragging the little hand horizontally over the **Green** portion of the Display Range Bar will allow you to scroll the current Zoom range forwards and backward in time. i.e.: the zoom will remain the same, however the waveform area being viewed will be shifted either forward or backwards.

Right-click and dragging the magnifying glass on either the left- or right-hand boundary of the Display Range Bar's **Green** area will let you shrink or expand the waveform's actual zoom range. Dragging on the Left-hand portion of the range will shift the range at the waveform's leading edge, while retaining the waveform's right-hand edge position. Dragging on the Right-hand portion of the range will do the reverse.

Time Ruler

The **Time Ruler** shows the time at any point along the waveform's timeline. The vertical line and yellow connecting arrows that fall above and below the waveform indicate the current playback cursor position. In addition, whenever a wave is being played back, a vertical bar will travel along the ruler's timeline, showing the current playing position.



Ruler Display

Whenever a waveform is zoomed in, **Left-click and dragging** the little hand horizontally over the **Time Ruler** will allow you to scroll the current Zoom range forwards and backward in time. i.e.: the zoom will remain the same, however the waveform area being viewed will be shifted either forward or backwards.

Right-click and dragging the magnifying glass over the **Time Ruler** will automatically zoom the waveform window to the specified in- and out-points.

Right-clicking on the **Time Ruler** will call up a pop-up menu that lets you to quickly and easily select various Zoom, Snap, Time Display and Edit Tempo options.

Zoom Pop-up Menu

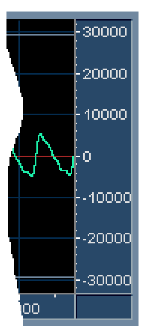
Double-Clicking on the ruler changes the displayed time format.

When you have zoomed in far enough, *Cool Edit Pro* 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).

Further information on the display time format can be found within the **/View/Display Time Format** section.)

Amplitude Ruler

The **Amplitude Ruler** displays the relative amplitude of a waveform over time. The ruler's display format can be set to either Samples (exact sample value of the data), a percentage (from -100% to 100%, where 100% is 0dB) or as a normalized value (-1 to 1) in **Waveform View**. In **Spectral View**, the vertical ruler is always in frequency (Hz) format. The display format can easily be changed, using the following mouse commands:



Vertical Ruler

Left-click and dragging the little hand vertically over the **Amplitude Ruler** will allow you to scroll through *Cool Edit Pro*'s visible amplitude ranges that might extend beyond the traditional amplitude scale values.

Right-click and dragging the magnifying glass over the **Amplitude Ruler** will automatically zoom the waveform window to show only the specified amplitude scale.

Double-clicking on the **Amplitude Ruler** will toggle the display between either Samples (exact sample value of the data), a percentage (from -100% to 100%, where 100% is 0dB) or as a normalized value (-1 to 1) in **Waveform View**. In **Spectral View**, the vertical ruler is always in frequency (Hz) format.

Right-clicking on the **Amplitude Ruler** will call up a pop-up dialog window that lets you to quickly and easily select various Zoom and display options.

Further information on the vertical scale format can be found within the **/View/Vertical Scale Format** section.)

Horizontal (Timeline) Zoom

The following items are included within *Cool Edit Pro*'s Timeline Zoom controls:

Zoom in to Center:	Zooms in on the center of the visible waveform window.
Zoom Out:	Zooms out from the center of the visible waveform window.
Zoom Out Full:	Zooms out to display the entire waveform or waveforms that are contained within a session.
Zoom to Selection:	Zooms in on the actively selected waveform range.
Zoom to Left Selection:	Zooms in on the left-hand boundary of the actively selected waveform range.
Zoom to Right Selection:	Zooms in on the right-hand boundary of the actively selected waveform range.

Vertical (Amplitude) or Track Zoom

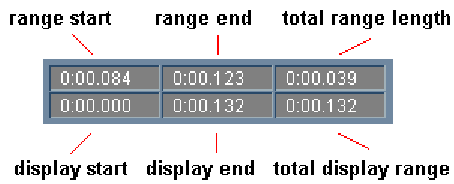
The following items are included within *Cool Edit Pro*'s Vertical Zoom controls:

Zoom Out:	Edit Mode: Decreases the vertical scale resolution of a waveform's amplitude display. Multitrack Mode: Increases the number of tracks that are displayed within the Waveform Window.
Zoom In:	Edit Mode: Increases the vertical scale resolution of a waveform's amplitude display. Multitrack Mode: Decreases the number of tracks that are displayed within the Waveform Window.

Other Important Display Functions:

Time Display Windows

The **Time Display** windows in the lower right area of the main window shows the Starting and Ending points of the current range selection, as well as for the portion of the wave that's currently being viewed. The waveform range and display range will be shown using the current time display format.



Time Display Windows

Left-Clicking on any of these windows will let you directly enter numeric time display data into the appropriate window.

Clicking on both mouse buttons within any of these windows will call up a copy/paste and options selection box.

Status Bar

The **Status Bar** windows at the very bottom of the screen display the format of the wave in Channels, Sample Rate, and Bits Per Sample. In addition, data relating to file size, free disk space and SMPTE time may also be displayed

Wave Format Windows

Right-Clicking on any of these windows will call up an option selection box, which allows for specific format options to be turned on or off.

Wave file formats Supported by *Cool Edit Pro*

The following is a list of the various file formats that *Cool Edit Pro* 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 (.WAV)

Microsoft Windows format. Windows WAV files support both mono and stereo files at a variety of resolutions and sample rates. This file type follows the RIFF (Resource Information File Format) specification, and allows for extra user information to be embedded and saved with the wave file. The standard Windows PCM waveform contains PCM coded data, which is pure, uncompressed pulse code modulation formatted data.

Options

These formatting options are available only to 32-bit files, and will appear grayed-out at other times:

24-bit (type 1) PCM	Standard used for 24-bit cards
32-bit (type 1) 16.8 float	A float type whose range is +/-32768.0
32-bit normalized (type 3)	A standard float type whose range is +/- 1.0

Microsoft ADPCM (.WAV)

The Microsoft ADPCM format consists of 4-bit per channel compressed data (providing 4:1 compression). Files saved in this format will automatically be expanded to 16-bits when loaded, regardless of their original resolution. For this reason, it is best to save to this format from 16-bits, rather than 8-bit as the quality will be much greater. After expanding, the 16-bit data can still be quickly converted to 8-bit during playback.

Options

Choosing the Multiple Pass option will take longer to save, but the quality will be better. The time taken to read an ADPCM compressed file is the same no matter which option you choose.

VBase ADPCM (.VBA)

This is Dialogic VOX with a small header, and allows for sections to be marked which show up in the cue list. It will only save mono 16-bit audio, and like other ADPCM formats, it compresses to 4-bits/sample (for a 4:1 ratio). Unlike the Dialogic VOX format, sample rate information is retained with the file.

DVI/IMA ADPCM (.WAV)

The International Multimedia Association (IMA) flavor of ADPCM compresses 16-bit data to 4-bits/sample (4:1) using a different (faster) method than Microsoft ADPCM, and has different distortion characteristics, which can give better, or worse results depending on the sample being compressed. As with Microsoft ADPCM, it is best to save to this format from 16-bit rather than 8-bit. This format also allows for 3-bit compression (5.3:1) as well at a slightly lower quality, though few sound drivers support the 3-bit ADPCM (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/IMA compressed data properly, but lately this has been changing as other manufacturers are providing DVI/IMA 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 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. These compression rates are less compatible than the standard 4-bit, and may not work with other systems. We have found that the preceding index tables provided the best quality.

This compression scheme can be a good alternative to MPEG; it provides reasonably fast decoding of 4:1 compression, and it degrades sample quality only slightly.

A/mu-Law Wave (.WAV)

A-Law and mu-Law formats (CCITT standard G.711) are common in telephony applications. These encoding formats compress original 16-bit audio down to 8 bits (for a 2:1 compression ratio) with a dynamic range of about 13-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 bit more distortion than the original 16-bit audio. The quality is higher than you would get with 4-bit ADPCM formats.

Options

Mu-law 8-bit is the international standard telecommunications encoding format, while A-Law is a slight variation found in European systems.

Files saved in this format will automatically be expanded to 16-bits when loaded, so you should not save to this format from 8-bit.

Sound Blaster (.VOC)

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

Apple AIFF (.AIF, .SND)

This is Apple's standard wave file format. Like Windows WAV, AIFF files support mono or stereo, 16-bit or 8-bit, and a wide range of sample rates. *Cool Edit Pro* only supports the PCM encoded portion of the data, even though this format (like Windows WAV) can contain any one of a number of data formats. The AIFF format is a good choice for PC/Mac cross-platform compatibility. When opening in *Cool Edit Pro*, you can simply add the .AIF extension to the file and load it using this file filter. When transferred to a Mac, you can add the four character code "AIFF" in the file's resource fork to have it recognized (The Macintosh identifies a file through its “resource,” which is stripped away when a file is opened on a PC. However, many Mac applications which support AIFF can recognize the PCM data without this identifier.)

NMS VCE (.VCE)

Natural MicroSystems (NMS) ADPCM This is an optimized G.721 ADPCM variant used in telephony applications. A .vce file contains a single voice message in 2, 3, or 4-bits per sample at 8000 samples per second. This format supports only mono, 16-bit at 8kHz. The 8-bit sample format at 8000 samples per second is framed A-law or mu-law. The .vce file is a flat file with no header. Multiple .vce files and a header are contained in a single NMS .vox file which is not supported by *Cool Edit Pro*.

Options

Options displays the various compression schemes for this format.

ASCII Text Data (.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

etc...

Options

An optional header can be placed before the data. Data can also be normalized between -1.0 and 1.0, or written out and read in raw sample values.

8-bit signed (.SAM)

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. 8-bit signed raw format 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.

Next/Sun (.AU, .SND)

This format is the standard found on NeXT and Sun computers, and has many data types. *Cool Edit Pro* supports the CCITT mu-Law, A-Law, G.721 ADPCM, and linear PCM data variants. Like Windows WAV and AIFF files, this support can support mono or stereo, 16-bit or 8-bit, and a wide range of sample rates when saved as linear PCM. The most common use for the AU file format is for compressing 16-bit data to 8-bit mu-law data. AU is used quite extensively for distribution on the Internet, and for inclusion in JAVA applications and applets.

Options

Cool Edit Pro offers three compression schemes for NeXT/Sun: mu-Law, A-Law, and CCITT G.721 (ADPCM at 32 kbits/sec), as well as linear PCM.

SampleVision (.SMP)

The SampleVision format is used by Turtle Beach’s SampleVision program. It supports only mono 16-bit audio. If your data is in a different format, you will be asked to convert it before saving. This format supports loop points, which can be edited using *Cool Edit Pro*’s 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: 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)

The Dialogic ADPCM format is commonly found in telephony applications, and has been optimized for low sample rate voice. It will only save mono 16-bit audio, and like other ADPCM formats, it compresses to 4-bits/sample (for a 4:1 ratio). 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 a sample rate unless "Don't Ask" is checked.

Take note of the sample rate of your audio before saving as Dialogic VOX, you will need to enter it upon reopening the file.

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

This format is simply the PCM dump of all data for the wave. No header information is contained in the file. For this reason, you are asked to select the sample rate, resolution, and number of channels upon opening. By opening audio data as PCM, you can interpret almost any audio file format! You must have some idea about the sample rate, number of channels, etc. You can also interpret the data as A-Law or mu-law compressed. When guessing at these parameters upon opening, the waveform may sound incorrect (in different ways, depending on which parameters you have mixed up). Once the waveform is loaded and sounds fine, you may hear clicks at the start, end, or sometimes throughout. These clicks are various header information being interpreted as waveform material. Just cut these out, and Voilà! You have read in a wave in an unknown format!

Options

Raw Data options include Intel or Motorola variety of uncompressed PCM, and A/mu-law encoding/decoding. When saving raw data files, an optional header can be written to a separate .DAT file to make reloading easier.

Amiga IFF-8SVX (.IFF, .SVX) (*.*)

The Amiga 8SVX format is an 8-bit mono format from the Commodore Amiga computer, which can also be compressed to a 4-bit Fibonacci delta encoded format. A variety of sample rates are supported.

Options

Choose between 8-bit uncompressed or 4-bit Fibonacci delta encoded format.

Pika ADPCM (.VOX, *.*)

Pika ADPCM is a format found in telephony applications. It is a nibble-reversed version of the standard Dialogic VOX format. To save to Pika your file must be 8kHz 16-bit mono. You will be prompted to convert before saving if it is not.

ACM Waveform (.WAV)

Any file format supported by the Microsoft Audio Compression Manager (ACM) can be loaded or saved. When saving, only the ACM formats that are compatible with the format of the current waveform 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. Other formats can be provided through other companies If you own a SoundBlaster card, for example, the Creative ADPCM file format will also be available. If an option is not available for your specific sound card(s), ask the sound card provider for ACM drivers that support their file formats.

Options

The available ACM formats are displayed under Options.

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).

Name: This displays the name of any saved format presets. Three are provided initially: CD, Radio, and Telephone Quality (standard PCM).

Save As: Click this button to save preset format/attribute combinations.

Format: This is the ACM format you wish to save in. Note that this list will display only those formats that are compatible with the current waveform's properties (stereo/mono, resolution, etc.).

Attributes: The available attributes will depend on the original attributes of the file (sample rate, etc.) and the Format chosen.

DiamondWare Digitized (.DWD)

This is the audio format used by DiamondWare's Sound Toolkit, a programmer's library that lets you quickly and easily add high quality interactive audio to games and multimedia applications. It supports both mono and stereo files at a variety of resolutions and sample rates. See http://www.dw.com for more information.

RealAudio 3.0 (.RA)

This is Progressive Networks' compressed format used for real time audio streaming over the Internet. RealAudio files can be encoded using several different algorithms. Each encoding algorithm is optimized for a particular type of audio and connection speed bandwidth. Click on the Options button in /File/Save As to select the specific sub-format you want. You can save to this format from 8- or 16-bit, mono or stereo, with valid sampling rates of 8 kHz, 11.025 kHz, 16 kHz, 22.05 kHz, and 44.1 kHz. You can use a stereo source to produce a mono or a stereo RealAudio file, however you cannot use a mono input file to produce a stereo output file. In this case you will be prompted to convert to stereo before saving.

Options

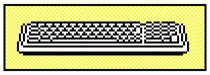
Format options are given in Music (higher quality), and Voice (lower quality) categories. A description of the currently selected algorithm is given in the options window.

NOTE: *Cool Edit Pro* can export to RealAudio format only; it cannot import .RA files.

Keyboard and Mouse Command Shortcuts

Cool Edit Pro offers a wide variety of keyboard command shortcuts that can be used to speed up the editing process. The following list fully details the default shortcut commands that can be executed, either from your computer's keyboard or via the mouse.

These commands, as well as commands that can be given using the keys on a standard MIDI keyboard instrument, can be easily edited from the **/Options/Shortcuts (Keyboard and MIDI Triggers)** menu.



KEYBOARD

Selection controls:

Ctrl+B	Select Both Channels
Ctrl+L	Select Left Channel
Ctrl+R	Select Right Channel
Ctrl+A	Select Entire Waveform
Left Arrow	Adjust left side of highlight one pixel to the left
Right Arrow	Adjust left side of highlight one pixel to the right
Shift+Left Arrow	Adjust right side of highlight one pixel to the left
Shift+Right Arrow	Adjust right side of highlight one pixel to the right
F4	Adjust Settings - colors, sound card devices, memory buffering, et cetera
Escape (Esc)	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:

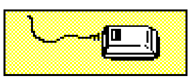
Space	Toggle Play / Stop
Ctrl+Space	Toggle Record / Pause
Alt+S	Stop (like Space when already Playing)
Shift+Space	Toggle "Play from Cursor To End" / Pause

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 screen-full (doesn't affect cursor)
Page Up	Scroll backward one screen-full (doesn't affect cursor)

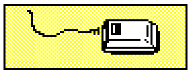
Miscellaneous:

Ctrl+N	Creates a new, initially blank, file
Ctrl+O	Opens an existing audio file
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 waveform 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.



MOUSE (cont'd.)

- 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
- Function Presets

Many of the functions in *Cool Edit Pro* have presets that are available for easily storing and recalling your favorite settings. New presets can be added at any time. All preset information is saved in the COOL.INI, usually in your Windows directory.

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

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

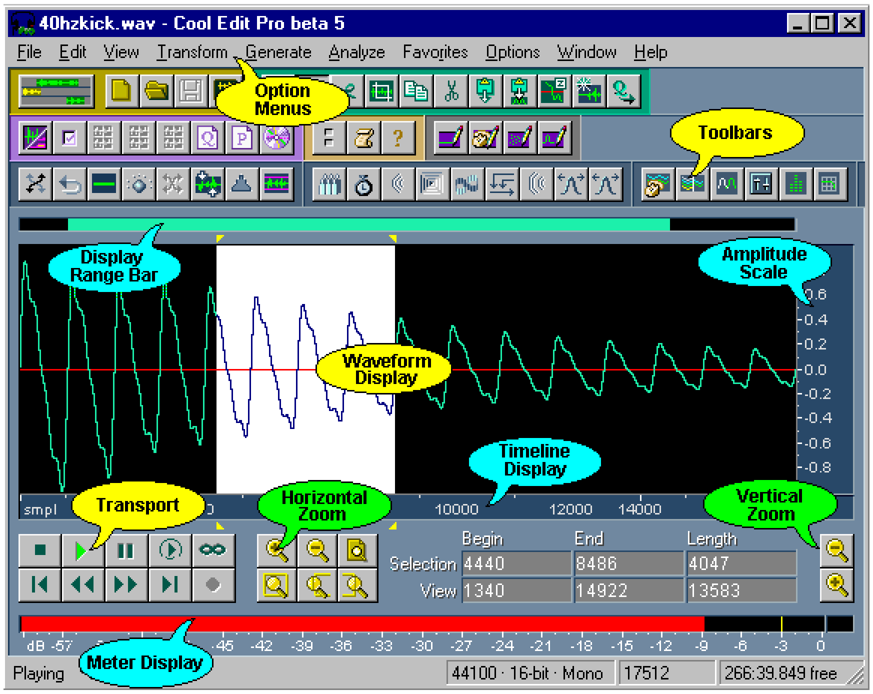
To remove a preset from the list, choose the preset, and press **Del**.

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

If you find that your presets aren't being saved, please check to ensure that the COOL.INI file is not more than 64K in size. This should never happen, but it just might be possible if you have hundreds and hundreds of presets.

Navigating the Edit View screen

The following is a typical layout for *Cool Edit Pro*'s **Edit View Screen**:



Navigating in Edit View

Before you get started, I'd like to clue you in on one of the cooler little tricks that has been implemented into *Cool Edit Pro*: the **Right-Click Button**. Whenever you see a simple function button, window, or waveform action, you might try right-clicking on it. Chances are that you'll be surprised by a useful shortcut menu or a set of handy options that can make you life just a little bit easier. Go ahead, give it a try!

Menus

The Edit View's pull-down **Menus** allows for quick and easy access to all of *Cool Edit Pro*'s Session and audio file handling, editing and signal processing functions.

Customizable Tool Bars

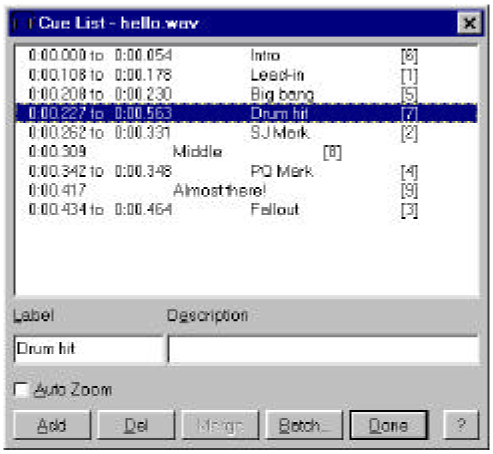
Many of *Cool Edit Pro*'s most commonly used functions are represented as icons within the Toolbar, giving you instant access to functions at the press of a button! By simply holding the mouse over any of these icons will bring up a **Quick Help pop-up message** that describes the function in simple terms. The toolbar can be arranged in any order by changing button layout within the **/Options/Toolbars** dialog box listing. More than one item within the list may be highlighted at a time.

The same listing can also be quickly invoked by right-clicking on any toolbar.

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**.

Markers (F8)

The **View/Cue List** option can be used anytime to mark your current selection so that you can return to it later. If you would like for *Cool Edit Pro* to remember your highlighted selection, or just your current cursor point, click Add in the cue list, and quickly type a name for your selection. In the future, if you want to return the cursor to that point, or re-highlight that selection, double-click the name or choose the name and click Goto. 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.



Cue List

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

A **Red Arrow**, when present, represents a Cue List entries for a single marked cue point.

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.

Pressing **F8** will automatically add the currently defined range to the **Cue List**. Once a cue point has been entered into the Cue List, the **yellow** marker points will be replaced with static **blue** markers. Either the **Blue** or **Red** cue entries can easily be referenced back to by simply double-clicking on the entry within the list.

A maximum of 96 cues may be entered into the Cue List.

Right-clicking on the **yellow** marker points will pull up a pop-up menu that will let you find the waveform's **next/previous zero crossing**, **Find Next Beat**, and enter a range into the **Cue List (F8)**.

File Menu (Edit View)

The file menu displays all of *Cool Edit Pro*'s Edit View file-handling options. These are:

New (Ctrl+N)

Use **/File/New** to create a new waveform. When you create a new waveform, *Cool Edit Pro* will ask you to specify the waveform's properties.

Higher sampling rates, stereo, and/or higher bit resolutions will result in higher quality sounds, but do require more hard disk space.

Sample Rate: The sampling rate you choose determines the overall bandwidth (how many frequencies can be encoded within the audio signal), with higher sampling rates yielding a wider bandwidth. You can enter any sample rate directly, or choose a common sample rate from the list. Note that most sound cards support only certain sample rates.

Channels: Mono waveforms support one channel of audio information. Stereo files require twice the disk space, because they represent two channels of information, a left channel and a right channel.

Resolution

Select the bit resolution for the file:

8-bit:	Select this for use in telephony/Internet applications.
16-bit:	Select this for standard professional/CD quality.
32-bit:	Work at the 32-bit level for best quality when processing your audio. When done, you can convert down to 16- or 8-bit for output using dither and achieve better results then if staying in the 8- or 16-bit domain. You can also use 32-bit for sound cards that go beyond the 16-bit standard (as in a 24-bit card). Press the **Options** button in the **Save As...** dialog to access 32-bit options.

To achieve higher quality, you may want to work at the 32-bit level while processing your audio. Since current sound cards cannot playback audio that has been stored and processed using the full 32-bits, this rate must be converted downward for playback. It's best to use this mode when using audio cards that have resolutions greater than 16-bit (20-bits as of this writing).

Certain combinations of sample rate, channels, and resolution may not be available on your system. To see the capabilities of your system, check the Devices tab under **/Options/Settings**. Although *Cool Edit Pro* can create and edit those files, your sound card may not be able to play them properly.

Open (Ctrl+O)

Opens an existing sound file. *Cool Edit Pro* supports a wide variety of data types. When you load a file for editing, *Cool Edit Pro* converts the waveform type to its own internal temporary file type for faster editing and better handling of larger file sizes.

Recent Directories: This displays the most recently used directories for quicker access to common locations.

Look In: Navigates through all available drives and directories.

File Name: You can specify the name of the file you wish to open here, if it resides in the current directory. Entering *.extension will display all files with that extension at your current location, and *.* will display all files.

Files of Type: Select from the list of supported file types to display any files with a corresponding extension.

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

Show File Information: Displays basic information about the audio file, such as the file format, uncompressed file size, and running time.

Auto Play: With Auto Play enabled, any selected file in the Open dialog will be auditioned, provided that its format can normally be played on your system. 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).

You can open several files at once (with each being 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.

Whenever multiple files are open, you can easy select between these files using the Control-Shift keys (which shifts between open files in the forward direction) or using the Control-Shift-Tab keys (which shifts between open files in the reverse direction).

Open As

Like Open, Open As lets you specify the sample format before opening a file to allow automatic conversion to a sample-rate, bit-rate or channel-type. However, Open As offers a "quick-and-easy" conversion method: if the target sample rate is different, it will not try to pre-filter or post-filter the samples. This function is handy for opening files as different bit rates and number of channels, but for differing sample rates, open the file with **Open**, and then use **Convert Sample Type**.

Open Append

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'll get best results if you only append files of the same sample rate, because no number-crunching would be required to change from one rate to the native rate.

You'll notice that appended file cue points are placed within the Cue List, making it easy to find the edit boundaries at a later time.

Open an audio file, choose **/File/Open Append** and append another waveform at the end. If the two don't have the same format, *Cool Edit Pro* will convert the appended waveform to the current format.

Revert to Saved

Reloads the current waveform from disk, discarding any edits that were made since the last time the file was saved.

Close

This will close the currently opened audio file (bye, bye!). When a waveform is closed, the associated temporary file is removed, thus freeing up the hard disk space that was previously being used.

Close All

This command has options for closing multiple waveforms at once.

Waves Not in Use: Closes any open waveforms not currently in use by the current Session (not inserted in the multitrack environment).

Waves and Session: Closes all open waveforms and the currently open Session (if any) in Multitrack View.

Save

Saves the audio file under its current name and format.

Save As

Use **/File/Save As…** to save the current wave to a new name or location, or format.

Recent Directories: This displays the most recently used directories, for quicker access to common locations.

Save In: Navigates through all available drives and directories.

File Name: Specify a name for the file.

Save as Type: Select the format you wish to save as from the supported list. Some formats cannot be written to from certain sample types. In this case, *Cool Edit Pro* will ask if you wish to convert before saving.

Save extra non-audio information: If you don't want to save header fields such as the copyright, author, and others, un-check the this box.

Options: Some file formats support various options that can be modified here.

Save Selection

/File/Save Selection allows for saving of just the highlighted selection to a file. Its properties are identical to those in **Save As…**

This feature is very useful for saving small segments of a larger audio file. For example, Should you wish to record a number of samples from a friend's sampler, you could record then as a single, continuous audio file. At a later time, you could highlight and process each file (EQ, normalize, etc.) separately and then save them to their own directory, using the **/File/Save Selection** feature.

Recent Directories: This displays the most recently used directories, for quicker access to common locations.

Save In: Navigates through all available drives and directories.

File Name: Specify a name for the file.

Save as Type: Select the format you wish to save as from the supported list. Some formats cannot be written to from certain sample types. In this case, *Cool Edit Pro* will ask if you wish to convert before saving.

Save extra non-audio information: If you do not want to save header fields such as the copyright, author, and others, un-check the this box.

Options: Some file formats support various options that can be modified here.

Flush Virtual File

Cool Edit opens a .wav file and uses it directly without making a backup in the temp directory. If the .wav file needs to be closed (to delete, or be opened by another program for writing), then flushing it will copy the .wav file's contents to the temp directory and close the file. A file is automatically flushed if the entire file is modified by *Cool Edit*, since all changes are written to the temp directory.

In a nutshell: this option allows a audio file that's currently being used by *Cool Edit Pro* to be simultaneously used by another program or resource application.

Free Hard Drive Space...

The Free Hard Drive Space function provides a way to manage *Cool Edit Pro's* use of the space available on your drive. This dialog can be entered into at any time by choosing it from the from the **File** menu, or it will automatically pop up if space available nears 0k. On systems with lots of hard drive space, undo levels will be cleared automatically, so this box may not appear often.

Waveform: The Waveform list displays the names of the currently open waveforms. Select the name of the file for which you would like to remove Undo instances. You can also press the Close File button to the right to close the selected file if no longer needed. This will free up hard drive space as well.

Undo History: Lists the "undone actions" that are currently being retained in memory. Each instance listed here consumes hard drive space equivalent to the amount used by the original file (the one highlighted in the Waveform list). For example, if the file selected in the Waveform list is a 500k file, each item in the Undo History list uses an additional 500k. To remove items from the History list, select it from the list and press the Clear Undo(s) button. All items at the selected level and below are removed.

Lower Hard Drive Reserves: *Cool Edit Pro* creates temporary files for use when performing edits on your audio. The Primary and Secondary drives listed here are those set in **Options/Settings/System**, and are used to store these temp files. You can enter an amount (in MB) to keep free on each drive, which *Cool Edit Pro* will not make use of should available space get that low. The Set Reserves button registers the Reserve amounts you have entered with the rest of the program. Press this button before exiting the dialog if you wish to keep the settings listed there.

Total Available Space: This displays the total amount of available space for use by *Cool Edit Pro* (the sum of your Primary and Secondary Drives). This value should grow as you free up space.

Cancel Last Operation: You can press this button to stop any action in progress, such as a Transform, or other edit. This becomes useful only if the Free Hard Drive Space dialog has been invoked automatically by *Cool Edit Pro*, indicating you have run out of space.

MRU List

The "most recently used" audio file list, displays the eight files that were most recently opened. You can quickly open any of these files again by selecting it from the list with the mouse or by typing the number that's associated with the desired file.

This is an easy one to show you. Simply open up a few audio files and then shut *Cool Edit Pro* down. Upon re-opening *Cool Edit Pro*, you'll see a listing within the File Menu that shows the files that you've recently opened. Simply click on a favorite file, and it pops up in the Waveform View window, ready for playback or editing.

Exit

Closes *Cool Edit Pro*. (I'm outta here, man!) At close time, you will be asked if you would like to save any unsaved files which were modified, and any temporary files that were created will be removed.

Edit Menu (Edit View)

The edit menu displays all of the options that relate to basic waveform editing in the Edit View mode. These options are:

Undo **(Ctrl+Z or Alt+Backpsace)**

Undoes the last action.

Can't Undo

Ya gotta do something first, before you can undo it!

Undo information is stored as a temporary file on your hard drive. If the Undo function is enabled in the **/Options/Settings** option, an unlimited number of undos can be retained (actually, the number's only limited by your hard drive space.) This means that those of us who are less than perfect, can back out of almost any corner that we might paint ourselves into, as the information will be automatically saved as one or more files (~NDOnnnn) in your temporary directory until the changes have been saved as a audio file.

When working with very large audio files, it's possible that you might not have enough disk space to save the Undo information before continuing on with an operation. It's also possible that the time required to save the Undo information of very large audio files can slow your work down. In either case the problem can be solved by turning the undo function off in the settings dialog box by removing the check from **/Edit/Enable Undo** before processing, or you can simply press the "Skip" button in the Saving Undo Data dialog while Undo information is being saved.

If you find that you don't have enough disk space to save the Undo information, you can change the Temp Directory to a different drive, if available. You can do this under **/Options/Settings/System/Temp Directory**.

Enable/Disable Undo

Turns the Undo functions on and off.

Because Undo requires extra disk space for its temporary files, and time to save them before processing, you may want to disable the Undo feature. 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 to disk.

Repeat Last Command (F2 or F3)

Guess what? This function repeats the last command. Select **/Edit/Repeat Last Command** or press **F2** to repeat the last function that modified waveform data. Press **F3** to immediately repeat the command, bypassing any settings dialogs. In the latter case, *Cool Edit Pro* will automatically apply the last settings that were used.

Windows Clipboard

You can use this option to copy waveform 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 Pro*'s internal clipboard is not. However, *Cool Edit Pro*'s internal clipboard doesn't have the memory limitations of the Windows clipboard.

Copy: Copies selection or wave to the Windows clipboard.

Paste: Inserts waveform data from the Windows clipboard at the insertion point, or replacing the currently selected data. If the format of the waveform data in the clipboard differs from the format it is being pasted into, *Cool Edit Pro* converts it before pasting.

Paste to New: Creates a new file and inserts the waveform data from the Windows clipboard. The new file will automatically adopt the properties (sample rate, etc.) from the original clipboard material.

Copy (Ctrl+C or Ctrl+Insert)

This copies the current selection or wave to the internal clipboard.

Cool Edit Pro 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. If you want to copy data to another application, use **/Edit/Windows Clipboard/Copy**.

Cut (Ctrl+X or Shift+delete)

Removes the current selection and places it on the internal clipboard.

Paste (Ctrl+V)

Inserts data from the internal clipboard at the insertion point, or replaces the currently selected data. If the format of the waveform data on the clipboard differs from the format of the file it is being pasted into, *Cool Edit Pro* will convert it before pasting.

Paste to New

Creates a new file and inserts waveform data from the internal clipboard. The new file automatically adopts the properties (sample rate, etc.) from the original clipboard material.

Mix Paste (Ctrl+M)

Use **Mix Paste** to mix any audio data from either the Windows or the internal clipboard with the current wave. Clipboard data is inserted or overlapped starting at the current insertion point or selection. If the format of the waveform data on the clipboard differs from the format of the file it is being pasted into, *Cool Edit Pro* will convert it before pasting.

Volume

Use the volume slides to paste an amplified/attenuated version of the clipboard wave into the current waveform. You can paste single channels by adjusting the volume slides. Volume is represented as a percentage. For example, 10% is about -10dB, 50% is -6dB, etc.

Invert

Choose Invert to invert the data on the clipboard before pasting. This is very handy in taking the difference between two samples. For example, after filtering, you can listen to the audio that was filtered by copying the selection, choosing Undo, then Mix Paste with Invert checked. After auditioning, you can restore the original sample with Undo.

Lock left/Right

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

Insert

Inserts clipboard waveform data at the current location or selection, replacing any selected data. If no selection has been made, *Cool Edit Pro* inserts clipboard material at the cursor location, moving any existing data to the end of the inserted material.

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 waveform. If the clipboard waveform is longer than the current selection, the waveform will continue beyond the selection.

Modulate

Modulates the clipboard data with the current waveform. This is similar to overlapping, except that the values of the two waveforms 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 (in milliseconds)

Set a fade time (in milliseconds) to apply to the beginning and end of the clipboard data. *Cool Edit Pro* fades in the first n-milliseconds and fades out the last n-milliseconds of pasted data. Use this option for smoother transitions to/from pasted material.

From Clipboard

Choose From Clipboard to paste audio data currently on *Cool Edit Pro*'s internal clipboard.

From Windows Clipboard

Choose From Windows Clipboard to paste audio data currently on the Windows clipboard.

From File

Pastes the contents of a file. If the amount of data you want to paste is too large for the clipboard, you can use Save Selection to save 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. The Select File button pops up a window similar to *Cool Edit Pro*'s Open dialog.

Loop Paste

When checked, the clipboard waveform is pasted the number of times entered. If the clipboard waveform is longer than the current selection, the waveform will continue beyond the selection.

Open a waveform and highlight a range portion or all of it and copy it into the clipboard (Ctrl-C). Now open a new wave and use **Mix Paste** to mix the files together at any beginning point you want to (using the dialog's default values for now.)

Insert in Multitrack

Inserts the currently highlighted range as a waveform block within the multitrack View window. The block will be consecutively inserted into the next available track at the beginning of the window's timeline.

An easy way to enter a waveform range from the **Edit View** into the **Multitrack View** is by selecting the **/Edit/Insert In Multitrack** option. This simple-yet-powerful function simply takes the highlighted **Edit View** range and pastes it into the **Multitrack View** window (within the next available track at the beginning of the session's timeline).

Say, for example, that you wanted to loop a short percussion riff. A simple way to accomplish this would be to use the **/Edit/Find Beats** function to easy search for the loop points and the select **Insert In Multitrack**, switch to the **Multitrack View** mode and Choose the **/Edit/Loop Duplicate** function to create the number of loops that you want.

Select Entire Wave (Ctrl+A)

This option selects the entire waveform (from zero to the end of the wave) as the active range. It makes no difference whether the view is zoomed in or not (unlike double-clicking on the waveform, which selects only the visible portion of the wave.)

You can also select the entire wave by simply **double-left clicking** within the waveform display window.

Delete Selection (del)

Removes the current selection. The deleted portion is not copied to the clipboard, and can only be retrieved through Undo, or Revert to Saved if you haven't saved the file since deleting.

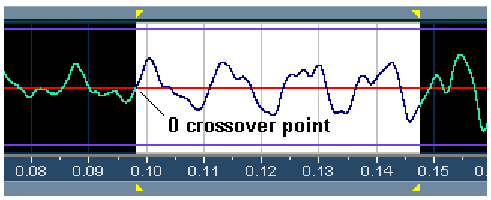
Trim (Ctrl+T)

Basically, this is the exact opposite of Delete Selection, meaning that everything is deleted except the selected portion. Use Trim to remove unwanted material around a selection. The deleted portion(s) are not copied to the clipboard, and can only be retrieved through Undo, or Revert to Saved if you haven't saved the file since trimming.

Zero Crossings

Adjusts the beginning and end points of the current selection to the nearest place where the waveform crosses the center line (zero amplitude point). Having a selection begin and end on a Zero Crossing is useful in deleting or inserting a selection in the middle of a waveform. If you don't line up the amplitudes (at zero, in this case) on both sides of the selection, the endpoints drawn together when the highlighted selection is removed are at different amplitudes, often resulting in an audible pop or click at that point.

This feature is often useful whenever you want to delete a portion of a wave or insert a portion from the middle of a wave.



Example of Zero Cross Point

To use Zero Crossings:

1 Select the portion you want to delete (or position the cursor where you want to insert)

2 Choose **/Edit/Zero Crossings** and the appropriate submenu choice:

Adjust Selection Inward (F4)	Adjusts both region boundaries inward to the next available zero crossing.
Adjust Selection Outward	Adjusts both region boundaries outward to the next available zero crossing.
Adjust Left Side to Left	Adjusts the left region boundary leftward to the next available zero crossing
Adjust Left Side to Right	Adjusts the left region boundary rightward to the next available zero crossing.
Adjust Right Side to Left	Adjusts the right region boundary leftward to the next available zero crossing.
Adjust Right Side to Right	Adjusts the right region boundary rightward to the next available zero crossing.

If the wave is not centered, you may hear audible clicking in quiet parts after processing. To adjust a waveform's DC offset to zero, highlight it and choose the "Center Wave" preset from the Amplify function. Because centering takes out all frequencies below about 16Hz, it is completely safe to do without any ill side effects.

If the amplitude levels between two waveforms don't match up at an edit or loop point, you may hear audible clicking as an edit passes or a loop repeats. To adjust for this, you can revert to the original source file and use the Zero Crossings feature to match initial and ending levels. If there is a DC offset, this DC level difference can be set to zero by highlighting it and choosing the "Center Wave" preset from the Amplify function.

Open a waveform and zoom in until the individual waveform excursions become visible. Now select a range (simply click-drag the cursor over the waveform), then choose any zero-crossing option to watch the boundaries move to the nearest zero-crossing point(s).

Find Beats

Cool Edit Pro lets you find the boundaries that make up musical beats in an existing music audio file. This allows you to easily find start and end loop points for constructing drum loops and similar phrases.

To use Find Beats:

1 Place the play cursor in the waveform to the left of your targeted loop starting point.

2 Select **/Edit/Find Beats/Find Next Beat (Left)** to locate the beginning of the current beat; and press **F11** (to repeat) until you arrive at the desired loop starting point.

3 Select **/Edit/Find Beats/Find Next Beat (Right)** to select from the current cursor position (now the loop start) to the next beat; and press **F12** until you arrive at the desired end for your loop.

4 Press the **Play Loop** button in the transport toolbar to audition your loop.

5 After any necessary tweaking you can then save, paste, or add the loop to the cue list.

If the waveform levels don't match up at the loop point, you may hear audible clicking as the loop repeats. To adjusts for this, you can revert to the original source file and use the Zero Crossings feature to match initial and ending levels. If there is a DC offset, this DC level difference can be set to zero by highlighting it and choosing the "Center Wave" preset from the Amplify function. Because centering takes out all frequencies below about 16Hz, it is completely safe to do without any ill side effects.

Snapping

In this case, the term "snap" simply means that whenever a waveform cursor is placed within a certain distance of a defined location point timeline or cue location point, that cursor will automatically "jump" or "snap" to that precise boundary point.

Snap to Ruler (Fine)

This option allows the selected waveform to snap to each of the subdivisions (decimal, SMPTE, samples, etc.) within the timeline window. Zooming in on the timeline (which is done by left-clicking and dragging within the timeline across the desired waveform area) will generally break the display down into more accurate sub-divisions, letting you place a waveform more accurately within the timeline. Both the (Fine) and (Course) snap settings can be easily accessed by right-clicking on the timeline.

Snap to Ruler (Course)

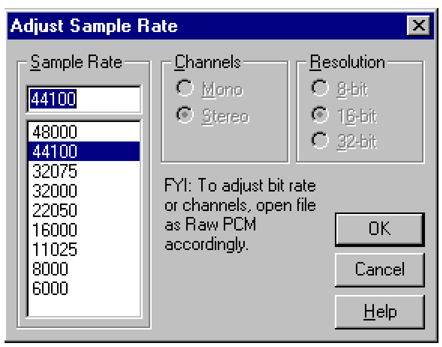
This option allows the selected waveform to only snap to the major numeric divisions (decimal, SMPTE, samples, etc.) in the timeline window. Both the (Fine) and (Course) snap settings can be easily accessed by right-clicking on the timeline.

Snap to Cue

This option allows the selected waveform to snap to a defined Cue point (see **/View/Cue List** section for information on how to define Cue Points.)

Adjust SampleRate

This setting lets you play an audio file at any sample rate that is supported by your sound card. It will not convert the audio file data to the new rate (the only thing changed is the playback speed not the data itself). If you choose a higher sampling rate, *Cool Edit Pro* plays back the sound at a higher pitch. A lower rate results in a lower pitched sound.



Adjust Sample Rate

Adjust Sample Rate will let you change only the audio file's sample rate. In order to change the bit resolution or mono/stereo interpretation, you must re-open the waveform as Raw PCM and choose the desired settings.

Certain combinations of sample rate, channels, and resolution may not be available on your system. To see the capabilities of your system, check the Devices tab under **/Options/Settings**. Although you can create and edit any sample type, your sound card may not be capable of playing it properly.

Sample Rate

The sampling rate you choose determines the overall bandwidth (how many frequencies can be encoded within the audio signal), with higher sampling rates yielding a wider bandwidth. You may enter any sample rate directly, or choose a common sample rate from the list. Note that most sound cards support only certain sample rates.

During the sampling process, an incoming analog signal is sampled at discrete time intervals. At each interval, this analog signal is momentarily "held" for observation and thus represents a specific, measurable voltage level. A mathematical conversion is used to generate a digital series of numbers that represent the signal level at that particular instant in time. Once the conversion has been made, it can be digitally stored or processed.

The **sampling rate** is defined as the number of samples (or snapshots) that are taken of an audio signal per second. Since sampling is tied directly to the component of time, a system's sampling rate will determine a system's overall bandwidth (how many frequencies can be encoded within the audio signal), with higher sampling rates yielding a wider bandwidth. The most commonly encountered sample rates used for digital audio editing are:

11,025 Hz	Poor AM Radio Quality/Speech (low-end multimedia)
22,050 Hz	Near FM Radio Quality (high-end multimedia)
32,000 Hz	Better than FM Radio Quality (standard broadcast rate)
44,100 Hz	CD Quality
48,000 Hz	DAT Quality
96,000 Hz	DVD Quality

Non-standard sample rates can be entered directly into the numeric sample rate window.

Channels

You cannot adjust this setting in **Adjust Sample Rate**.

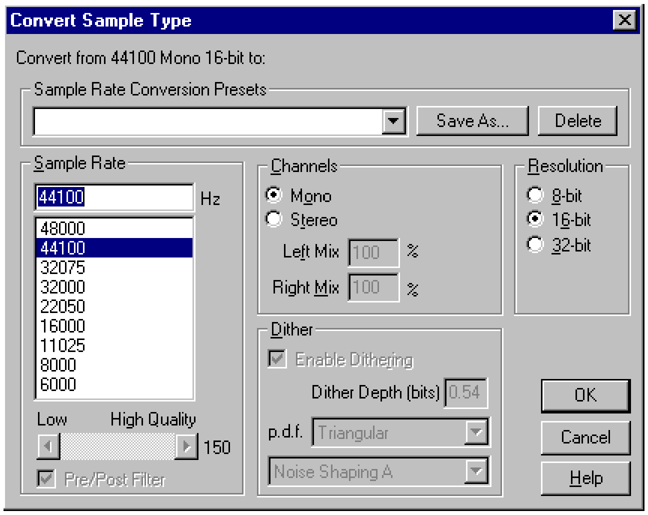
Resolution

You cannot adjust this setting in **Adjust Sample Rate**.

Convert Sample Type

This function directly converts the sample-rate, bit-rate, and channel format of an audio file that's currently loaded in *Cool Edit Pro* into a new format type (such as 44KHz/16-bit/stereo to 22KHz/8-bit/mono). Unlike **/Edit/Adjust Sample Rate**, Convert Sample Type directly processes the samples within the file, or re-samples the data, so that the audio will retain the same pitch and duration as the original file.

You can choose various levels of quality, and changes in overall and relative L/R volume levels can easily be made when converting between mono and stereo formats. You can also add Dither to reduce granulation noise when converting to lower bit-rates. Various other options are also available, allowing you to customize the conversion process to suit your own particular needs.



Convert Sample Type

Conversion Presets

You can choose any saved presets that you've created from the drop down list. If you have a sample type that you convert to often, use the Save As… button to add those settings to the Preset list. Use Delete to remove entries from the list.

Sample Rate

The sampling rate you choose determines the overall bandwidth (how many frequencies can be encoded within the audio signal), with higher sampling rates yielding a wider bandwidth. You may enter any sample rate directly, or choose a common sample rate from the list. Note that most sound cards support only certain sample rates.

Low/High Quality

Higher values retain more high frequencies while still preventing aliasing of higher frequencies to lower ones. A lower quality setting requires less processing time, but will result in certain high frequencies being "rolled-off", leading to muffled-sounding audio. Because the filter's cutoff slope is much steeper at higher quality settings, the chance of ringing at high frequencies is greater (frequencies just below the Nyquist may be abnormally boosted in level). Usually values between 100 and 400 do a great job for most conversion needs.

Higher quality settings will take longer to process, but it's worth the wait, as the resultant waveform will best retain the original audio file's overall fidelity.

You should use high quality settings whenever you downsample from a high sample rate to a low rate. When upsampling, a low quality setting will sound almost identical to a high quality setting. The difference lies in the larger phase shift that exists at higher frequencies, but since the phase shift is completely linear, it's very difficult to notice. Downsampling, at even the lowest quality setting, generally won't introduce any undesired noisy artifacts. Instead, it may just sound slightly muffled because of the increased high end filtering.

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 limit, thereby keeping them from generating false frequencies at the low end of the spectrum. <u>In general, you should enable this option for best results</u>.

Channels

Mono/Stereo: Choose whether the new sample type will be mono or stereo. If you decide to convert to Stereo from Mono, or to Mono from Stereo, you can adjust the left and right channel volumes.

Left Mix: When converting from Mono to Stereo, you can choose the relative amplitude with which the original mono signal will be placed into each side of the new stereo signal. This way, you can place the mono source on the left channel only, the right channel only, or any balance point in between. If you are converting to from Stereo to Mono, this percentage controls the amount of signal from the respective channel that will be mixed into the final mono waveform. The most common mixing methods are to use 50% of both left and right channels when converting to Mono, and 100% for both values when converting to Stereo.

To do a vocal cut you can convert the stereo waveform to mono with a Left Mix of 100% and a Right Mix of -100%. Most vocal tracks are positioned in the middle of the stereo field in-phase, so converting the signal so that it's out-of-phase will often greatly reduce the vocal track's level.

Right Mix: When converting from Mono to Stereo, you can choose the relative amplitude with which the original mono signal will be placed into each side of the new stereo signal. This way, you can place the mono source on the left channel only, the right channel only, or any balance point in between. If you are converting to from Stereo to Mono, this percentage controls the amount of signal from the respective channel that will be mixed into the final mono waveform. The most common mixing methods are to use 50% of both left and right channels when converting to Mono, and 100% for both values when converting to Stereo.

To do a vocal cut you can convert the stereo waveform to mono with a Left Mix of 100% and a Right Mix of -100%. Most vocal tracks are positioned in the middle of the stereo field in-phase, so converting the signal so that it's out-of-phase will often greatly reduce the vocal track's level.

Resolution

Choose the bit resolution to convert to. Converting to a lower resolution enables the Dither options below.

Dither

Use Dither to reduce noise and distortion to figures below their current levels through the addition of small amounts of white noise (a random signal that includes all the frequencies across the entire audio spectrum). Although a small amount of noise is introduced into the circuit, the result is far preferable to the increased distortion that you would otherwise be listening to at low signal levels. When converting to lower resolutions, dithering allows you to hear sounds that would otherwise be masked by the noise and distortion limits of 8-bit audio (or quieter than the 16-bit limit provides when converting from 32-bit.) If Dither isn't checked, quiet audio passages will just fade in and out of a disruptive, "choppy" sound that resembles falling rain or static. Whether or not dithering is used depends on the audio that's being converted and, of course, your preferences.

Enable Dithering: Enables/disables Dithering. If Dithering is not enabled, *Cool Edit Pro* truncates instead (bits that are not used are simply chopped off and discarded). This gives a "crackly" effect that fades in and out on very quiet audio passages.

Dither Depth (bits): This sets the bit amount of Dither to be applied. Generally, values of 0.2 to 0.7 give the best results without adding too much noise. If Dithering is disabled, *Cool Edit Pro* 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, you can hear a soft constant hiss in the background instead. Note, however, that as this value is lowered, other unwanted harmonic distortion noise will appear. You can usually get away with lower values for the Dither Depth when you use noise shaping curves.

p.d.f.: The "Probability Distribution Function" controls how the dithered noise is distributed away from the original audio sample value. A rectangular function means there is an equal chance that the noise value added will lie anywhere between +1 and -1 (i.e. the likelihood of a value of -0.8 being chosen is the same as that of 0.2, or 0.3, or any value between -1 and +1). The Triangular function chooses random numbers that are generally closer to 0 than to the edges -1 or +1 (i.e. the chance of 0 being chosen is twice as great as the chance of choosing 0.5 or -0.5). All dithering distribution functions can linearize the quantization noise, meaning that the noise that's heard doesn't depend on the frequency of the dithered audio, thus no harmonic distortion appears. The SNR Loss is measured against the undithered case. If modulation is present, the audible noise floor will rise and fall depending on the amplitude of the signal, so generally one doesn't want this either.

p.d.f.	**SNR Loss**	**Modulation**
Rectangular	3dB	Yes
Triangular	4.8dB	No
Gaussian	6dB	Negligible
Shaped Triangular	4.8dB	No
Shaped Gaussian	6dB	Negligible

Generally, a Triangular p.d.f. function is a wise choice, because it gives the best tradeoff between SNR, distortion, and noise modulation. The shaped versions force the noise to be skewed so that more high frequency noise and less low frequency noise exists.

Noise Shaping: The various noise shaping curves determine the placement when moving noise to different frequencies. The same amount of noise overall is present, but you can choose less noise to be placed at one frequency at the expense of placing more noise at another. So different curves will result in different types of background noise. The type of curve to use depends on the source audio, final sample rate, and bit resolution. By introducing noise shaping, you may be able to get away with lower Dither Depths to reduce the overall background noise level, without introducing much unwanted harmonic noise.

Curve	Sample Rate
A	<= 32KHz
B	<= 32KHz
C1	>= 44.1KHz
C2	>= 44.1KHz
C3	>= 44.1KHz
D	48KHz

Edit Tempo

This option calculates the tempo, or Beats per Minute (BPM), based on the currently highlighted selection. *Cool Edit Pro* uses the BPM value in displaying ruler information for the Bars and Beats time format. Also, any settings that follow the base time format will work in Bars and Beats mode as well (like Viewing Range).

Extract From Selection

Bars Highlighted: Displays the number of bars highlighted in a selection, according to the Bars and Beats time format. If this number is incorrect (which it likely will be if you have not yet defined the tempo), you can enter the correct value here to be used in extracting tempo information.

Extract: Use this button to extract tempo information from the highlighted selection. First, you have to fill in correctly the Beats per Bar field below (4 for 4/4 music, 3 for 3/4,etc.), and enter the number of known highlighted bars. The "Extract" button will then calculate the tempo and fill in the Beats per Minute and the Offset values.

Offset

Current Beat At: This is the bar/beat information for the left edge of the selection (or the current cursor point, if no selection is made). When extracting from a selection, *Cool Edit Pro* assumes that this is a down-beat (:1). You can change the actual bar and beat number for the left edge of the selection (or the current cursor point). Changing Current Beat At will update the Song Start value based on the current tempo settings.

Song Start: Song Start is the number of milliseconds into the file before the measure 1:1.00 begins. You cannot edit this value.

Tempo

Beats per Minute: The number of beats that occur over a 1 minute interval. Calculate this value by using the Extract button.

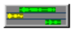
Beats per Bar: Use Beats per Bar to assign the right number of beats that occur to form one measure. This number is usually the top number in the time signature (e.g. use 4 for 4/4 time, 6 for 6/8 time, or 3 for 3/4 time, etc.).

Ticks per Beat: The number of sections each beat is divided into, or the value displayed after the decimal point: 1:1.12. You can set Ticks per Beat to any value from 2 to 3600. If you use 32 ticks/beat, then a time setting of 4:2.16 would be the eighth note (half way) between beats 2 and 3, in common time.

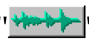
Open an audio file that is rhythmic in nature. Right-click on the waveform window's Timeline Display and choose the **Bars and Beats** timeline option. Now, highlight a range that spans several musical measures and select the **/Edit/Edit Tempo** option and press the **Extract** button. You'll notice that the beats and measures have now been automatically calculated. Once OK is pressed, the Timeline Display will be set to the correct musical Beat/Bar timing relationship for that piece of music.

View Menu (Edit View)

The View Menu displays options that relate to changing the display or viewable area when in the Edit View Mode. These options are:

 *Edit View/Multitrack View*

Cool Edit Pro offers two main edit modes: **Edit View** and **Multitrack View**. To choose between these options, select the one you want from the View menu or click-on the icon to toggle between the two operating states. A checkmark will appear next to the view mode that's currently selected.

The Edit mode icon that is currently displayed on the screen represents the alternate operating mode that the system will "jump" to when pressed. For example, when working in the Edit View mode, the Multitrack View icon "" will appear. Conversely, when working in the Multitrack View mode, the Edit View icon "" will appear.

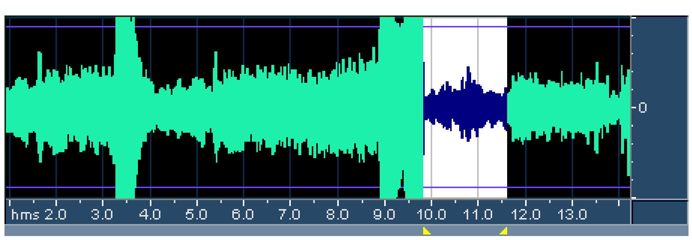
Edit View - When in the **Edit View Mode**, *Cool Edit Pro* can be thought of as being a single-waveform editor that can be used to edit and process mono and stereo waveforms. Once edited, the audio files can be saved or played back through any sound card that has been installed within your computer.

Multitrack View - When in the **Multitrack View Mode**, *Cool Edit Pro* can be thought of as being a multitrack hard disk recording system that can digitally mix numerous audio files (using up to 64 tracks!) to either a single sound card or multiple sound cards, while also providing for real-time level and pan mix capabilities in a non-destructive editing environment.

 *Waveform/Spectral View*

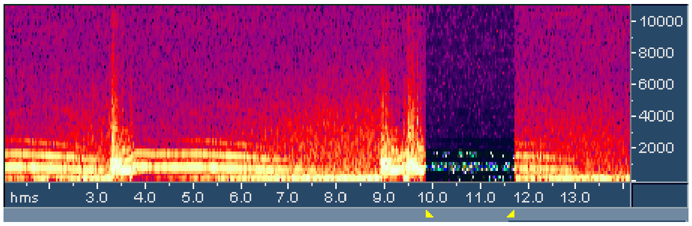
Cool Edit Pro offers two modes for viewing waveform data: Waveform and Spectral View. To choose between these options, either click-on the Waveform/Spectral view icon or select the mode that you want from the **View menu**. A checkmark will appear next to the currently selected view mode.

Waveform View displays audio data in the familiar green-on-black sound wave format, where spikes in the x-axis (vertical) indicate increased amplitude, and the y-axis (horizontal) represents time. Waveform View is the default display mode.



Waveform View

Spectral View, on the other hand, is unique in that it displays waveform data in a way, such that it's easy to view the frequency components of a audio file over time. This mode is handy for analyzing your audio data, to see which frequencies are most prevalent throughout the audio file.



Spectral View

In Spectral View, the greater a signal's amplitude component within a specific frequency range is, the brighter the displayed color will be. Colors range from dark blue (next to no amplitude components exist in this frequency range) to bright yellow (frequencies in this range are high in amplitude.)

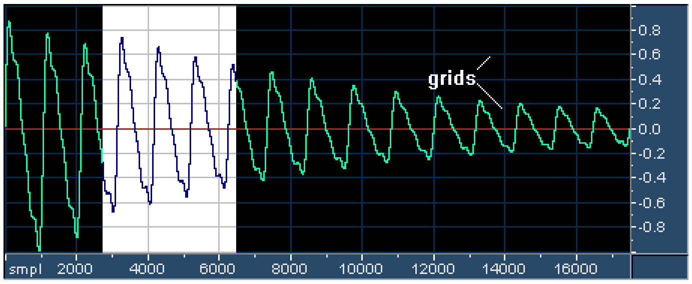
Lower frequencies are displayed near the bottom of the display, while higher frequencies are displayed from the middle-to-top part of the scale. The top of the spectrum represents frequencies at just below the Nyquist frequency, or 1/2 the sample rate. So if a bright spot appears near the top of the display for a signal sampled at 44.1 kHz, the frequency being represented is near 20 kHz.

On 256 or higher color displays, there will be more gradations between the colors, and the frequency resolution, window type, colors, and energy plot can all be fine tuned. For more information on the various settings that relate to either the Waveform or Spectral View modes, see the **Settings** topic in the **Edit View Options Menu**.

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 displayed frequency value will be one half the new sample rate.

Show Grid

When in the **Waveform View Mode**, this option turns on or off a grid display that marks off time (in the horizontal axis) and amplitude (in the vertical axis.)

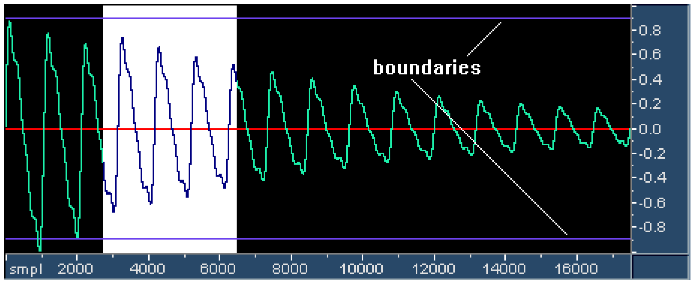


Show Grid

The Show Grid option is not available when in the Spectral View mode.

Show Boundaries

When in the Waveform View Mode, this option turns on or off the Blue horizontal boundary lines that visually indicates where the waveform's amplitude exceeds or approaches the clipping level (0 dB).



Show Boundaries

The Show Boundaries option is not available when in the **Spectral View mode**.

Show Cue Bars

Displays/hides vertical bars that mark Cue List positions.

Show Status Bar

Displays/hides the status bar (at the bottom of *Cool Edit Pro*'s main screen), which displays audio file format and disk status information.

Show Level Meters

Displays/hides *Cool Edit Pro*'s level meters (near the bottom of *Cool Edit Pro*'s main screen), which displays both record and playback level information.

Right-click on the level meter to call up several handy metering options.

 *Show CD Player*

If your computer has a CD-ROM that makes use of the [MCI] CD Audio driver, you can control the playback of CD audio from within *Cool Edit Pro*. Choosing **/View/CD Player** or clicking on the CD icon  in the View Toolbar will toggle the CD Player toolbar and controls on/off. When CD Player is on, *Cool Edit Pro* displays the controls along the bottom, below the Level Meters. The controls consist of transport icons and fields for title, track number, and location. *Cool Edit Pro* allows you to assign a title for the CD and for each track, and displays these titles the next time you use the CD. A listing of the supported functions include:

Tracks List

Clicking on any track number starts the playback of that track.

Time Readout

Displays the track's current time in minutes:seconds.

Title Display

Displays the title of the CD. You can edit this field simply by clicking in the field and typing a new title. The title defaults to the length of the CD. If you are currently playing a track (the track number is highlighted in the track list), the field displays and allows you to edit its title. Use the **TAB** key to jump to the next track to easily enter the titles for an entire CD.

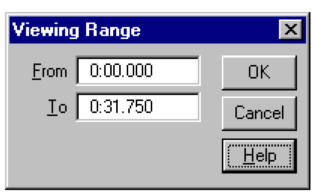
Cool Edit Pro saves titles in the file(s) COOLCDx.INI in your Windows directory, where x ranges from 0 to 99. Once one of the COOLCDx.INI files reaches about 64K in size, the next file stores new data. Song titles created with the Windows CD Player will also appear correctly, and you can edit these in *Cool Edit Pro* as well. You can enter thousands and thousands of CDs using this methods without fear of losing your favorite tracks.

CD Player control buttons:

[Stop]	Stops playing the CD. Play will resume at the start of the CD.
[Pause]	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]	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]	Rewinds the CD 10 seconds.
[Scan Forward]	Forwards the CD 10 seconds.
[Mark]	Mark the location currently being played with a red **X**.
[Goto Mark]	Go to the **X** location that was marked earlier.
[Eject]	Spits out the CD, if that's possible on your player. The icon changes to if no CD is in the drive, and if your drive can do it, the CD will insert when you press the button.

Viewing Range

Use Viewing Range to define the portion of a waveform you want to be displayed in the window. This option lets you numerically enter the left- and right-most values to serve as the visible boundaries for the x-axis (time) ruler. These values have the format currently selected in **/Edit/Display Time Format**; SMPTE, samples, etc. Once you've defined the range to be viewed, you can then select this range by double-clicking on the waveform itself.



Viewing Range

You can also double-click on the green/black display range bar (located above the waveform) to bring up the Viewing Range window. You can also enter Display From/To values in the Location Status Fields at the bottom right of the *Cool Edit Pro* window. These fields also reflect values entered in the Viewing Range window, and vice versa.

Display Time Format

The Display Time Format function lets you choose between several time display readouts, including: Decimal, SMPTE Drop, Samples, or Custom time code frame rates. *Cool Edit Pro* will then use the format chosen here when referencing time: the Time Window, the x-axis (time) ruler, time status boxes, Status Bar, etc.

You can selectively cycle through the above time formats by double left-clicking on the waveform ruler or the Begin time window. In addition, right-clicking on the horizontal time waveform ruler displays a pop-up window that will allow you to select between formats.

Decimal (mm:ss:ddd)

This display type numerically indicates time in an mm:ss:ddd (minutes, seconds, decimal) format.

SMPTE Drop

If a time code reader set up to read the monochrome rate of 30 fr/second were to be used to read a videotape that was time encoded with the standard frame rate of 29.97 fr/second, the time code readout would pick up an extra .03 frame for every second that has passed. Over the course of an hour, the readout will differ from the actual tape address by a total of 3.6 seconds (or 108 frames.)

In order to correct for this discrepancy (such that the time code readout for color video and the actual elapsed time would be in agreement), a means of frame adjustment was introduced. Because the object is to drop 108 seconds over the course of an hour, the code used for color has come to be known as SMPTE drop-frame. Using this time code system, two frame counts for every minute of operation are omitted, with the exception of minutes 00, 10, 20, 30, 40, and 50. This has the effect of adjusting the frame count to agree with the actual elapsed program duration.

In addition to the color 29.97 drop-frame code, a 29.97 non-drop-frame color standard is also commonly used in video production. When using this non-drop time code, the frame count always advances one count per frame, with no drops in the count. As you might suspect, this results in a disagreement between the frame count and the actual clock-on-the-wall time over the course of the program. Because no frame compensations have to be taken into account for dropped frames, the non-drop mode has the distinct advantage of being easier to calculate whenever time adjustments are required during the video editing process.

Samples

This display type indicates time numerically using the actual number of samples that have passed since the beginning of the edited file as a reference.

Bars and Beats

This displays time in a musical measures format of bar:beat:ticks per beat. You can adjust the number of ticks per beat, tempo, and other properties in **/Edit/Edit Tempo**.

Custom (XX frames/sec)

This option will default to the timecode setting that was last selected in the Define Custom sub-menu. For example, if a custom timecode setting of 30 frames/sec (non-drop) was chosen in a previous session, this option would show as **Custom (30 frames/sec)**. However, if 25 frames/sec (EBU - the standard European frame rate standard) was chosen this option would show as **Custom (25 frames/sec)**.

Edit Tempo

The Edit Tempo window allows you to adjust the settings for the Bars and Beats format.

Define Custom

This option allows you to select a custom frame rate for use as a timeline display reference (with the most common being 30 frames/sec (non-drop) and 25 frames/sec (EBU - the standard European frame rate standard), although other rates may be encountered within computer-based multimedia production.

Vertical Scale Format

Lets you choose between several vertical scale formats for use in the y-axis (vertical) ruler, including Sample Values, Normalized Values, Percentage, and Hz (Hertz).

As with the Display Time Format, you can select this window by choosing **/View/Display Scale** Format from the menu, or you can selectively cycle through the above time formats by double left-clicking on the waveform ruler. In addition, right-clicking on the vertical scale waveform ruler displays a pop-up window that will let you select between formats.

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

Sample values

This display type numerically indicates amplitude as the data's exact sample value of the data.

Normalized Values

This display type numerically indicates amplitude on a normalized scale value that ranges from -1 to 1.

Percentage

This display type numerically indicates amplitude on a percentage scale value that ranges from -100% to 100%.

Hz

When the display is in the Spectral View mode, the vertical ruler is always in frequency (Hz) format. In this mode, lower frequencies are displayed near the bottom of the display, while higher frequencies are displayed from the middle-to-top part of the scale.

Info (Alt+I)

You can embed extra user information in Windows .wav files using the RIFF LIST INFO and DISP type 1 formats. Store summary information such as who played what, who was the engineer, etc. Provided that other audio editors support this information, this information remains with your audio file throughout its lifetime. Be sure to enter the appropriate information here!

Display Title

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

Original Artist

The one who created the sound initially. Examples are: Beatles, Fred Flintstone... or you!

Name

The title of the wave. This is your chance to put a name with your audio "artwork.

Genre

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

Key Words

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

Digitization Source

Where was the sound digitized from. A tape deck, CD, or maybe directly from a microphone? You might want to describe the sound card used here, too.

Original Medium

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

Engineers

This field is used to store the name(s) of the engineer(s) who worked on the file, or edited the file. Please separate names by a semicolon and a space. When a new person edits the file, they can add their name to the list.

Digitizer

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

Source Supplier

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

Copyright

Any copyright information for this file should go here. Example: (c)1997 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

Fill Fields Automatically

If the "Fill * Fields Automatically" box is checked, the Software Package and Creation Date fields are automatically filled by *Cool Edit Pro*. If you don't want this extra information to be tagged with your wave files, simply un-check this box.

Comments

This is for making any comments you wish. 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 don't use any line returns. End each sentence with a period. For Example: It took 12 hours to get this recording right. John added echoing effects using *Cool Edit* and it's finally ready to be mastered.

Subject

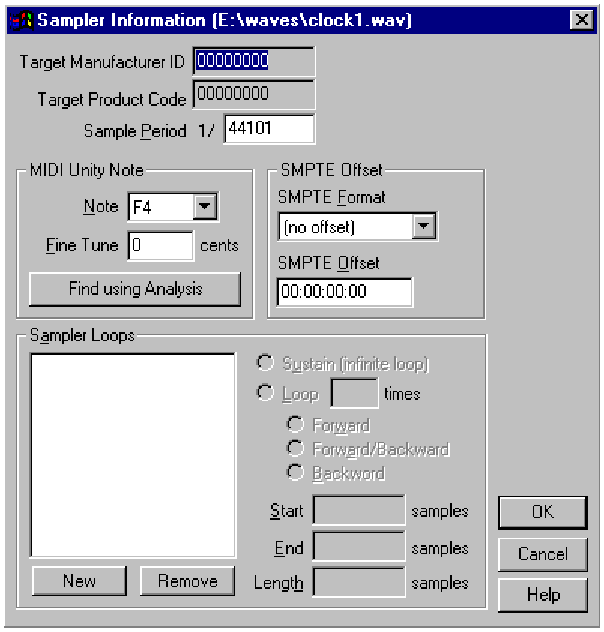
This Describes the content of the file. Feel free to include a description of the instruments used, where someone can find the song recorded, etc. Line returns are OK, and are created by pressing Ctrl+J. Sometimes copyright information is placed here as well. For Example: The shakuhachi of Japan.<Ctrl+J>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

Bitmap

Insert any DIB or BMP bitmap file into this field (preferably a 32 X 32 pixel 16-color file). Windows 95 or NT's Media browser will then use this icon to display a picture that represents the sound. Other OLE compatible applications can also use the above display title, and/or the bitmap to represent your waveform.

Sampler Info

Technical information relating to other devices, systems or programs (such as synthesizer up/downloading software) can be directly imbedded within your .WAV files using the /View/Sampler Info information window. As with the RIFF Info window, this information should stay with your audio file through its lifetime.



Sample Info

Target Manufacture ID and Target Product Code

This device ID isn't supported yet by any known applications, and is defaulted to 0 for this version.

Sample Period

The sample rate (or within 1Hz of it) is automatically placed into this box. You can 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 indicates the "base" or "root" note on a sampler that the current audio file is to be assigned to. The audio file's original pitch will be preserved whenever this key is played on a sampler.

Fine Tune

The actual tone can be entered in values as precise as 1/100th of a cent. Enter the number of cents above the Note that the tone actually is.

Find using Analysis

This function can be used to analyze the audio file, so as to automatically determine the Note and Fine Tune values. If a sampler loop is selected in the Sampler Loops list, the frequency at the center of that loop will be entered into this field 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

This field can be used to enter in the SMPTE frame rate format and SMPTE trigger offset point for the currently loaded sample. For example: a audio file for a film soundtrack that would need to be triggered at 45 minutes, 15 seconds and 29 frames, might have a frame rate setting of 30 fr/sec with an offset of 00:45:14:29.

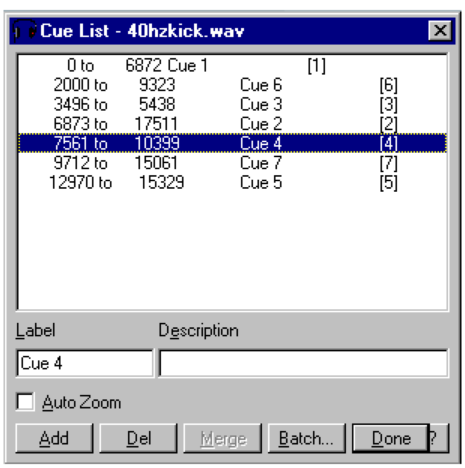
Sampler Loops

New loops can be added by first highlighting the range area, and 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 in 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 infinitely (as with a sustain loop, and the infinite loop would decay once the synth key is released).

Currently this information is only saved in .wav files.

Cue List

Opens the Cue List window. A cue list is a list of time offsets, or locations defined within an audio file. A cue can be either a point, specifying a cursor position, or a range, specifying a selection. You can define and save up to 96 cues in the Cue List for later recall or for assembling in *Cool Edit Pro*'s Play List. The Cue List is a floating window, meaning you can click in the waveform on the main *Cool Edit Pro* window (to define your cues) while leaving the Cue List window on top. You can also jump to a cue position in a waveform by double-clicking on the cue in the list. If the cue is a range, *Cool Edit Pro* will select it automatically. Ranges can later be arranged in the Play List to be played back in any order, with any number of loops.
Cool Edit Pro display Cues in temporal order, with the earliest cue position at the top of the list.



Cue List

Cool Edit Pro displays Cue markers in the main waveform as **red arrows** above and below the wave and Cue ranges as **blue brackets** above and below the waveform.

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

Use the Cue List with the Find Beats and Zero Crossings commands to capture the perfect loop:

1 Find the loop start and end with /Edit/Find Beats/Find Next Beat (Left) and /Edit/Find Beats/Find Next Beat (Right).

2 With your loop selected use /Edit/Zero Crossings/Adjust Left Side to Left to place the left edge at a zero point.

3 With your loop selected use /Edit/Zero Crossings/Adjust Right Side to Right to place the right edge at a zero point.

4 Click Add in the Cue List or simply press F8 (add to Cue List)

You now have a loop transition that's completely smooth with both beginning and end points at zero. By clicking on this selection within the Cue List, you can recall this loop point at any time (as long as the source audio file is currently active within the Waveform window.)

You can also assign any cue range you've added to a key on your keyboard: give the cue range a label of the form KEY N, where "N" is any key on the keyboard (CAPITOL 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.

Label

Lets you enter a short text label describing the selection.

Description

Lets you add a text description of the wave data, if necessary. Also can be used as a comment field.

Add (F8)

Adds the currently highlighted selection, or cursor position to the Cue List. *Cool Edit Pro* displays the items in temporal order, with the earliest cue position at the top of the list. Press F8 when editing a waveform to add the current range or cursor location to the Cue List.

You can press **F8** to automatically add the currently selected position or range to the **Cue List**. Once a range or point has been entered into the Cue List, the **yellow** marker points will be replaced with static **blue** markers. Individual cursor positions can be added to the list and will be displayed as a **red** marker. These cue entries can easily be referenced back to by simply double-clicking on the cue entry within the list.

Don't underestimate the power of the Cue List within the Edit View mode. If a waveform range or cursor position that you know will be of particular interest to you in the near future, simply press **F8**. This will add the selected range to the Cue List. Calling up the Cue List will allow you to recall any range or cursor point within the session, simply by double-clicking on the cue selection.

Del

Removes the selected cue from the list. To select more than one cue item in the list, hold down on the **SHIFT** key for contiguous selection or the **CTRL** key for noncontiguous selection.

Merge

The Merge function is able to take any two selected cue items (whether they are ranges or markers themselves) and merge them into a single, combined cue, spanning the earliest and latest locations in the two cues. To select more than one cue item in the list, hold down on the SHIFT key for contiguous selection or the CTRL key for noncontiguous selection. The new merged item inherits the name from the earliest item chosen (the highest item in the list). You loose the information entered in the Name and Description fields for the subsequent merged items.

Done

Closes the Cue List window.

Play List

Opens *Cool Edit Pro*'s Play List window. The play list is an arrangement of Cue List entries that you can play back in any order and loop a specified number of times. The Play List can include up to 64 entries at any one time. As with the Cue List, the Play List is a floating window meaning you can click in the waveform in the main *Cool Edit Pro* window (to define your cues) while leaving the Play List and Cue List windows on top.

Cool Edit Pro saves the Play List in .WAV file format in the 'plst' chunk.

Add Before

Adds 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

Removes the selected play list item from the list.

Loops

The number of cue ranges that are to be looped within the play list.

Play

Plays the cue ranges in the order listed, looping selections if necessary. Play begins at the currently highlighted item in the Play List. If [end] or if you haven't selected a cue range, *Cool Edit Pro* plays the entire list.

Autocue

Plays the currently highlighted item in the play list (or the first item if nothing is highlighted), and stops on the next item in the play list. Thus, the next item in the play list will be played every time **Autocue** is pressed.

Done

Closes the Play List window.

Let's say that we'd like to re-arrange a musical phrase or spoken word into an entirely different order. No problem! All that we need to do is define the ranges that we'd like to re-arrange as a series of cues within the Cue List (See the above tutorial and info on creating Cue List entries.) The next step would be to call up the Play List window along side the Cue List window, select the Cue that you would like to be played back first, and then press the **<- Add Before** button within the Play List window... Now select the next cue to be entered into the Play List and repeat the steps until you've finished. Once done, simply press **Play** to hear the results. Changes can be made with relative ease, by **Removing** the offending cue and then re-inserting it into the list at the proper position.

Time Window

Displays or hides *Cool Edit Pro*'s Time Window, a resizeable readout of the current cursor location both while idle and in playback/record. *Cool Edit Pro* uses the format for the display (SMPTE drop, Bars and Beats, etc.) currently selected in **/Edit/Display Time Format**. You can make the readout as large as you like for easy viewing from across the room. The window floats, meaning it will stay on top of the rest of *Cool Edit Pro* when open. Close the Time Window by choosing **/Edit/Time Window**, or by right-clicking in the bottom area of the window and choosing **Close**.

Time Window

Transform Menu (Edit View)

The Transform menu displays all the options that relate to waveform transformation (signal processing) functions when in the Edit View mode. These options are:

Invert

This function simply inverts the samples, so that all positive offsets are negative and all negative offsets are positive. Inverting does not produce an audible effect, but it can be useful in lining up amplitude curves when creating loops, or pasting. On stereo waveforms, both channels are inverted.

Reverse

Causes a selection to play backwards by reversing the order of its samples. Useful for creating special effects.

Silence

This option will silence out the selected range. Unlike deleting, or cutting, a selection (which splices the surrounding material together), Silence leaves the duration of the selection intact, and simply zeros the amplitude within it.

DirectX

Cool Edit Pro fully supports third-party plug-in signal processing applications that conform to Microsoft's new DirectX architecture (which makes the plug-ins available to any Windows program on your system that supports DirectX). In plain English, this means that plug-in applications that conform to the DirectX specification can be directly accessed, at any time, by *Cool Edit Pro*. Note: DirectX is also known as ActiveMovie.

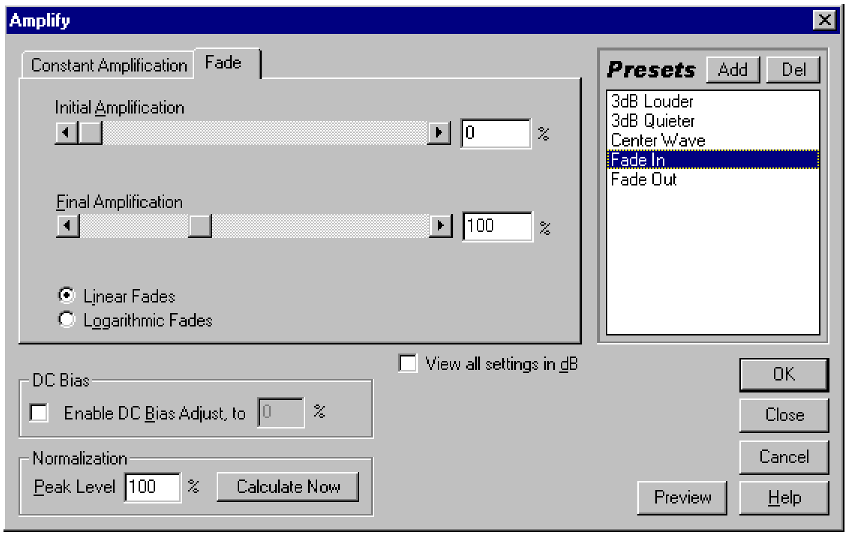
Microsoft's DirectX drivers must be loaded onto your system in order for you to use DirectX filters within *Cool Edit Pro*. *Cool Edit*'s Setup program can install these drivers for you. To install them, insert the *Cool Edit Pro* CD into your CD drive, run Setup, and click on the item marked Microsoft DirectX Media Runtime.

Amplitude Effects

The options in this menu can be used to alter the amplitude (volume level) of a audio file or selected range in any of the following ways:

Amplify

Amplify increases or decreases the volume of a waveform or selection. You can choose between Constant Amplification (in which the same amount of gain change is applied throughout the audio file) or Fade (in which gain varies over the course of the waveform or selection) by clicking on the appropriate tab.



Amplification Dialog

Amplification (Constant Amplification)

Adjust the slider(s) for the amount of amplification or attenuation to be applied to the selection. The signal will remain unchanged whenever an amplification value of 100% or 0dB is selected.

Initial Amplification (Fade)

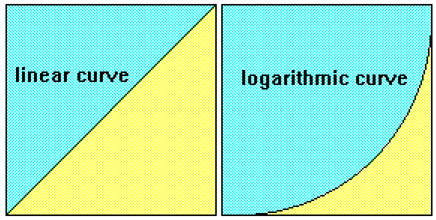
Adjust the slider(s) for the amount of amplification or attenuation that will take place at the beginning of the selection. Adjust the Final Amplification to fade the selection in or out. The signal will remain unchanged whenever an amplification value of 100% or 0dB is selected.

Final Amplification (Fade)

Adjust the slider(s) for the amount of amplification or attenuation that will be applied to the end of a selection. To achieve a fading in effect, choose a Final Amplification greater than that of your Initial Amplification. For fade outs, do the opposite by setting the Final Amplification to the lesser value. Setting both the initial and final amplifications to the same value results in a constant amplification. The signal remains unchanged whenever an amplification value of 100% or 0dB is selected.

Linear/Logarithmic Fades (Fade)

With linear fades, sample values are faded in an even, linear fashion, producing a smooth slope from beginning to end. Logarithmic fades, also known as "Power fades", fade the amplitude of a signal at a constant rate, producing a steeper slope at one end of the fade, depending on whether you are fading in or out.



Lock Left/Right

With stereo waveforms, the Left and Right channels may be individually amplified at separate values. With Lock checked, the scroll bars for the left and right channels move together. Uncheck Lock to adjust the channels separately. Changes in stereo balancing or effects such as panning from left to right can be achieved using this option.

View all Settings in dB

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

DC Bias

Some recording hardware may introduce a DC Bias, which results in the recorded waveform appearing to be above or below the normal center line. Some waveform transformations require that the signal be centered, and choosing Enable DC Bias Adjust (0%) will do just that. To introduce a DC Bias by skewing the entire selected waveform above or below the center line, enter a positive or negative percentage. For example, a setting of 50% moves the entire waveform up halfway, and one of -50% moves it down halfway.

If the amplitude levels between two waveforms don't match up at an edit or loop point, you may hear audible clicking as an edit passes or a loop repeats. To adjusts for this, you can revert to the original source file and use the Zero Crossings feature to match initial and ending levels. If there is a DC offset, this DC level difference can be set to zero by highlighting it and choosing the "Center Wave" preset from the Amplify function. Because centering takes out all frequencies below about 16Hz, it is completely safe to do without any ill side effects.

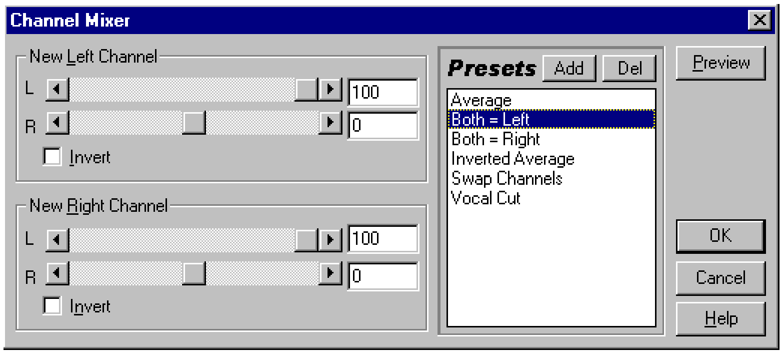
Normalization

Normalization enables you to set a desired Peak Level to which you want a file or selection raised. Use Normalization to achieve the greatest amount of amplification that will not result in clipping (when set to 0dB or 100%.) If the left and right scroll bars aren't locked, the left and right values are computed separately, potentially amplifying one channel more than the other. To normalize to less than the maximum range, enter a negative dB level or the percentage of maximum to which you want to normalize. For example, if you choose 50%, *Cool Edit Pro* 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 that have been normalized to 50% are overlapped, the resulting wave will be not exceed the boundaries, and will not clip.

To normalize in one step, use the Normalize function. The Normalize button only calculates the values needed for the desired normalization. If you are recording a script, only the final values are stored. If you want to add normalization to a script, use the Normalize function instead.

Channel Mixer

The Channel Mixer enables you to alter the stereo image's balance so as to create new stereo mixes using the existing right and left channels as input sources. By recombining and inverting the channels you can create some very cool stereo imagining effects. The default values have been set so that the wave's original gain values will be unchanged.



Channel Mixer

Use the Vocal Cut preset to remove the vocals from stereo recordings. This preset will sum the left channel with the inverse of the right, and place the result into both channels. On music where the vocals are equally loud on both channels, the vocals will disappear, or come close to disappearing. Note that the Vocal Cut preset is ineffective on monophonic recordings and stereo recordings in which the vocals are not in the center of the stereo image.

With different combinations, you can create the effect of swapping channels, which results in a monophonic-sounding wave that is equal to the left channel only, the right channel only, or a mixture of both channels. You can also create waves whose left channel is the inverse of the right.

New Left Channel

Adjust the slide bars to the determine the percentage of the current left and right channels that will be mixed into the new left channel. For example, choose an L of 50 and an R of 50 to sum equally the current L and R channels to the new left channel after mixing. Choose an L of 0, and an R of 100 to make the new left channel contain the material of the original right channel.

New Right Channel

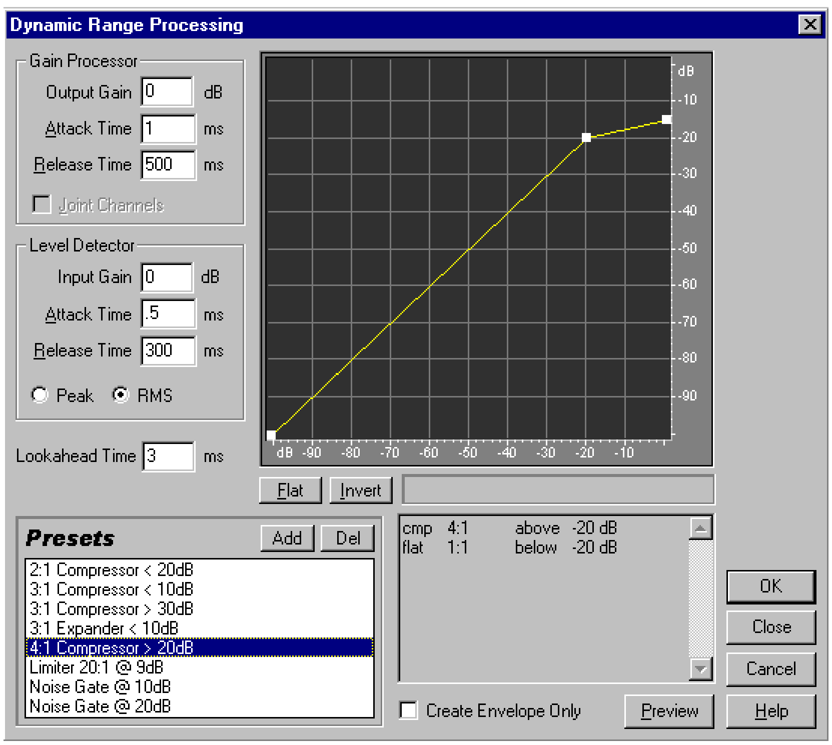
Adjust the slide bars to determine the percentage of the current left and right channels that will be mixed into the new right channel.

Invert

Choose invert for either channel to invert that channel's phase polarity (i.e.: the peaks will become valleys, and valleys will become peaks.) Whenever both channels are inverted, there will be no perceived difference in sound, but inverting only one channel places the channels out-of-phase and greatly changes the sound when played back (whereby the previously in-phase information is canceled out).

Dynamics Processing

The dynamics processor varies the output level of a waveform, based on the input level. This lets you limit or compress the dynamic range of a sample so that the perceived loudness is kept below a defined limit, or so that the waveform's overall dynamic range is kept at roughly the same level. You can also expand or gate the signal so that low-level signals are reduced in level, thereby increasing the perceived dynamic range, or so that signals that fall below a certain threshold [i.e. noise] are eliminated. This is all accomplished by use of a transfer function that is drawn using a 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 linear line that flows from lower-left to upper-right (default) depicts a signal that has been left unchanged, since every input value goes to the exact matching output value. For the not-so-faint-at-heart, Weird transfer functions can also be drawn. For example, you can boost all input that has a level of around -20dB, and leave everything else unchanged. You can also draw an inverse line (a line from upper-left to lower-right) that will dramatically boost low amplitudes while dramatically suppressing high amplitudes (that is, all quiet sounds will be loud, and all loud sounds will be quiet.)



Dynamics Dialog

Compressors are used for purpose of reducing 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 that fall above the threshold. For example, 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.) Compressors can also be useful to compensate for the wide variations in the signal level produced by a singer who moves frequently or has an erratic dynamic range.

Limiters are compressors with a compression ratio of 10:1. This has the effect of reducing or "limiting" input signals that exceed a specified threshold level, so that the output will not increase in gain beyond that point. In other words, a limiter will only allow the dynamic range at its input to increase up to a certain point (determined by the threshold setting), beyond this level, as the input continues to increase in gain, the output level will remain relatively constant and will not increase in volume.

Expanders are used to expand the dynamic range of an audio signal (basically, they're the opposite of a 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 or eliminate noise below a threshold level. It does this by heavily attenuating signals with levels that fall below the threshold. It's often used to totally cut off the signal level during a musical pause so as not to pass background noise. It can also be used to silence the pauses in speech.

Gain Processor

The Gain Processor section affects the signal before output.

Output Gain

This is a gain applied to the output signal and is the last step performed on the audio.

Attack Time

This is the attack time applied just before output.

Attack time determines the time in milliseconds that it takes for the processed output signal to reach its specified 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, such as a "vibrating" sound, can be heard at around 1000 Hz/ms total. For example, 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 as low as you can go without introducing these effects.

Release Time

This is the release time that is applied just before output.

This is the time it takes the end of a previous output level to reach the specified output volume. For example, where the Attack is the time it takes for 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.

Joint Channels

In Stereo, each channel can be compressed independently, sometimes causing the surrounding background noise to get louder on one channel at a time. This can 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 will be amplified together by the same amount (thus preserving the stereo center-channel image). For example, a loud drum beat on the left channel will also cause the right channel to be reduced in level by an equal amount.

Level Detector

This section gets the current amplitude information from the audio to determine the amount by which the original signal will be amplified.

Input Gain

This is the gain added to the signal before it goes into the Level Detector (the section that detects the current level). This essentially "pushes" the graph up or down by the gain given.

Attack Time

This is the attack time that is applied when retrieving the current amplitude information.

Attack time determines the time in milliseconds that it takes for the processed output signal to reach its specified 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, such as a "vibrating" sound, can be heard at around 1000 Hz/ms total. For example, 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 as low as you can go without introducing these effects.

Release Time

This is the release time applied when retrieving the current amplitude information.

Release Time is the time it takes for the end of a previous output level to reach the specified output volume. For example, where the Attack is the time it takes for 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.

Peak

Peak mode is a graph interpretation method that is a little more difficult to use than RMS, and is a bit outdated. It equates to the RMS value times two. That is, if the RMS value is -20dB, then the equivalent peak value will 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.

RMS

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

Low Cutoff

This is the lowest frequency that dynamics processing will affect. You can define a band, or range, to which compression or expansion is applied, within the current frequency range. The Low Cutoff point is the bottom boundary in defining this band. For example, with values of 1000 for Low Cutoff and 5000 for High Cutoff, dynamics processing only affects audio in the frequency range of 1kHz to 5kHz. To use the entire frequency range of the source material, leave this setting at 0.

High Cutoff

This is the highest frequency that dynamics processing will affect. The High Cutoff point is the top boundary in defining this band. For example, with values of 1000 for Low Cutoff and 5000 for High Cutoff, dynamics processing only affects audio in the frequency range of 1kHz to 5kHz. To use the entire frequency range of the source material, leave this setting at 0. To use the entire frequency range of the source material, this setting should be at 1/2 the current sample rate (24000 for 48kHz, 11025 for 22kHz, etc.).

Leaving this setting at the default of 24000 will affect the entire frequency range for all sample rates below 48kHz.

Lookahead Time

Lookahead Time is used to handle sharp spikes that may occur at the onset of a louder signal. At times, and for brief instances, these onsets can go beyond the limits of your compressor settings, which may be desirable in certain compression scenarios since it can enhance the impact of, say, a drum hit. However, this is obviously not desirable if you are using limiting in order to reduce the maximum amplitude of the audio.

The spikes occur because it takes a little time to determine (the Level Detector's attack value) and react (the Gain Processor's attack value) to the current signal level, so Lookahead Time will actually cause the attacks to start before the audio gets loud, instead of right on top of the transient. Otherwise, with a Lookahead Time of 0, a spike will stay loud until all of the attack times have elapsed.

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 that flows directly from the lower-left to the upper-right (default) depicts a signal that has been left untouched, since every input value goes to the exact matching output value. Adjusting the shape of this line will adjust the input or output assignments, thereby altering the dynamic range. For example, you can boost all input that has a level of around -20dB, leaving everything else unchanged. You can also draw an inverse line (a line from upper-left to lower-right) that will dramatically boost low amplitudes while dramatically suppressing high amplitudes (that is, all quiet sounds will be loud, and all loud sounds will be quiet.)

To add a point to the graph, click in the grid at the location where you would like the point placed. To remove points, simply click-on the one to be removed and drag it off the edge of the graph.

To move a point on the graph, click and hold on the point and drag it to a new location. When the mouse cursor is located over a point, you will see it change from an arrow to a hand.

To numerically enter input and output signal levels for an Edit Point, right click on the point to bring up the edit box.

Flat

The Flat button resets the graph to its default state of an unchanged signal, removing all control points.

Invert

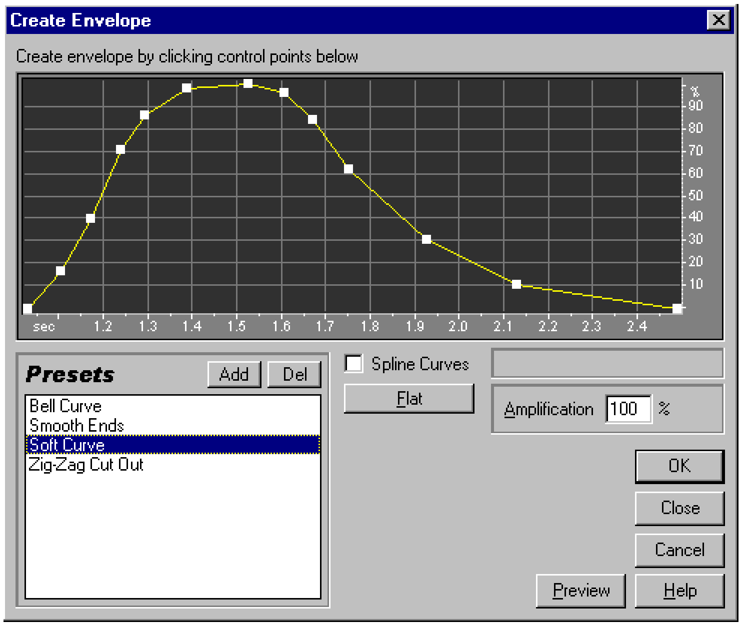
The invert button changes the graph to one that will function as the exact opposite. For example, if a transfer function with a compressor characteristic is being displayed, pressing Invert will change the graph to one with the corresponding expander characteristic. For a graph to be invertable, it must have points in the two default corners (-100,-100 and 0,0) and its output level must always increase from left to right (i.e. each Edit Point must be higher than the one to its left).

Create Envelope

You can use this option to apply any dynamics processing, and have the result returned as an amplitude envelope. This envelope can then be copied and used with /Edit/Mix Paste (modulate) to modulate it with another sound's amplitude. Of course, it is also useful for just seeing what the amplitude envelope of your audio is.

Envelope

Envelopes give you control over which parts of your wave are amplified, and by how much. The top of the graph represents 100% (normal) amplification, the bottom represents full attenuation (silence.) This function is handy when modifying tones that are generated with *Cool Edit Pro*, so as to create more realistic sounding instruments and effects. To add control handles, simply click in the graph area. Once created, you can simply drag control points up and down, or drag a control point off the graph area to remove it.



Graph

The graph depicts time along the x-axis (left and right) and the new output level along the y-axis (up and down), with the yellow line representing amplitude change.

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.

Spline Curves

Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use spline curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

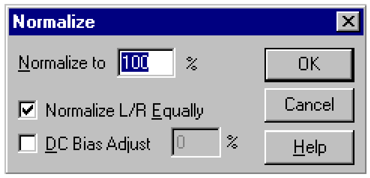
Flat

The Flat button resets the graph to its default state by removing all control points.

This function is useful for "re-drawing" the dynamics (amplitude over time) of a sound. Try loading in a short audio file and actively select the entire audio file as the range (this is done by double-clicking within the Waveform window... the entire audio file should now be highlighted. Now, select **/Transform/Amplitude/Envelope**, select "Soft Curve" from the Preset list and press **OK**. Now, you can listen to the results... Select **/Edit/Undo Envelope** to undo the processing and repeat the steps using the "Zig-Zag Cut Out" Preset option.

Normalize

Amplifies the highlighted selection to within the specified percentage of the maximum level. Use Normalization to achieve the greatest amount of amplification that will not result in clipping (when set to 0dB or 100%).



Normalize Dialog

Use the 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.

Normalize to

Enter the percentage of maximum to which you want to normalize. For example, choose 50% to compute values needed to amplify the selection no more than 50% of maximum (resulting in a 3dB attenuation from maximum output.) Choose 100% (default) to apply the greatest amount of amplification possible without clipping.

Normalize L/R Equally

Check this box to use both channels of a stereo waveform in calculating the amplification amount. When it is unchecked, *Cool Edit Pro* computes the left and right values in a stereo waveform separately, potentially amplifying one channel more than the other.

DC Bias Adjust

Some recording hardware may introduce a DC Bias, which results in the recorded waveform appearing to be above or below the normal center line. Check DC Bias and set it to 0% to center the waveform on the center (zero voltage) line. To skew the entire selected waveform above or below the center line, enter a positive or negative percentage.

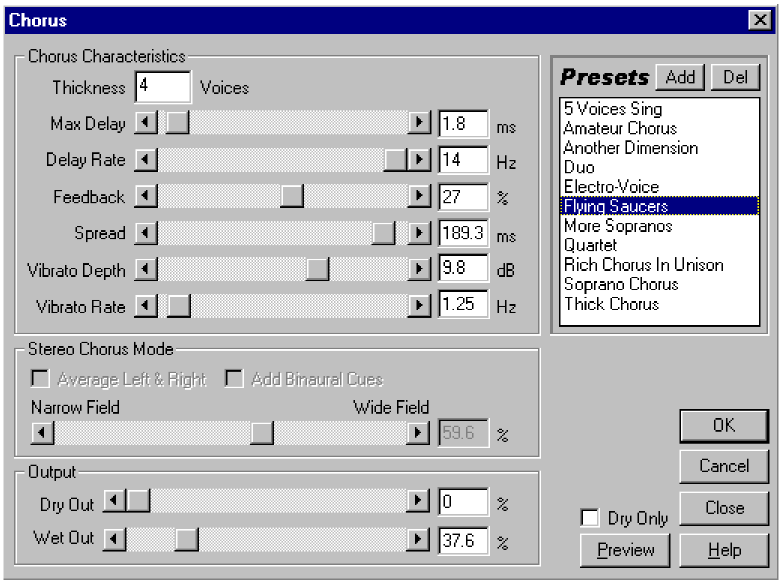
Open an audio file that has a moderate overall level, actively select the second half of the waveform and choose the Normalize option. Now play the entire file back **(Note: You may want to turn your system's volume down, as the normalized section can get pretty loud!)**

Delay Effects

The options in this menu can be used to introduce a series of "repeats" or delayed echoes of the original input signal at regularly spaced or user-defined intervals. A audio file or selected range can be delayed in any of the following ways....... in any of the following ways...... in any of the following ways.......

Chorus

The Chorus effect adds richness to a sound by making it sound as though several voices or instruments are being played at once. Chorus is a great way to add a degree of "presence" to an existing track: you can use it to give a stereo effect to a mono sample (where left and right channels are identical), or to add harmony or "thickness" to a vocal track. You can also create some truly off-the-wall special effects. *Cool Edit Pro* uses a direct-simulation method of achieving the chorus effect, which means that each voice is made to sound distinct from the original by slightly varying the timing, intonation, and vibrato. You can use the Feedback setting to add extra detail to the result.



Chorus

Chorus Characteristics

These are the characteristics used for each layer, or voice, in the chorus. While the properties below apply to each voice, they represent ranges of random values, so each voice will be unique in each of these characteristics.

Thickness (1 to 12): Thickness determines the number of layers or voices that will be simulated in the chorus effect. The final result may end up with an additional voice if the Dry Out setting is used (which mixes the original sample in with the chorused result). The more layers you use, the longer it will take to compute, but the result will have extra richness and fullness.

Max Delay: An important component of chorusing is the introduction of short delays (often in the 35-15 ms range) that vary in duration over time. Use this setting to limit the maximum amount of delay that will ever occur. If the setting is very small, all the voices will start merging into the original, and an unnatural flanging effect may be noticed. If it is set high, everything will start to sound "warbly", rather like a tape that is being eaten by the cassette deck.

Delay Rate: This parameter determines the time taken for the delay to cycle from its zero-to-maximum delay setting. Because the actual delay used varies over time, the pitch of the sample will increase or decrease over time, placing each voice slightly out of tune with the others (which is what gives the effect of a separate voice). For example, a value of 2Hz means the delay used could vary from no delay to the maximum delay and back twice each second (sort of a pitch vibrato at 2 times a second). Please note that this is only a maximum; if you set it to 2Hz, it may only go part way between zero and the maximum delay, and then start cycling back before reaching the maximum. If this setting is very low, the individual voices won't vary much in pitch at all. If it is set high, the voices may vary so quickly that "warbling" effects can be heard.

Feedback: The final mix of chorused voices can be recycled back into the mix, which can give an extra echo or reverb effect.

> Be very careful with this setting. Just a little feedback (less than 10%) can give an extra richness to the effect (and even this depends on the delay and vibrato settings). If too much feedback is used, more traditional feedback will be heard, like loud ringing or other artifacts, and these may get so loud that they will clip and destroying the signal. Sometimes this can be a desired effect, as in the "Flying Saucers" setting, where it generates sounds that are reminiscent of warbly saucers whizzing around your head.

Spread: The spread setting gives an added delay to each voice, separating them in time by as much as 200 milliseconds (1/5th of a second). High spread values will cause the separate voices to start at different times – the higher the value, the further apart the onset of each word may be for example. With low values, the effect is of all the voices singing in unison. Depending on other settings, low values may also bring out some flanging effects, which may be undesirable if your goal is a realistic chorus effect.

Vibrato Depth: Another property that varies with each voice is vibrato, which describes how the amplitude varies over time. This value determines the maximum variation in amplitude that will occur. For example, a value of 5dB may alter a chorused voice by varying its amplitude by as much as 5dB louder or quieter than the original. If this setting is extremely low (less than 1dB) the vibrato may be unnoticeable unless the Vibrato Rate has been set extremely high. If vibrato depth is set too high, however, the sound may cut in and out, creating an objectionable warble. Natural vibratos occur around 2dB to 5dB. Please note that this is a maximum depth -- just as in the delay settings, the vibrato volume may not go as low as the setting indicates at times. This limitation is intentional, as it gives a more natural feel to the effect.

Vibrato Rate: The Vibrato Rate determines the maximum rate at which vibrato will occur. With very low values, instead of a vibrato effect, the resulting voice will slowly get louder and quieter, like a singer that cannot keep his or her breath steady. If the vibrato rate is set high, the result can be very unnatural-- more like a singer who had too much coffee. With very high settings, you can achieve interesting special effects (as in the preset "Another Dimension").

Stereo Chorus Mode

These settings (which are only active when working with stereo files) determine where the individual voices will be placed in the stereo field, as well as how the original stereo signal will be interpreted. For greatest effect, convert mono files to stereo first before applying the chorus effect.

Average Left and Right: *Cool Edit Pro* can either average the original left and right channels or keep them separate in order to preserve any stereo image that may already exist. If you leave the box unchecked, when processing a stereo source file, the stereo image will be preserved (for example, spatial binaural cues such as those that exist in reverberated audio or live stereo recordings will be preserved.) Note that there is no need to check this box if the sample was originally monophonic, as it will have no result, except that if the box is checked, *Cool Edit Pro* will process the sample more slowly.

Add Binaural Cues: Check this option to add separate delays to the left and right outputs of each voice. This can make each voice appear to be coming from a different direction, but only when played through headphones. If the audio is meant to be played through speakers, this option should be turned off. In addition, when you add binaural cues, the volume of the right channel for a voice panned all the way to the left is still significant, whereas if no cues are added, no output would be sent to the right channel. This is why greater separation can be heard when listening through speakers if binaural cues are not used.

Stereo Field Settings: Stereo field denotes where in space instruments or other sources are placed within the L/R image of a stereo waveform. A waveform that contains sound sources that are spread widely from left to right is said to have a wide stereo field, while those having images that are centered around the middle are said to have a narrow stereo field. In the case of *Cool Edit Pro*'s Stereo Field Settings, the narrower the stereo field, the greater the chance that the chorused voices will be placed in the center of the L/R stereo image. At a setting of 50%, all the voices will be spaced evenly about a half circle from left to right. For example, if a thickness of 5 is used, each voice will be panned such that voice #1 will be to your left, voice #2 will be left-of-center, voice #3 will appear in the center, voice #4 will be right-of-center and voice #5 will be on your right. At settings higher than 50%, the voices start migrating to the outer edges: voices to the left go farther to the left, and those to the right go farther right. If you use an odd number of voices, then there will always be one voice directly in the center. With an even number of voices, and very high stereo field settings, all voices are pushed either hard left or hard right.

Output

The final output can be a mix between the original input (dry) signal and the chorused (wet) signal. Ordinarily, both settings should be less than 100%; otherwise the overlaying of several voices may cause clipping.

Dry Out: This setting determines how much of the **un**processed signal is mixed into the final output. If you set it to zero, *Cool Edit Pro* will add the original voice to the number of processed voices (which is determined by the Thickness setting).

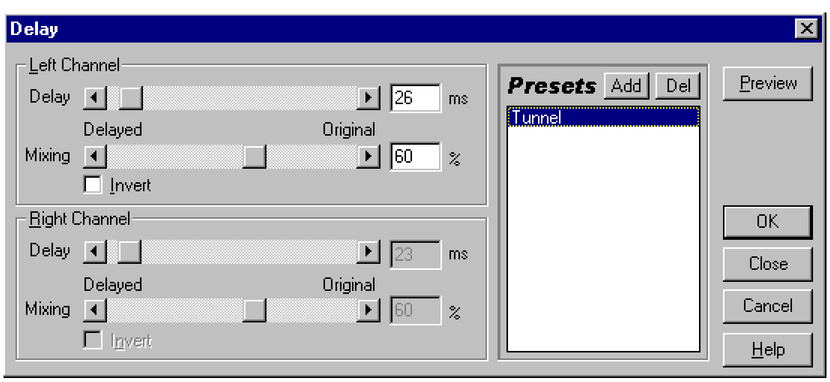
You can keep the original signal near 100% and reduce the Wet Out to give the singer or instrument a "backup chorus". Reduce the blend to 30% or so to blend the original signal with the processed chorus. You can also use *Cool Edit Pro*'s multitrack mixing capabilities to dynamically bring in and fade out the chorus: chorus a copy of the original audio and set this value to zero to create a chorus-only version of the original. In the mixer, insert both the original and the fully chorused versions. Use the volume envelope control to adjust the volume of the chorus over time, or just tweak the final amplitude of the background chorus with the track's volume settings. This technique can be handy for emphasizing certain portions of the singing with a backup chorus.

Wet Out: This setting determines how much of the processed signal is mixed into the final output. Generally, this value should be set lower whenever more voices are used. For example, with a Thickness setting of 3, a setting of 40% would be appropriate, but with Thickness at 10, 20% might be better. The best value will vary depending on the number of voices, and the desired stereo image field settings.

Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Delay Effects/Chorus** option and select the "5 Voices Sing" preset and choose **OK**. Now sit back and be amazed!

Delay

Delay can be used to create single echoes, as well as a number of other effects. Delays of 35 milliseconds (ms) or more will be perceived as discrete echoes, while those falling within the 35-15 ms range can be used to create a simple chorus or flanging effect. (These effects won't be as effective as the actual chorus or flanging effects within the transform menu, as the delay settings will be fixed and won't change over time.) By reducing the delay times further down into the 15-1 ms range, you can spatially locate a mono sound (same information for both left and right) so that it appears as though the sound is coming from the left or the right side, even though the actual volume levels for left and right of the wave are identical.



Delay

Delay (0 to 500 ms)

Adjust the slider to determine the actual amount of time to delay the channel in question.

Mixing

Adjust the slider to set the amount of delayed signal (wet) and unprocessed (dry) to be mixed into the final output. A value of 50 will mix the two evenly.

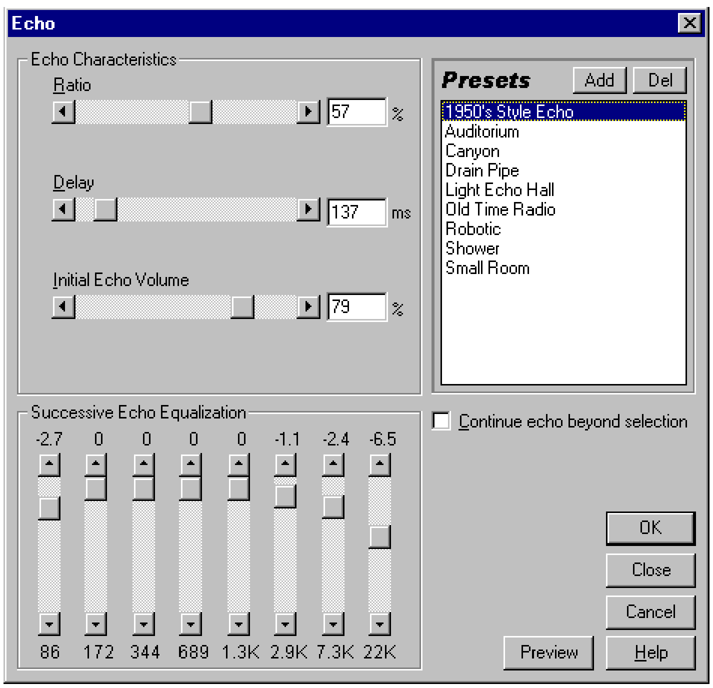
Invert

If this is checked, the delayed signal is inverted. Inverting the delayed signal can be used for special effects, such as creating a quick-and-dirty comb filter.

Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Delay Effects/Delay** option and select the "Spatial Echo" preset and choose **OK**. Now sit back and be amazed!

Echo… echo… o…

This function adds a series of repeated, decaying echoes to a sound (for a single echo, use the Delay function instead). You can achieve effects ranging from a Grand Canyon Hello-ello-llo-lo-o to "boingy" drain pipe sounds by varying the delay amount. You can also change a room's characteristic sound from one with reflective surfaces (yielding echoes that have a bright, shiny high-end) to one that is almost totally absorptive (very little reflected high-end sounds) by equalizing the delays. Note that you can create striking stereo echo effects by using different Left and Right channel values for the Ration, Delay, and Initial Echo Volume controls.



Echo

Decay

Decay is the falloff ratio. Each successive echo will trail off at a certain percentage less than the previous one. A decay setting of zero results in no echo at all, while a decay of 100 produces an echo that never gets quieter.

Delay

This is the number of milliseconds that is placed between each echo. For example, a setting of 100 milliseconds results in a 1/10th-second delay between successive echoes.

Initial echo volume

Adjust the slider to set the amount of echoed signal (wet) to be mixed with the original (dry) signal in the final output.

Lock Left/Right

With stereo waveforms, the Left and Right channels may be processed separately. When Lock is checked, the scroll bars for the left and right channels move together, maintaining the same settings for each channel. Uncheck Lock to adjust the channels separately.

Echo Bounce

Select this option to make the echoes bounce back and forth between the left and right channels. If you want to create one echo that bounces back and forth, select an initial echo volume of 100% for the one side (left or right), and 0% for the other. Otherwise, the settings for each channel will bounce to the other, creating two sets of echoes on each channel.

Successive Echo Equalization

The echo "quick filter" enables you to choose which frequencies are removed from the echo first. Each successive echo is passed back through the quick filter, allowing for control in simulating the natural absorption of a room. A setting of zero will leave the frequency band unchanged.

Continue beyond selection

When this box is checked, the echo effect continues beyond the right-hand boundary of the selected range. Sounds that extend beyond this boundary will not be effected. In other words, the selected range will be echoed, and this echo will naturally decay beyond the selection boundary, while unselected sounds will not be echoed.

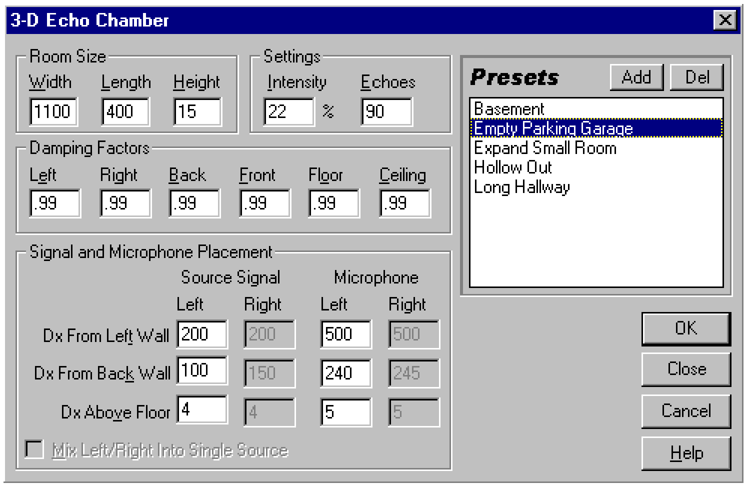
The echo effect will stop at the right-most waveform boundary visible in the waveform window. So if the window is zoomed in, the echoing will stop before the range or file ends.

Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Delay Effects/Chorus** option and select the "Drain Pipe" preset and choose **OK**. Now sit back and be amazed....amazed....amazed.... !

3D Echo Chamber

The 3D Echo Chamber function can create almost any room ambiance, with settings for specifying room size and surface characteristics, along with the placement of microphones within the virtual room. The number of echoes is adjustable, up to about 25,000. The more echoes there are to calculate, the longer it will take for the function to complete.

One great use for this function is to convert Mono audio to Stereo by adding ambiance. Choosing a "left" microphone placement that is one or two feet away from the "right" microphone will simulate a listener's ears and will give the effect of "being there" (try monitoring this with headphones for best effect). Use **/Edit/Convert Sample Type** to split the mono signal into two channels so that you can choose 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 if you were playing them through speakers in real life. For example, if your stereo speakers are 6 feet apart, try placing the left and right microphone settings 20 or 30 feet apart.



3-D Echo Chamber

Room Size

The length, width, and height of the room can be entered in feet (there are approximately 0.3 meters per foot). When entering source and microphone locations, they must lie between zero and the room's width for the "Distance from Left" parameter, and between zero and the room's length for the "Distance from Back" parameter. Room sizes can be as large as you like, but memory requirements will grow as room size increases, so you may run out of memory if you use very large room sizes.

Intensity

This determines the amplitude of the direct (original) signal. Because echoes (picked up by the "microphone") add to the signal's overall amplitude, you should set the intensity to something less than 100% to avoid clipping. In fact, the more echoes there are, the lower this value should be set so as to prevent clipping. For example, a setting of about 30% would be appropriate for 100 echoes, whereas 15% would be better for 1,000 echoes.

Echoes

This is the number of actual echoes to produce. To achieve a nice Reverb and ambiance effect, use at least 300 echoes. The more echoes that are generated, the "truer" the result will sound, but larger numbers of echoes require more processing time.

In order to reduce wait time when testing out a chamber's size and overall room sound, you may want to generate only 100 or so echoes. Once you achieve the sound you want, you can undo the test effect and increase the number dramatically for the final production. Up to 25,000 echoes can be generated, depending on the virtual room size and available memory.

Damping Factors

Use damping factors to describe the type of room in which the audio is being played. These 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, but in this simulation all frequencies are reflected equally. For example, the fact that cement reflects high frequencies better than low ones isn't accounted for, but great effects can still be achieved (and they're much more realistic than those of basic Echo functions). The effects of speaker placement enhancing or canceling certain frequencies, though, is still accurate. A damping factor of 1.0 is the highest, simulating total reflectivity, while a factor of 0.0 is the lowest, which represents complete sound absorption (no reflecting surface).

To give more control over the environment, damping 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 (rather 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 other objects in the room, which effectively absorb some of the audio.

Source Signal Placement

The source (originally highlighted audio range) can be placed anywhere in the room, with settings for distance from the back wall, left wall, and floor. The signal will then simulate a single, non-directional point source, meaning that the sound will radiate outwards in all directions. The distance between the source and the walls affects which frequencies are enhanced and is crucial to the overall ambient effect. When using a stereo source, each channel can be placed independently of each other.

Use the Distance values in conjunction with Room Size values to determine exact 3D placement. If you use a values greater than the dimensions of the room (a Dx Above Floor of 21 ft. in a room with a Height of 20 ft., for example), Cool E will use the greatest possible value (20 ft. in this case).

Microphone Placement

You can place up to two virtual microphone pickups in the room. Stereo signals will have two pick up microphones, while mono signals will have only one. Each microphone is routed to its own respective channel of a stereo audio file. The resulting echoes are exactly what the microphone would pick up if it were in the room at the specified location. Place the microphones in a stereo setting one foot apart to simulate human ears. The placement of and distance between the microphones gives the brain cues about the directions of each echo and the size of the room. Try listening with headphones to microphones that have been placed far apart; this gives a very large "aural" or "Spacey" feeling to the sound.

Always place the microphone(s) sufficiently far away 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 will hear the sound only (due to its loudness).

Mix Left/Right Into Single Source

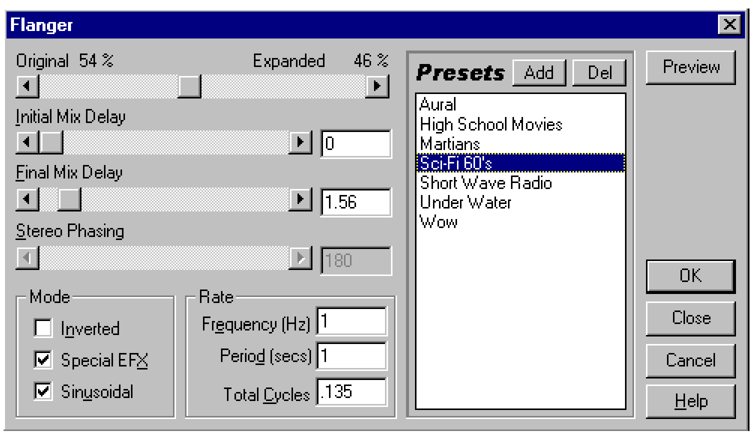
When working with stereo audio, there are actually two source signals (left and right) that can be placed independently. In most instances, the stereo effect will be dramatically enhanced and sound fuller and often richer when the Combine L&R is turned off. This does, however, require twice as many calculations as a single audio source, so for faster processing, check this option this option will sum the left and right channels of a stereo source file before routing to the effects processor.

= unchecked = checked

Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Delay Effects/Echo Chamber** option and select the "Empty Parking Garage" preset and choose **OK**. Now sit back and be amazed!

Flanger

Flanging was originally achieved by sending an audio signal to two reel-to-reel tape recorders, and then physically slowing down the reels of one machine. The resulting sound is a phase-shifted, time delay effect characteristic of the psychedelic sounding recordings in of the 60's. *Cool Edit Pro* can be used to create the same effect by slightly delaying and phasing a signal at predetermined or random intervals.



Flanger

Original - Delayed Slider

Use this slide control to adjust the mix between the amount of original (dry) and flanged (wet) signal. If the Original is at 100%, the result is no flanging at all. If the delayed signal is at 100%, the result is a wavering sound, rather like a bad tape player. Portions of both signals must be present if you want to achieve the characteristic cancellation and reinforcement that occurs during flanging.

Initial Mix Delay

This sets the point in milliseconds at which flanging will start behind the original. The flanging effect occurs by cycling over time from an initial delay setting to a second (or final) delay setting.

Final Mix Delay

This sets the point in milliseconds at which flanging will end behind the original. It refers to the flange effect's secondary (or final) delay setting , with the flange being cycled between this and the initial delay at a specified rate.

Stereo Phasing

This control lets you set the left and right delays at separate values. For example, a setting of 180 puts the right channel at the initial delay value at the same time that the left channel is at the final delay value. This setting can reverse the initial/final delay settings for the left and right channels, thus creating a circular, psychedelic effect.

Inverted

Check this option to invert the delayed signal when flanging. This causes the waves to cancel out periodically, instead of reinforcing the signal. If the Original - Delayed mix settings are set at 50/50, the waves will cancel out to silence whenever the delay is at zero.

Special EFX

This effect mixes both normal and inverted flanging effect, with the delayed signal being summed and the leading signal being subtracted out.

Sinusoidal

If this option is 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 between delays.

The Invert, Special EFX, and Sinusoidal options give you a lot of control over the flanging effect, so try experimenting with them to achieve the effect you want.

Frequency (Hz)

The Frequency, Period, and Cycles settings refer to the rate at which the delay cycles between the initial delay and the final delay. The flanging will cycle (using settings that include frequency times per second, period seconds per complete cycle, or a total of complete cycles) over the entire selection. Different settings can result in widely varying effects. For example, a setting of 0.5 cycles causes the effect to 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.

Period (secs)

The Frequency, Period, and Cycles settings refer to the rate at which the delay cycles between the initial delay and the final delay. The flanging will cycle (using settings that include frequency times per second, period seconds per complete cycle, or a total of complete cycles) over the entire selection. Different settings can result in widely varying effects. For example, a setting of 0.5 cycles causes the effect to 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.

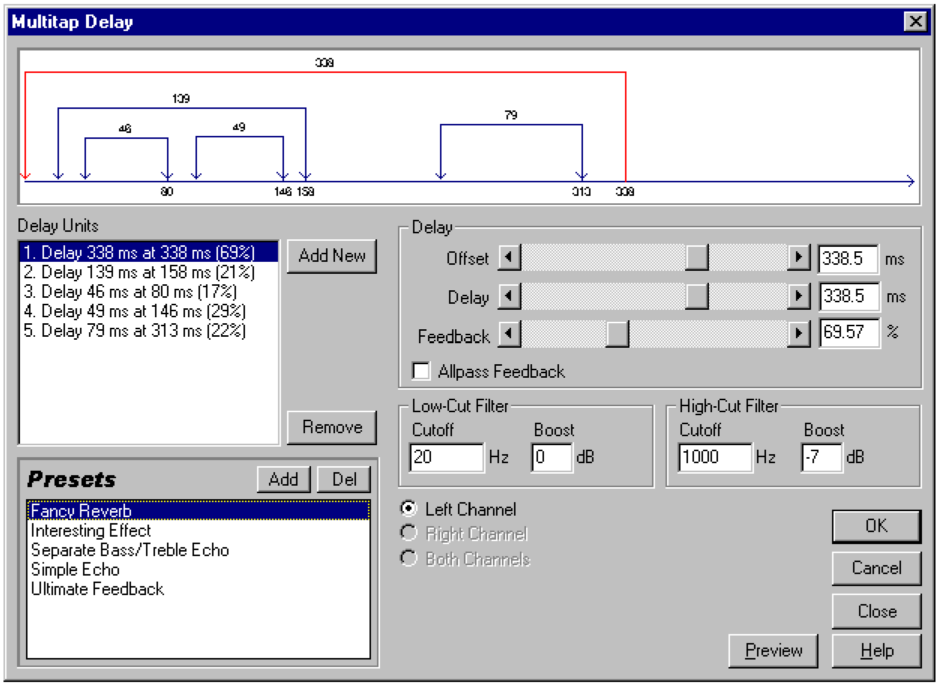
Total Cycles

The Frequency, Period, and Cycles settings refer to the rate at which the delay cycles between the initial delay and the final delay. The flanging will cycle (using settings that include frequency times per second, period seconds per complete cycle, or a total of complete cycles) over the entire selection. Different settings can result in widely varying effects. For example, a setting of 0.5 cycles causes the effect to 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.

Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Delay Effects/Flanger** option and select the "Sci-Fi 60s" preset and choose **OK**. Now sit back and be amazed!

Multitap Delay

Multitap Delay is sort of a combination of Delay, Echo, Filter, and Reverb effects. Up to 10 delay units can be created, each with their own delay, feedback, and filtering settings. If one delay unit is placed inside another (as viewed in the chart above the controls) then the echo will occur more often than just once. As audio travels down the delay line (represented in the chart by the bottom horizontal arrow pointing to the right) portions at any point can be fed back into the delay line anywhere behind the given offset and at any feedback amount, with any high or low cut filter. With experimentation, you can achieve some very interesting effects.



Multitap Delay

Delay Units

This is a numbered list that displays the different Delay Units and their settings in the format: Delay [delay time] at [Offset] ([Feedback percentage]). Click on a delay unit in the list to adjust the Delay Settings sliders for that particular Delay Unit. The Add button creates a new Delay Unit with the current Delay Settings, and Remove removes the currently Delay Unit. You can create a maximum of 10 Delay Units. Each delay unit is represented in the graph above as a back-leading arrow starting at the Offset and going back the number of milliseconds stated under Delay. A single delay unit is much the same as the Echo function, but with a slightly different filtering setup (using two sliding bands with variable cutoff points instead of 8 bands of filtering).

Delay Settings

These are the characteristics used for the delay settings of each delay unit. Click on a different delay unit in the list (or add a new one) to bring up the Delay settings for that particular delay unit.

Offset

This is the point in the delay line from which Cool E will take the audio. It is then mixed into an earlier point in the delay line, which will cause echoing. It is the relative positions of the offsets of the delay units that make a difference, not their absolute position. That is, if you have two delay units at offsets of 200 and 500, the resulting audio will sound the same if they were at 100 and 400, for example. The difference between the offsets is what is important.

Delay

Cool E feeds audio back into the delay line after a certain delay. This setting adjusts that delay, in milliseconds. The result is an echo with a period of the delay given to be generated. With several delay units of varying delays added, the final echo pattern can become very complex. Very short delays give ringing or robotic sounding events, while longer delays give more distinct echoes.

Feedback

This is the amount of signal to feed back into the delay line. If the feedback is set too high, definite ringing and true feedback will occur, where the audio gets louder and louder until it clips and becomes distorted. Sometimes you may want this effect—it is like the feedback you hear when a microphone is set too close to the speaker. If the feedback percentage is extremely low, then not very much of the original signal will be fed back into the loop, resulting in a very subtle effect.

Allpass Feedback

To help prevent the DC component from getting out of hand (the waveform tending upwards or downwards until it clips), try enabling Allpass Feedback. When turned on, audio from the destination of the delay loop is mixed back into audio from the originating delay offset. Instead of going one way (from the offset back a certain number of milliseconds) it also goes from the destination up to the source—a sort of forward feedback, or "feedforward". This setting is handy when designing reverb effects.

Low-Cut and High-Cut Filters

Audio being fed back into the delay line can also be filtered before going back in. With a low-cut filter, the low frequencies are reduced (or boosted) depending on the Cutoff and Boost settings. The High-Cut filter can also cut or even boost the high frequencies. This will make each successive echo filtered slightly differently, for interesting effects. When designing a reverb, it helps to cut some of the high frequencies to simulate absorption of the high frequencies by the surrounding walls.

Cutoff

Frequencies below this setting are affected by the Low-Cut filter, or above this setting for the High-Cut filter. Changes in the cutoff value affect the tone of the echoes, as more or less of the frequencies are affected by the filter.

Boost

This is the amount of filtering to perform. Boost settings are usually negative, which means the audio is being cut or reduced in the affected frequency range. So lower negative values result in more audio being cut. Positive values result in boosted frequencies. Generally, when echoing, frequencies are not boosted, but you can create interesting effects by entering positive values. Boosting a low-cut filter while reducing the feedback setting is identical to reducing a high cut filter and increasing the feedback setting.

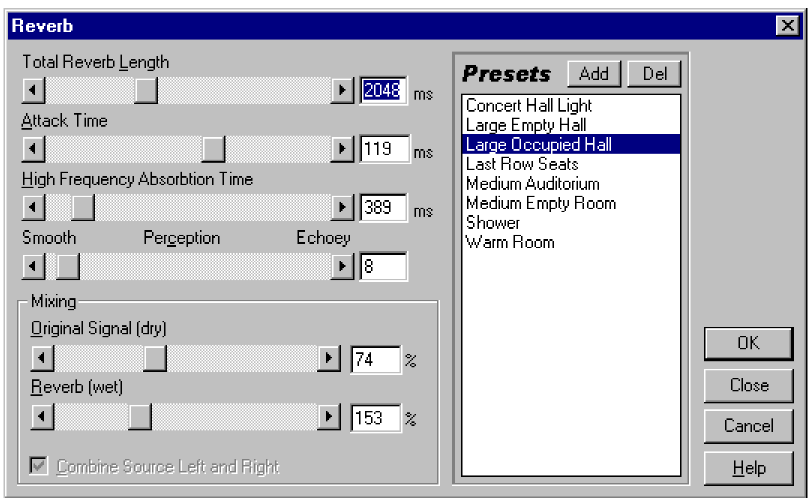
Channels

With a stereo source, each delay unit can be set to affect both channels, or just the left or right channel individually. To spread out the effect in the stereo domain, at least one delay unit in the group should affect the left channel only, and another should affect the right channel only.

Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Delay Effects/Multitap Delay** option and select the "Fancy Echo" preset and choose **OK**. Now sit back and be amazed!

Reverb

Reverb is an effect used to simulate acoustic space, and consists of both early reflections and echoes that are so closely spaced that they are perceived as a single decaying sound. Reverb is different from the basic echo function in that the delays aren't repeated at regularly spaced intervals. *Cool Edit Pro*'s Reverb function can create a wide range of high-quality reverb effects. It can reproduce such acoustic or "ambient" environments as a coat closet, tiled bathroom shower, concert hall or grand amphitheater. These echoes can be spaced so closely together, and made to occur at such random times, that the signal's reverberated "trail" will decay smoothly over time, and will have a warm and natural sound. Alternatively, initial early-reflection delays can be used to give a sense of room size, which can be either large or small, depending upon the initial delay times.



Reverb

To simulate specific rooms that have echoes and reverb, use the Echo function first to get the 'size' of the room sound, and then use Reverb to make it sound more natural. This function can create a sense of spaciousness in a monophonic signal (one that has been recorded as or converted into a stereo audio file.) Even as little as 300ms can open up the perceived spaciousness of a dry sound (one that was recorded without any effects or reverb.)

Total Reverb Length (0 to 6000 ms)

This is the time it takes for the reverb signal to trail off to infinity (about -96dB.) Values below 400 often create a small room environment, while values between 400 and 800 simulate medium sized rooms, and values above 800 simulate concert halls. You can go all the way up to the 3000 ms (3 second) for trails of giant amphitheaters.

Attack Time

The amount of time it takes for the reverb to gain full strength is known as the attack time. For short reverb times, the attack time should be smaller. In general, a value of about 10% of the total reverb time works well; however, you can create interesting and subtle effects by using longer attack times with shorter reverb lengths, or, conversely, by combining very short attack times with long reverb lengths.

High Frequency Absorption Time

In acoustic environments, higher frequencies tend to attenuate faster than lower frequencies. Use this parameter to simulate this natural absorption so that the high frequencies are reduced in level during the reverb's decay time. Faster absorption times simulate rooms that are occupied and have furniture and carpeting, like night clubs or theaters. Slower times (especially over 1000ms) simulate rooms that are emptier, like gymnasiums or auditoriums, where higher frequency reflections are more prevalent.

Perception

This parameter gives subtle qualities to the environment by enabling you to change the characteristics of the reflections that occur within a room. With lower values, the reverb is "smoother" and doesn't have as many distinct echoes. Higher values cause more variation in the reverb amplitudes and add more spaciousness by creating distinct reflections over time. In general, higher values (up to 60%) simulate large rooms, and lower values (down to 0%) simulate smaller rooms.

A setting of 100 and a reverb length of 2000 (2 seconds) or more creates interesting canyon effects.

Mixing - Original Signal

This setting determines how much of the unprocessed (dry) signal is mixed into the final output. To add spaciousness to an instrument, keep the dry signal higher, or at 100%. If you're trying to achieve a special effect with reverb, you might want to reduce the volume of the original signal. If the reverb is so great that audio begins to clip, try reducing both the dry and the reverberated signal 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 work well.

Mixing - Reverb

This setting determines how much of the reverberated (wet) signal is mixed into the final effect output. In modern production, the wet amount should ordinarily be lower than the dry signal to add spaciousness to a track; however, you may want to increase this amount to simulate physical distance from the audio source (where reverb is heard in greater proportions to the original signal).

Combine Source Left and Right

When working with stereo audio, there are actually two source signals (left and right), that can be placed independently. This does, however, require twice as many calculations as a single audio source, so you can check this option for faster processing. When it is checked, Cool E sums the left and right channels of a stereo source file before routing to the effects processor. When unchecked, the original signal and its respective reverbed signal remain in its own channel.

= unchecked = checked

If you know that both channels are identical (that is, if they originated from a monophonic sample), you should definitely check this option to minimize processing time.

Try experimenting with the different parameters to find just the reverb you want. The "Large Occupied Hall" simulates a very nice live theater atmosphere, while the "Concert Hall Light" setting creates a nice professional performance reverb—this preset is especially useful in enhancing a non-reverberated vocal track.

Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Delay Effects/Reverb** option and select the "Large Occupied Hall" preset and choose **OK**. Now sit back and be amazed!

Sweeping Phaser

Like the Flanger effect, a Phaser introduces a variable phase-shift to a split signal and recombines it, creating special effects popularized by guitarists of the 60's. The Sweeping Phaser sweeps a notch or boost type filter back and forth about a center frequency. A phase is similar to a flange, except that instead of using a simple delay, frequencies are phase-shifted over time. When used on stereo files, the stereo image can be dramatically altered to create some truly interesting sounds.

Sweep Gain: Sweep Gain is the gain applied to the phased signal. Take care to avoid clipping when applying higher positive values

Center Frequency: This is the frequency around which the phase will sweep. Frequencies closer to the middle of the dynamic range of the selected audio will produce more dramatic results.

Depth: This setting determines the degree of phasing. The value is given as a Q value, which is a ratio of width to center frequency. Greater Depth settings cause the sweep to extend farther away from the center frequency in both directions (covering a greater frequency range), producing a wider tremolo effect.

Resonance: This is the amount of phase-shift that is applied to the signal. You can think of it as a "strength" setting for the phase. The value is given as a Q value, which is a ratio of width to center frequency.

Sweeping Rate: The Sweeping Rate is the speed at which the filter sweeps around the center frequency, covering the dynamic range specified by the Depth setting. Values are given in Hertz (Hz), or cycles per second, Period (milliseconds), and Tempo (beats per minute). You can adjust the rate with the slider, or enter a value directly into one of the text boxes. To have the sweep occur in time with a song for example, enter the BPM of the music, or a fraction of it (entering 240 for a song with a tempo of 120 would sweep in eighth notes, for example).

Stereo Phase Difference: This is the degree which the sweep interval is shifted between the channels of a stereo waveform. Values farther away from 0 or 360 cause the sweep to occur at increasingly distant intervals between the left and right channels. A value of 180 yields a complete difference.

Sweep Modes: These settings determine the shape of the filter sweep used. Sinusoidal and Triangular determine if the sweep will follow a sine or triangle curve. Triangle curves will tend to be sharper. Log and Linear Sweeps determine if the sweep will be done in Logarithmic, constant fashion, or an even, linear fashion.

Master Gain: This adjusts the overall volume output. You can compensate for loss, or excessive gain, introduced by the effect by entering an overall master gain to be applied to the resulting audio. The default value of 0dB represents no master gain adjustment.

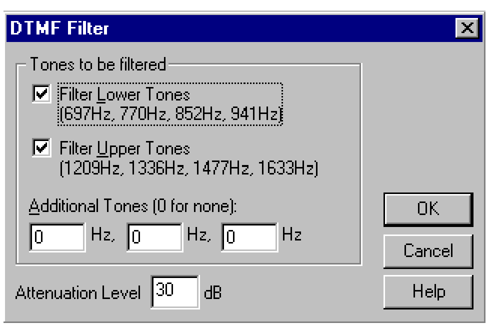
Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect.

Filters

Simply put, filters are used to modify a signal's frequency content. The options in this menu can be used to alter a audio file or selected range's frequency content in any of the following ways:

DTMF Filter

The DTMF Filter uses IIR notch filters to reduce just the frequencies that are present in standard DTMF touch-tone telephones. If your audio signal must be guaranteed not to trigger any DTMF systems (by the system interpreting some of the audio as actual DTMF tones) then one or both of the tone groups can be removed. Generally, only one group would need to be removed to prevent detection of tones by another system, because one tone from each group is necessary to create a DTMF pair. Three additional tones may also be filtered.



DTMF Filter

Filter Lower Tones

Filters 697Hz, 770Hz, 852Hz, and 941Hz.

Filter Upper Tones

Filters 1209Hz, 1336Hz, 1477Hz, and 1633Hz.

Additional Tones

Enter up to three additional frequencies to filter (leave at 0 for none).

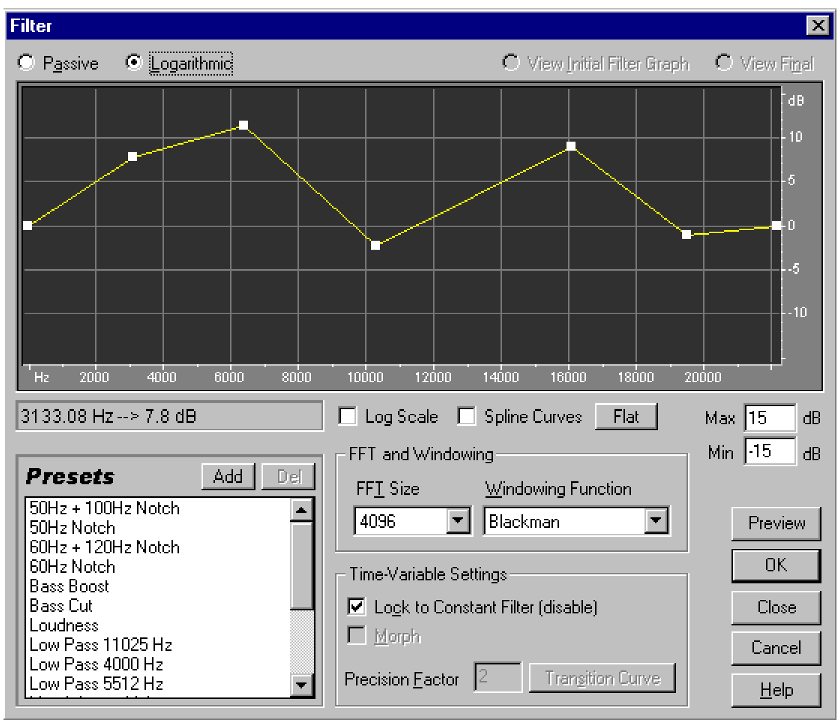
Attenuation Level

The attenuation level determines how much filtering is done. If the level is too high, not only will the tones, but audio surrounding those frequencies as well, will be removed. The trick is to attenuate as little as possible while still obtaining the desired result (filtering of the offending tones so as not to trigger a DTMF detector).

In addition to filtering out dial tone frequencies, you can use the DTMF Filter as a general purpose notch filter by not checking the lower or upper tone groups, and just entering the tones you wish to attenuate (up to three user-definable frequencies). For example, you could use it to remove 60 and 120Hz hum, or any other tones of known frequencies that might creep into the signal.

FFT Filter

The graphic nature of the FFT (Fast Fourier transform) Filter makes it easy to draw curves or notches for rejecting or boosting specific frequencies. Use it to create band-pass filters (which keeps only a certain band of frequencies), high-/low-pass filters (for reducing or eliminating the low or high-end frequencies), narrow-band pass filters (to simulate a telephone pickup) or notch filters (which eliminate very narrow frequency bands).



FFT Filter

The noise level of the FFT Filter is lower than that of 16-bit samples, so when processing audio at 16-bit resolution or lower, there should effectively be no extra noise induced by the FFT Filter (depending on the Window filter being used.) For example, the stop band noise of a Blackman window will be below the -96dB mark.

For best results, filter using 32-bit samples. If your source audio is 8-bit or 16-bit, try converting to 32-bit to do the filtering, and when done, you can convert back to the lower resolution with dithering. This will produce better results than processing at lower resolutions, especially if more than one transform will be performed on the audio.

Passive/Logarithmic

Select Passive mode to express frequency changes (boosts or cuts) in terms of percentages, where a setting of 100% represents no change. Select Logarithmic mode when you want to express changes in terms of dB, where 0dB represents no change.

View Initial Filter Graph/View Final

When the Lock to Constant Filter is not set, you can choose both an Initial and a Final filter setting. Filtering will gradually go from the initial state to the final state. The rate at which the filter migrates from the Initial to Final settings depends on the Transition Curve settings.

- To add a control point to the graph, click in the grid at the location where you want to place the point.
- To enter frequency and amplification for a control point numerically, right click on the point to bring up the edit box, or double-click on the curve.
- To move a point on the graph, click and hold on the point and drag to a new location.
- When the mouse cursor is located over a point, you will see it change from an arrow to a hand.

Log Scale

Check this option to display the x axis (frequency scale) logarithmically rather than in linear fashion. When Log Scale is checked, the graph represents the frequency curve more closely to the way 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 this option.

Spline Curves

Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use spline curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat

The Flat button will reset the curve to its default state of an unchanged signal, removing all Control Points.

Min/Max

These boxes allow you to enter a minimum and maximum value for the graph's y axis. Cool E displays the values in the ruler to the right of the graph. These settings affect the maximum boost or cut that the frequency curve can represent.

FFT Size

The FFT Size parameter specifies the size of the FFT to use, which can affect processing speed and quality. For cleaner sounding filters, use higher values. This value must be a power of two. Recommend values are between 1024 and 8192.

Use lower values (512 or so) when tweaking your filter for faster Preview response. When you have the settings the way you want them, switch to a higher FFT size for better quality when processing.

Windowing Function

The windowing function determines the amount of transition width and ripple cancellation that occurs during filtering, with each one resulting in a different frequency response curve. These range 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. The Hamming and Blackman filters give excellent overall results. Try different windowing functions if you these don't give you the effect you want.

Lock to Constant Filter

Check this option to apply a constant filter to the waveform. Uncheck it to choose both an Initial and a Final filter setting. The rate at which the filter migrates from the Initial to Final settings depends on the Transition Curve settings.

Morph

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 in linear fashion over time. For example, if you leave Morph unchecked and have a spike at 10kHz for the initial filter and a spike at 1kHz for the final filter, the spike at 10kHz will gradually decrease, and the spike at 1K will gradually increase over time, but frequencies between 1kHz and 10kHz will not be affected. If morphing is on, the spike itself will "ooze" from 10kHz down to 1kHz, passing many of the frequencies in between.

For a cool example of morphing, try choosing Passive mode, and set an initial curve with the first half at 100%, and the second half at zero. For the final curve, set the right 1/10th or so at 100% with the rest at zero. 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 include all the frequency combinations between the two filters. To see exactly what's 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.

Precision Factor

This determines how accurately you want to filter over time when separate initial and final settings are used. A larger number (low factor) means the filter settings will change roughly, or in chunks, from initial to final, while smaller numbers (higher factor) will make the transition much smoother. In any case, the higher the precision factor, the longer it will take to filter your selection, but the nicer it will sound. Since the FFT function takes a large group of samples, and filters them all at once, the precision factor determines how many samples from the entire group will actually be saved. A factor of 2 means that half of the samples are saved, while a factor of 10 means that 1/10 of the samples are saved. Since there can be only one filter setting for the entire group of samples, you may want a more accurate (or smaller) setting if the EQ curve varies wildly over short periods of time.

FFT Transition Curve

Displays a graphical representation of the transition from initial to final filter settings. The top graph shows time along the x axis (with the left representing the start of your sample, and the right side representing the end), and where in the transition you are allowed the y axis (with 0% representing your initial filter, and 100% representing your final filter). All points in between are a combination of your initial and final filter arrangements. The readout below the graph displays the current x, y position of your mouse.

Flat: The Flat button will reset the curve to its default state of an unchanged signal, removing all Control Points.

Graph response at point: Check this option to have the bottom graph update in response to your mouse position in the top graph.

The bottom graph shows the filter at any given point in the transition (provided the Graph response option is checked), and corresponds directly to the position of the mouse in the top graph (click, or click and drag to update the view). Watch how the filter settings change as you move your mouse up and down in the top graph—depending on the position you select, 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.

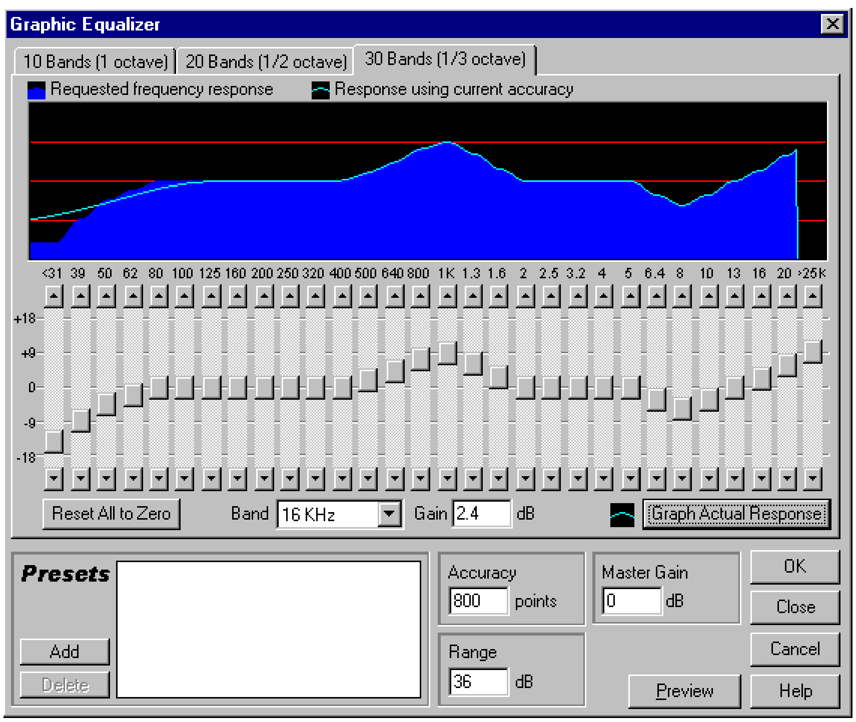
In migrating from the initial to the final filter configuration, the points in the "in-between" settings are just the average between the two settings. For example, a filter setting exactly between the initial and final would be the exact average of the initial and final filter settings.

Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Filters/FFT Filters** option and select the "Telephone Bandpass" preset and choose **OK**. Now sit back and be amazed!

Graphic Equalizer

With the Graphic Equalizer, you can boost or cut the signal at particular frequency bands, and have a visual representation of the overall EQ curve. Unlike the Parametric Equalizer, the Graphic Equalizer uses preset frequency band controls for quick and easy frequency equalization.

Cool Edit Pro's Graphic Equalizer is implemented as an FIR (Finite Impulse Response) filter, which maintains phase errors over the response curve, unlike IIR filters which can have phase error (often audible as a ringing quality). This means that to gain higher accuracy in the lower frequencies, a higher accuracy (Size of the FIR filter) must be used. The bands can be spaced at either 1 octave, ½ octave, or 1/3 octave intervals, with the appropriate Q setting such that if adjacent bands are boosted, the result has no drop outs at intermediate frequencies.



Graphic Equalizer

Bands

The bands are spaced at either 1 octave (10 band), ½ octave (20 band), or 1/3 octave (30 band) intervals. As you might expect, the 10-bands mode offers more general equalizing, while choosing 20 or 30 bands will let you zoom in on specific frequency ranges more precisely.

Reset All to Zero

This sets all sliders to 0dB, so there is no equalization.

Band

You can select the band to be modified from this drop down list.

Gain

The exact value for the gain being used in the chosen band.

Graph Actual Response

Because this is an FIR filter, the response may not actually match the desired equalization curve at lower accuracy levels. Press this button to see the actual response of the equalizer.

Accuracy

Higher accuracy levels (longer FIR filters) will give better frequency response in the lower ranges. Higher values require more processing time, but you can use lower accuracy levels if you only want to equalize higher frequencies. If you are equalizing very low frequencies, you should probably raise the accuracy. Values between 500 and 5000 work well.

Range (4dB to 180dB)

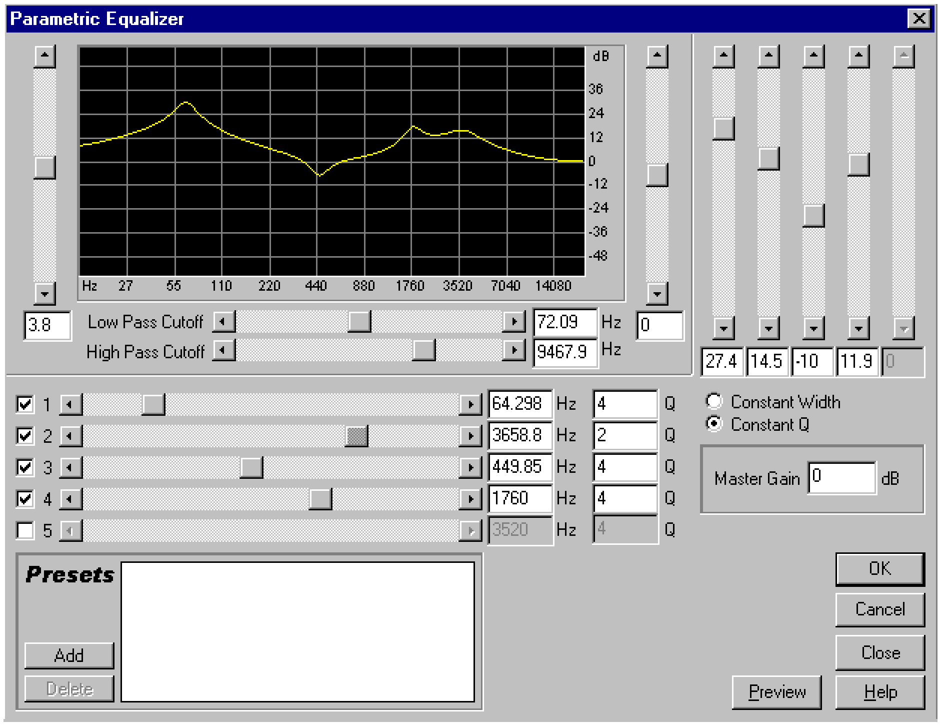
Defines the range of the slider controls. Standard hardware equalizers usually have a range of 30dB to 48dB.

Master Gain

After the EQ settings have been adjusted, the overall volume level might be too loud or too soft. You can compensate for this by entering an overall master gain to be applied to the resulting audio. The default value of 0dB represents no master gain adjustment.

Parametric Equalizer

Parametric equalizers are used for fine editing of frequencies within a sound. Parametric equalizers differs from equalizers that offer a specific number of frequencies and Q bandwidths (which determines how many overall frequencies will be effected by a single, selected frequency control) in that they give you continuous control over the frequency, Q and gain settings. For example, with the parametric equalizer, you can boost a broad range of frequencies that are centered around 1000 Hertz (1kHz), or you can boost the low frequencies while simultaneously inserting a 60 Hz notch filter.
Cool Edit Pro's Parametric Equalizer uses IIR filters (second order), which are very fast and can give very precise resolution, even at the lower frequencies (for example, it's easy to boost just the bass at 40Hz to 45Hz).



Parametric Equalizer

Graph

The graph depicts frequency along the x-axis (left and right) and amplitude along the y-axis (up and down), with the curve representing the amplitude change at specific frequencies. Frequencies in the graph range from lowest to highest in a logarithmic fashion (evenly spaced by octaves).

Low Shelf

The slider directly to the left of the graph determines the Low Shelf amplitude in dB. To increase or decrease the lows (bass) at any time, just adjust the slider, or enter a dB level in the box at the bottom of the slider.

Low Shelf Cutoff

High and low pass filters cut the extreme high- or low-end of the audio spectrum. The low pass (so named because it allows the low frequencies to pass, while reducing the highs) is often used to reduce hiss, amplifier noise, and the like. To fine-tune the range of frequencies being boosted or cut, adjust the Low Shelf Cutoff slider located directly below the graph, or enter a specific frequency in the box to the right of the slider.

Turn on Preview to hear the effect of moving the High/Low Pass sliders.

High Shelf

The slider directly to the right of the graph determines the High Shelf amplitude in dB. To increase or decrease the highs (treble) at any time, just adjust the slider, or enter a dB level in the box at the bottom of the slider.

High Shelf Cutoff

High and low pass filters cut the extreme high- or low-end of the audio spectrum. The high pass (so named because it allows the high frequencies to pass, while reducing the lows) is often used to reduce low-end rumble, hum, or other unwanted low-frequency sounds. To fine-tune the range of frequencies being boosted or cut, adjust the High Shelf Cutoff slider located directly below the graph, or enter a specific frequency in the box to the right of the slider.

Turn on Preview to hear the effect of moving the High/Low Pass sliders.

Intermediate Bands

You can place up to 5 intermediate bands into the EQ circuit. This gives you very fine control over the shape of the equalization curve. Check the slider box to activate the band and its corresponding volume slider. The vertical sliders in the upper right control the amount of boost or cut (you can also enter boost/cut amount (dB) in the entry box below each slider). The horizontal sliders located below the Cutoff sliders control the center frequency at which the boost or cut will occur (a frequency may also be entered in the box to the right of the slider).

Center Frequency

This is the frequency at which boost/cut will occur for a particular band. This number reflects changes made by the band's slider. You can also enter a frequency directly in this box.

Width

The Q or Width value (depending on Constant Q/Width setting below) controls the width of the affected frequency band. With lower Q values (or greater Width values), a larger range of frequencies will be affected. If a Q value is very high (above 100), only a very narrow band will be affected, which is ideal for notch filters where only a particular frequency needs to be removed, like a 60-cycle hum. You should be aware that whenever a very narrow band is boosted, it will tend to "ring" or resonate at the audio at that frequency. Q values of 1 to 10 are used most often for general equalization.

Constant Width/Constant Q

This parameter describes a band's width. This can be a Q value (which is a ratio of width to center frequency) or an absolute width value in Hz. Constant Q is the most common setting, but you may want to use a constant width if ,for example, you want the length of ringing to be a constant, no matter which frequency is being boosted.

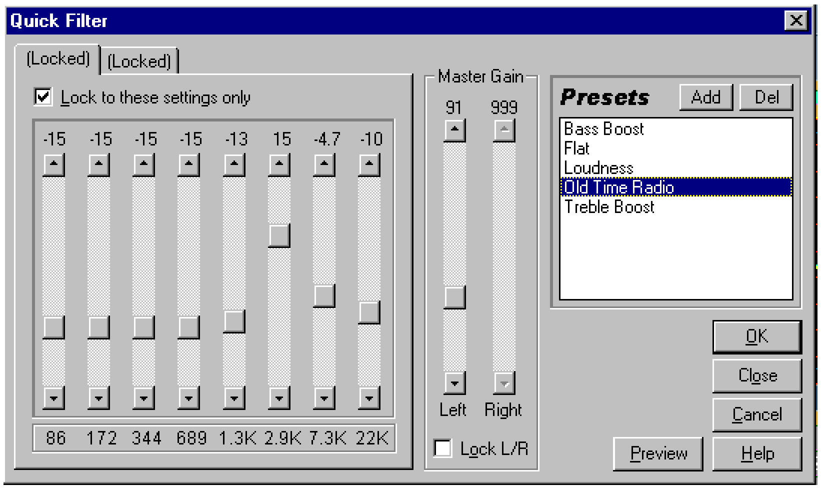
Master Gain

After the EQ settings have been adjusted, the overall volume level might be too loud or too soft. This can be compensated for by entering an overall master gain to be applied to the resultant audio.

Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Filters/Parametric Equalizer** option and activate filters 1, 2, and 3 and then vary the gain sliders for any effect that you'd like and choose **OK**. Now sit back and be amazed!

Quick Filter

The 8-band quick filter can be easily customized to suit most filtering needs. It works much the same as a standard audio equalizer does, except that the bands act a bit differently. The highest frequency band will increase or decrease the high end, but it will also increase frequencies all the way down to the lowest band. It will, however, increase the high frequencies more than the low ones, of course. The effect is close to an equalizer, but not quite. Essentially, this is a quick and easy function for changing the tone of your sample (such as noise) to make it more pleasing to the ears.



Quick Filter

Initial Settings

When Initial/Final isn't locked, choose this tab to select the initial EQ settings.

Final Settings

When Initial/Final isn't locked, choose this tab to select the final EQ settings.

Lock to These Settings Only

Check this box to equalize the entire selected range with the setting shown. Uncheck it to select separate initial and final equalization settings, so that the selection can smoothly glide from the initial equalization setting to the final setting over the selected range. Click on the appropriate tab to change Initial and Final settings.

Equalizer Bands

Adjust these sliders to increase or decrease the frequency component specified beneath the slider. Amplitude is shown above each slider.

Master Gain

This control adjusts the equalizer's overall level.

Lock L/R

With Lock checked, the scroll bars for the left and right channels will move together, maintaining the same settings for each channel. Uncheck Lock to adjust the channels separately.

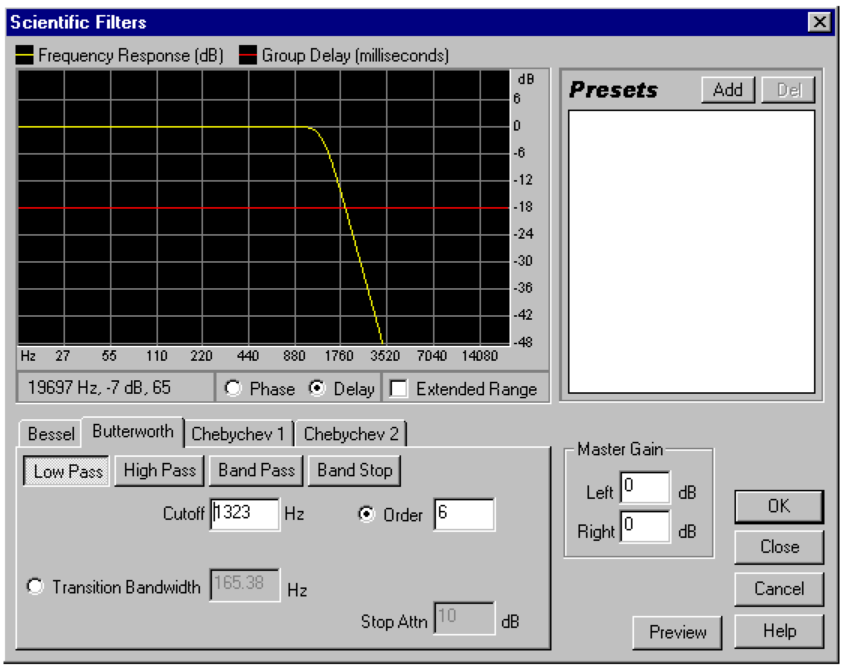
You can create some very interesting effects by changing the initial and final equalization settings so that they can "morph" from one curve type to another (such as having a "bassey" curve change to a flat one at the introduction of a song).

To produce a semi low-pass filter, set the higher frequency scroll bars to -30 to cut out higher frequencies. Similarly, you can create a high pass filter by reducing the lower frequencies.

Boosting the bass frequencies to very high values may lead to clipping.

Scientific Filters

Cool Edit Pro's Scientific filters provide high-order IIR (Infinite Impulse Response) filters for precise band pass, band reject (notch), or high or low pass filtering. The most common types of higher order filters are available: Bessel, Butterworth, Chebychev type 1, and Chebychev type 2. Butterworth is generally the best compromise between quality and desired precision. Please consult any technical reference on digital filtering for more details on these types of filters. In general, each one has different characteristics for how steep the transition bands are (at the cutoff points), and how much the filter attenuates.



Scientific Filters

Low Pass

This setting passes the low frequencies while rejecting (or removing) high frequencies. You must specify a cutoff point in order to determine the starting point at which the frequencies will be passed or rejected.

High Pass

This setting passes high frequencies and removes low frequencies. You must specify a cutoff point in order to determine the starting point at which the frequencies will be passed or rejected.

Band Pass

This mode preserves a range of frequencies (that is, a band) while removing all other frequencies. You must specify two cutoff points to define the edges of the band.

Band Stop (Notch filter)

Also known as a Notch filter, this mode is the opposite of Band Pass, and will reject any frequencies within the specified range. You must specify two cutoff points to define the edges of the band.

Cutoff

This parameter defines the frequency serving as a border between passed and rejected frequencies. It is at this point that the filter will switch from passing to attenuating or vice versa. In filters requiring a range (Band Pass and Band Stop), this serves as the lower frequency, while High Cutoff defines the high frequency border.

High Cutoff

In filters requiring a range (Band Pass and Band Stop), this serves as the higher frequency, while Cutoff defines the lower frequency border.

Order

The higher the order, the more precise the filter (with steeper slopes at the cutoff points, etc.), however very high orders can also have high levels of phase distortion.

Transition Bandwidth

Some filter types allow the specification of the width of the **transition band** (lower values have steeper slopes). If you choose a transition bandwidth, the Order will be filled in automatically, and vice-versa. In filters requiring a range (Band Pass and Band Stop), this serves as the lower frequency transition, while High Width defines the higher frequency transition.

High Width

In filters requiring a range (Band Pass and Band Stop), this serves as the higher frequency transition, while Transition Bandwidth defines the lower frequency transition.

Pass Ripple/Actual Ripple

Some high-order filter types give you the ability to choose the maximum allowable amount of ripple. Ripple is the effect of unwanted boosting and cutting of frequencies near the cutoff point.

Stop Attn

The Stop Band Attenuation control determines how much gain reduction is to be used when removing frequencies.

Master Gain

After the EQ settings have been adjusted, the overall volume level might be too loud or too soft. You can compensate for this by entering an overall master gain to be applied to the resulting audio.

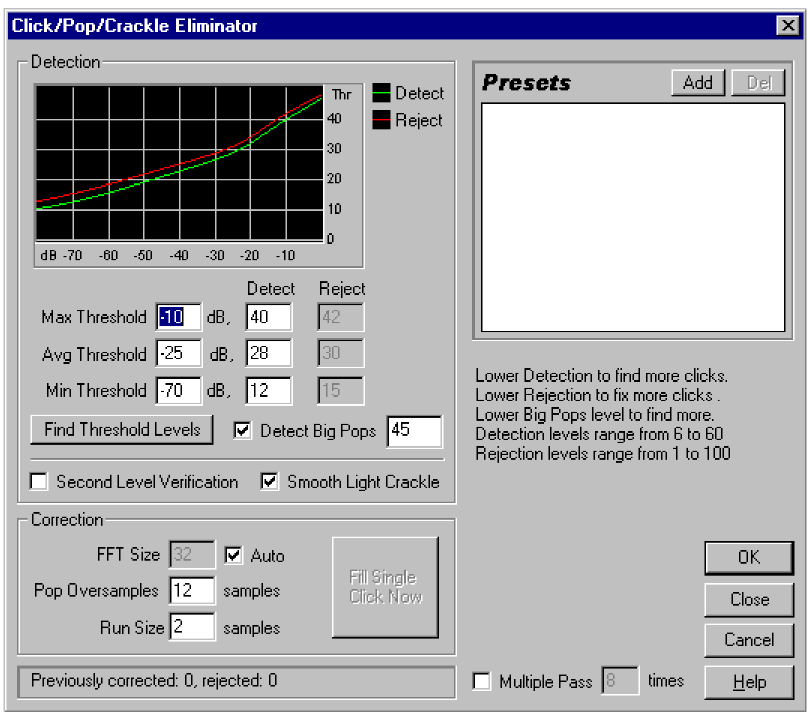
Noise Reduction [See the Manual Addendum for updates!]

The options in this menu can be used to reduce background noise and general broad band noise without having an adverse effect on the overall quality of the originally-recorded signal.

To reduce noise from a file quickly, 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.

Click/Pop Eliminator

The Click/Pop Eliminator works by searching for anomalies in the audio data that could be construed as clicks or pops (Detection), and then replacing or repairing the damaged location (Correction). Using the Click/Pop Eliminator is more accurate than just cutting out the click, or replacing the data with a straight line. You can correct an entire selection or instantly remove a single click if one is highlighted (hint: use a high zoom level to isolate the click). To speed up the repairing of single clicks, configure the **F3** key to correct them when they are highlighted by first choosing the Click/Pop filter and pressing "**Fill Single Click Now**". The **F3** key then repeats the last action without bringing up the dialog box. You can also create a quick key for filling in single clicks from the Favorites menu.



Click/Pop/Crackle Eliminator

Use the Spectral View feature with the spectral resolution set to 256 bands and a Window Width of 40% to see the clicks in a program. See /Options/Settings/Spectral to adjust these parameters. Clicks will ordinarily be visible as bright vertical bars that go all the way from the top to the bottom of the display.

To hear all the clicks that were removed, save a copy of the original file somewhere, then Mix Paste it (overlap it) over the corrected audio with a setting of 100% and Invert enabled.

It may take a little trial and error to find the right settings, but the results are well worth it—much better than searching for and replacing each click individually. The parameters that make the most difference in determining how many clicks are repaired are the Detection and Rejection thresholds (the latter of which requires Second Level Verification). Make adjustments to these will have the greatest effects; you might try settings from 10 for a lot of correction, 50 for very little correction on the detection threshold, or 5 to 40 on the rejection threshold. The next parameter that affects the output most is the Run Size. A setting of about 25 is best for high-quality work. If you have the time, running at least 3 passes will improve the output even more. Each successive pass will be faster than the previous one.

Thresholds Graph

This graph shows the exact threshold levels to be used at each amplitude, with amplitude along the x-axis (left and right) and the threshold level along the y-axis (up and down). Cool E will use values on the curve to the right (above -20dB or so) in processing louder audio, and settings towards the left in processing software sections. There are separate curves for detection and rejection.

Detection Parameters

Threshold Levels (Max, Avg, Min dB): There are three definable levels (Max, Avg, Min; given in dB) for which you can specify unique detection and rejection threshold settings. These levels are ordinarily set to the maximum amplitude, average amplitude, and minimum amplitude of the audio. For example, if your audio has a maximum RMS amplitude of -10dB then you may set the Max Threshold to -10dB. If the minimum RMS amplitude was -55dB then set Min Threshold to -55, etc. Press the Find Threshold Levels button to fill in these levels automatically with the maximum, average, and minimum levels. Generally, less correction is required for louder audio, as the audio itself will mask many of the clicks so that their repair is not necessary. Clicks are very noticeable in very quiet audio, so quiet audio will tend to require lower detection and rejection threshold levels. Once the levels are in place, assign appropriate detection and rejection thresholds to be used at each of the levels.

Detection Threshold (1 to 150): This setting determines how sensitive the filter is in finding clicks and pops, where lower detection thresholds result in more clicks being found. Start with a threshold to 35 for high-amplitude audio (above -15dB), 25 for average amplitudes, and 10 for low-amplitude audio (below -50dB). These settings will find most clicks, and often all of the louder clicks. If there is a constant crackle in the background of your source audio, try lowering the Minimum threshold level more, or increasing the dB level to which the threshold is assigned. This value can be as low as 6, but if the setting is placed any lower, then the filter may removed sounds that are not actually clicks. The more clicks that are detected, the more repair processing there is, which increases the possibility of distortion. With too much distortion of this type, the audio begins to sound flat and lifeless. If this happens, try setting the detection threshold rather low and enabling the Second Level Verification, which will re-analyze the detected clicks and throw out (that is, leave unfiltered) the ones that are less likely to be clicks. Generally, if you still hear clicks after filtering your audio, try lowering the detection threshold, but if this distorts the audio too much, try either increasing the threshold or enabling Second Level Verification.

Rejection Threshold (1 to 150): If Second Level Verification is enabled, the rejection threshold determines how many potential clicks (found using the Detection Threshold) are rejected. As with detection, lower settings result in more clicks being repaired. Increasing the rejection threshold will prevent more clicks from being repaired, as they may not be actual valid clicks. The idea is to reject as many detected clicks as possible, but still remove all audible clicks. A good starting value for this setting is 30. If a trumpet-like horn sound has clicks in it, and the clicks are not being removed, try lowering this value to reject fewer potential clicks. If a particular sound or instrument is "fuzzing out" and getting distorted, then increase the rejection threshold to keep repairs at a minimum (the fewer repairs that are needed to get good results, the better). Cool E may interpret some valid audio as clicks, depending on the nature of the waveform in question. Some waveforms have sharp, albeit periodic, spikes throughout. The rejection algorithm looks to see if this click is isolated (meaning it is a real click) or has neighbors (meaning it is probably part of a desired waveform).

Find Threshold Levels

Find Threshold Levels will automatically fill in the Max, Avg, and Min threshold level settings. Use this button to auto-set the threshold levels, before setting the Max and Min detection and rejection levels (it is a good idea to settle on Max and Min settings first, because once they are in place, they usually need not be adjusted much). After Max and Min, set the Avg level to about three quarters the way up between the min and max settings (for example, Max of 30, Min of 10, then try about 25 for Avg). After auditioning a small piece of repaired audio, you can go back and adjust the settings accordingly. For example, if a quiet part still has a lot of clicks, lower the min thresholds a bit. If a loud piece still has clicks, lower the Avg or Max settings. The loud, average, and quiet portions of the audio are handled differently based on the detection and rejection settings.

Detect Big Pops (30 to 200)

With some audio, large unwanted events (those more than a few hundred samples wide) may not be interpreted as clicks by the normal detection algorithm. Check this option if you want Cool E to remove these large clicks as well. Note that a sharp sound like a loud snare drum hit can have the same characteristic as a very large pop, so you should only enable this option if you know the audio has very large pops (like a recording from a record with a very big scratch in it). If you find that obvious, loud pops are not being fixed, then enable this option, and use settings from about 30 (find kind of quiet pops) to 70 or so (only fix the very loudest pops). If this option is enabled, and drum hits sound softer, then the onset of the drum is being considered a big pop and is being "repaired". If this happens, try increasing the threshold a bit until the obvious pops are fixed.

Second Level Verification

Enable Second Level Verification to start rejecting some of the potential clicks that were found by the detection algorithm. In some types of audio, such as trumpets, saxophones, female vocals, and snare drum hits, some of the peaks in the natural waveform may be detected as clicks. If these peaks are corrected as clicks, the resulting audio may sound muffled. With Second Level Verification enabled, these peaks will be rejected, and not corrected after all. True clicks, on the other hand, will not be rejected. Because this option slows down the restoration, you should first try correcting the audio with this option turned off, and only use it for sections that are very troublesome. See Rejection Threshold for more information.

Smooth Light Crackle

This will smooth out one-sample errors when detected, and can have the effect of removing more background crackle. If this makes the resulting audio sound thinner, flatter, or more tinny, leave this option turned off.

Correction Parameters

FFT Size (8 to 512): This is the size of the FFT that will be used to repair the clicks, pops, and crackle. In general, it is a good idea to just let the program itself decide on good FFT size choices by checking Auto. You may wish to override the automatic setting and give a specific FFT size to use for click repair when processing some types of audio. A good starting point is an FFT Size of 32. If the clicks are still quite audible, then try increasing this value to 48, then 64, etc. Any number between about 8 and 512 should work fine. The higher this value, the slower the correction will be, but with potentially better results. However, if the value is too high, there may be some extra unwanted low frequency distortion, which sounds "rumbly". If you are repairing clicks one at a time using Fill Single Click Now, then a high FFT size (128 to 256) should work rather nicely.

Pop Oversamples (0 to 300): This value is used to give the clicks a little extra buffer space when being repaired. When a potential click is found, its beginning and end points are marked as closely as possible. The oversamples value will expand that range, and consider more samples to the left and right of the click as being part of the click. If clicks aren't being corrected fully enough (for example, if clicks become quieter but are still evident), try increasing this value. Start with a value of 8, and increase to slowly as much as 30 or 40. Audio that does not contain a click should not change very much if it is corrected, so this buffer area should remain mostly untouched by the replacement algorithm. Increasing the oversamples will also force larger FFT sizes to be used when in Auto mode. This setting may remove clicks more cleanly, but if set too high, audio will start to distort where the clicks were removed.

Run Size (0 to 1000): The repair algorithm may not work well on two consecutive clicks that are extremely close together. The Run Size setting sets a ceiling on the potential distance between two clicks—that is, any two clicks closer together than the Run Size will be treated as one. A good starting point would be to set this value somewhere around 25 (or half the FFT size if you are not using the Auto setting). If Run Size is too large (over 100 or so) then the corrections may become more noticeable, as very large blocks of data are repaired at once. If you set the Run Size too small, then clicks that are very close together may not be repaired completely on the first pass.

Fill Single Click Now: You can correct a single click by highlighting it and pushing the Fill Single Click Now button. This is the manual override for click restoration. If FFT Size is set to Auto, then an appropriate FFT size will be used for the restoration based on the size of the area being restored. Otherwise, settings of 128 to 256 work very well when filling in single clicks. Once a single click is filled, the **F3** (Repeat Last Command key) key will be set to repeat this action, so filling in future clicks requires only pressing **F3**.

Previously Corrected/Rejected

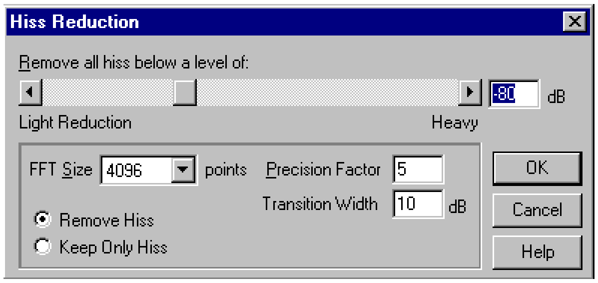
After running a click repair, re-enter the Click/Pop Eliminator to display the number of previously corrected clicks. The dialog will also show the number of rejected clicks that would have been corrected if Second Level Verification were not enabled.

Multiple Pass (2 to 32)

Some clicks may be too close together to be repaired effectively, so going over the audio with another pass will often catch those clicks that weren't found the first time around. If this option is checked, then any number of passes will automatically be performed (up to 32). *Cool E* will perform fewer passes if no more clicks are found and all detected clicks have been repaired. Generally, about half as many or fewer clicks are repaired on each successive pass. When using multiple pass mode, a higher detection threshold may lead to fewer interventions in repairing, and increase the quality of the result somewhat while still removing all clicks. In the same vein, higher rejection thresholds may lead to better results.

Hiss Reduction [See the Manual Addendum for updates!]

This function removes all audio in all frequencies that are below a certain threshold (generally the noise or hiss level). If audio has a constant background hiss (white noise hiss is removed most effectively) then that hiss can be "noise gated" or removed completely. Any audio in any frequency band louder than the cutoff level will remain untouched. Use Hiss Reduction to remove annoying hiss from cassette recordings, record albums (after click/pop removal), or microphone recordings.



Hiss Reduction

Remove all hiss below a level of:

Set this parameter to the amplitude threshold below which you want to remove all audio. Generally, values from -45dB (for very noisy audio) to -100 (for less noisy audio) work best. Try view the hiss level using Spectral Analysis to find the hiss level.

FFT Size

Larger FFT sizes result in a lower detected hiss level, because the hiss will be spread across more frequency bands (approximately 3dB per doubling of FFT size). Use smaller FFT sizes for faster processing.

Precision Factor

Precision Factor determines the accuracy of hiss reduction in the time domain, with larger values producing better results, but also at slower processing speeds. With lower precision factors, you may get a few milliseconds of hiss before and after the louder parts of audio; with higher values, this is reduced. Values higher than 40 or so will not ordinarily improve quality.

Transition Width

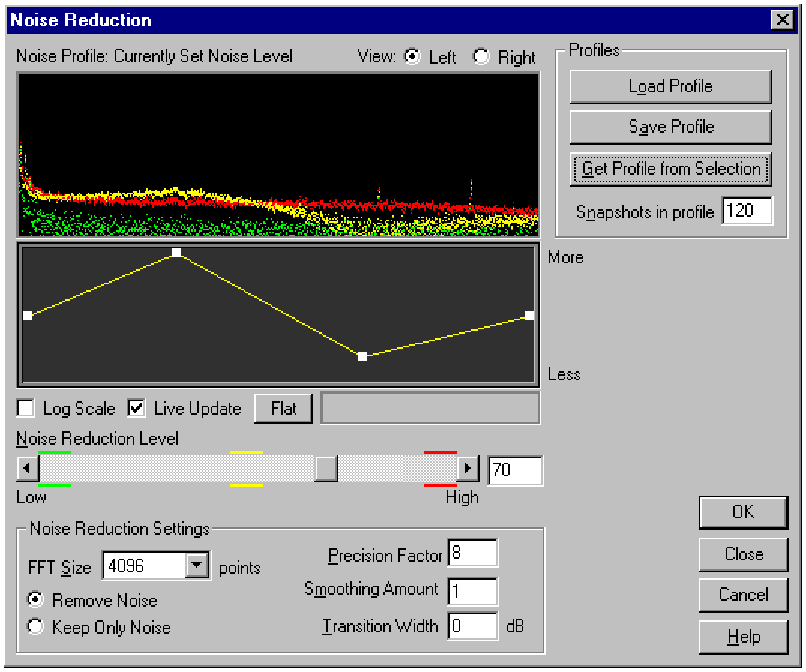
This setting produces a slow transition in hiss reduction, with no reduction taking place at the level specified for Remove all hiss below a level of: above, to 100% reduction at this many dB over that setting. For example, a hiss level of -76dB and a transition width of 4dB will remove 100% of the hiss below 80dB, but only 50% of the hiss at -78dB. If the transition width is too small, other background artifacts may be heard, such as little "tinkles" in the background. If the transition width is set too high, some hiss may remain after processing.

Remove Hiss/Keep Only Hiss

For normal operation, choose **Remove Hiss**. If for some reason you want to extract the hiss only, choose **Keep Only Hiss**, and all the audio will be removed, leaving only hiss.

Noise Reduction

Cool Edit Pro's Noise Reduction feature can dramatically reduce background and general broad band noise with minimal reduction in signal quality. It can also remove tape hiss, microphone background noise, 60 cycle hum, or any noise that is constant throughout the duration of your waveform. The amount of reduction required depends upon the type of background noise, and the allowable loss in the quality of the signal that is to be kept. In general, you should be able to attain increases in Signal to Noise ratios of 5dB to 20dB (noise is reduced 21dB and signal 1dB for example).



Noise Reduction

View Left/Right

Click to view either the left channel noise profile or the right channel noise profile. The amount of noise reduction is always the same for both channel. To perform separate levels of reduction on each channel, edit the channels individually instead of both channels at once.

Noise Profile Plot

The yellow points in the profile plot represent the amount of noise reduction that will be performed at any particular frequency. Move the Noise Reduction Level up or down to move the entire yellow region up or down.

Load Profile

Press this button to load any previously saved noise profile. You can load any *.fft file that *Cool Edit Pro* has saved. A noise profile can only be used on a sample of the same type as when saved. In other words, a 44K stereo 8-bit sample is not compatible with a 22 kHz mono 16-bit profile. Bear in mind, however, that because noise profiles are so specific to the recording environment of the waveform in question, even if the sample types are compatible, a profile for one type of noise is likely to not produce good results when used for another type of noise. 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.

Get Profile from Selection

Press this button to extract a Noise Profile from the highlighted selection. Try to select a portion of the waveform that has no important signal in it (only has background noise), and then press Get Profile from Selection. Cool E then gathers the statistical information about the background noise, after which you can remove all noise of this type from your waveform.

If the selection used for gathering the noise level is too small, then the Get Profile from Selection button will not activate. Reduce the FFT Size or select a larger section of noise by using Copy and Paste for reasonable results with very short noise samplings.

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 likely cut out more noise, but also cut out more original signal as well. However, a low noise reduction level with more samples will also cut out more noise, but likely will not disrupt the intended signal.

Reduction Graph

Use the reduction graph to set the amount of noise reduction at certain frequency ranges. For example, if you need noise reduction only in the higher frequencies, adjust the chart to give less noise reduction in the low frequencies, or alternatively, more reduction in the higher frequencies. The graph depicts frequency along the x-axis (left and right) and the amount of noise reduction (in percent) along the y-axis (up and down). If it is set to Flat, then the amount of noise reduction used is based on the noise profile exactly. The readout below the graph displays the current x, y position of your mouse.

- To add a point to the graph, click in the grid at the location where you would like the point placed.
- To move a point on the graph, click and hold on the point and drag to a new location.
- When the mouse cursor is located over a point, you will see it change from an arrow to a hand.

Log Scale

You can view the chart of the noise profile in either linear or logarithmic fashion. Check Log Scale to divide the graph evenly into 10 octaves. Uncheck it to divide it linearly, with each 1000kHz (for example) taking up the same amount of horizontal width.

Live Update

When you check this option, the noise profile plot updates as control points are moved around on the chart. Otherwise, the profile plot is only updated when the control point is placed in its new location.

FFT Size

The FFT Size parameter causes the most drastic changes in quality. Good settings for the size range from 4096 to 12000. The FFT Size determines the number of individual frequency bands that are analyzed—the noise in each frequency band is treated separately, so the more bands you have, the finer frequency detail you get in removing noise. For example, if there is a 120Hz hum, but not many frequency bands, frequencies from 80Hz on up to 160Hz may be affected. With more bands, there is less spacing between bands, so the actual noise can be detected and removed with more precision. However, with too many bands, time slurring occurs, which can make the result sound reverberant or "echoey" (with pre- and post-echoes). So the tradeoff is frequency resolution vs. time resolution, with lower FFT sizes giving better time resolution and higher FFT sizes giving better frequency resolution.

Remove Noise/Keep Only Noise

For normal operation, choose Remove Noise. If for some reason you want to extract the noise for other purposes, choose Keep Only Noise, and all the audio will be removed, leaving only noise.

Precision Factor

This 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 10 or so, there is no noticeable change in quality—only in the time it takes to compute. Try using 5 or 7 (odd numbers are best for symmetric properties).

Smoothing Amount

Smoothing Amount takes into account the standard deviation, or variance, of the noise signal at each band. Bands that vary greatly when analyzed (white noise) will be smoothed differently than constant bands (like a 60 cycle hum). Generally, increasing the smoothing amount (up to 2 or so) will reduce the 'burbly' background artifacts at the expense of raising the overall background broadband noise level.

Transition Width

This setting determines how sharp the division is between what is considered noise and what should be kept. For instance, with a Transition Width of zero, a sharp (noise gate type) curve is applied to each frequency band. If the audio in the band is just above the threshold, it stays; if it is just below, it is truncated to silence. Conversely, you can specify a range over which the audio will fade to silence based upon the input level. For example, with a transition width of 10dB, and a cutoff point (scanned noise level for the particular band) of -60dB, then audio at -60dB would stay the same, at -62dB it would be reduced some (to about -64dB), and so on until audio at -70dB would be removed entirely. Again, if the width is zero, then audio just below -60dB is entirely removed, while audio just above it would remain untouched. Negative widths simply go about the other side, so in the above example, a -10dB width would have ranged from -60dB to -50dB.

You can 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 in order to further reduce the noise level by up to 10dB.

Noise reduction works best on 16- and 32-bit samples. Although it will work with 8-bit audio, it is impossible to get the noise level to less than about -45dB if even that. Noise at -45dB is very audible, as owners of 8-bit sound cards can attest. You can get much better results by converting to 16 bits first, reducing the noise, and downsampling back to 8 bits, than by reducing noise in 8-bit samples directly.

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

You can generate interesting effects by choosing valid "foreground" audio as your profile "footprint" rather than background noise or hiss. For example, within a vocal line, you can select the vowel "O" to be used as the profile. Processing this vocal file (remember, it's best to process a copy of the original file) will then reduce or eliminate the "O" sounds... thereby creating a rather wild effect.

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

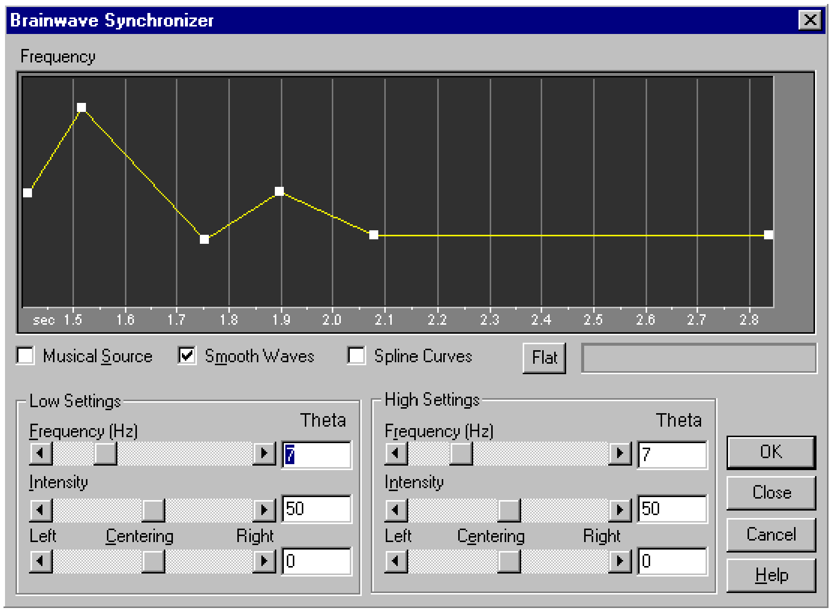
Record a piece of music that has tape hiss or other background noise. Select a short range (at least one half second) that contains only the noise that you'd like to reduce. Now choose the **/Transform/Noise Reduction/Noise Reduction** option and click on **Get Noise Profile** From Selection to sample 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.

Special

The options in this menu can be used to introduce some processing effects that are both innovative and wild-n-wacky. Have fun and experiment!

Brainwave Synchronizer

The Brainwave Synchronizer can produce files that, when listened to with stereo headphones, will put the listener into any desired state of awareness. For example, by listening to "waved" files, you can 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, it must be listened to with stereo headphones. The Brainwave 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.



Frequency Graph

Time is represented along the x-axis (horizontal), and frequency along the y-axis (vertical). 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 and high settings depending on where the graph dictates the signals should be. Choose the highest and lowest frequencies that are represented on the graph with the scroll bars located in Low/High Settings, below. The readout below the graph displays the current x, y position of your mouse.

Cycling between 4 to 5 Hz over a 2 minute span works nicely. If large variations are done in short time spans, the effects are not as pronounced. For example, after 5 minutes of listening to Theta waves, the listener will become slightly awake, if 30 seconds of alpha waves are generated, and then returned back to theta. The effect is sort of like shifting gears into different levels of awareness.

- To add a point to the graph, click in the grid at the location where you would like the point placed.
- To move a point on the graph, click and hold on the point and drag to a new location.
- When the mouse cursor is located over a point, you will see it change from an arrow to a hand.

Musical Source

If the selection being brainwaved is musical, check this setting to 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.

Smooth Wave

When checked, the audio that appears at the left and right channels will be smoothed out. 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 will follow a sine wave pattern, even though the brain will hear the audio as 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. The boundary between holding non-delayed and delaying signal 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, which will cause less noticeable distortion in either channel when heard independently.

Spline Curves

Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use spline curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat

The Flat button will reset the graph to its default state, removing all control points.

Low/High Settings

There are two sets of settings for control of brainwave frequencies. Low settings all correspond to the lower part of the graph (points dragged near the bottom), and High settings affect the top.

Frequency

This is the brainwave frequency that will be encoded into the final process. Different brainwave frequencies stimulate the brain into synching to different levels of consciousness (e.g. sleep, meditation, wakefulness, etc.)

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.

Intensity

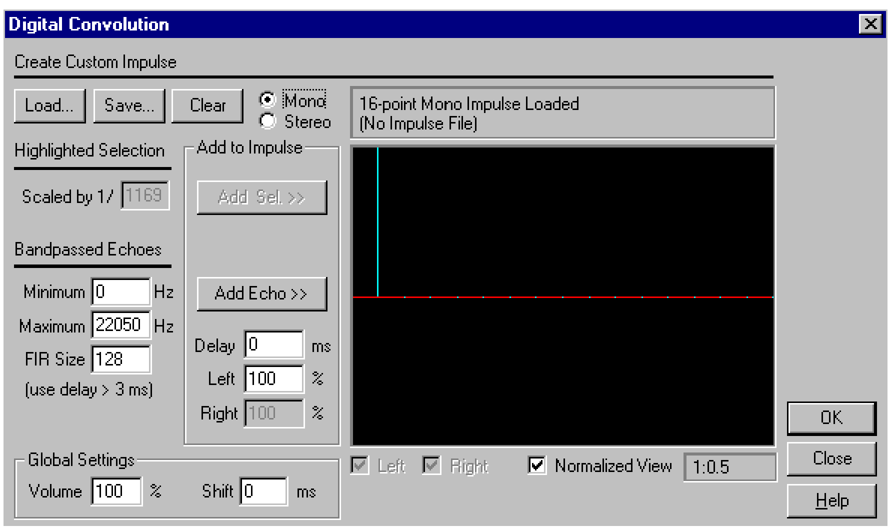
This controls the intensity of the brainwave encoding. Higher intensities work well with lower brainwave frequencies. Beta waves should have intensities below 25 or so, while Delta waves work better with intensities above 60.

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.

Convolution

Convolution is the effect of multiplying every sample in one wave or impulse by the samples that are contained within another waveform. In a sense, this feature uses one waveform to "model" the sound of another waveform. The result can be that of filtering, echoing, phase shifting, or any combination of these effects. That is, any filtered version of a waveform can be echoed at any delay, any number of times. For example, "convolving" someone saying "Hey" with a drum track (short full spectrum sounds such as snares work best) will result in the drums saying "Hey" each time they are hit. You can build impulses from scratch by specifying how to filter the audio and the delay at which it should be echoed, or by copying audio directly from a waveform.



Digital Convolution

With the proper impulses, any reverberant space can be simulated. For example, if you have an impulse of your favorite cathedral, and convolute it with any mono audio (left and right channels the same) then the result would sound as if that audio were played in that cathedral. You can generate an impulse like this by going to the cathedral in question, standing in the spot where you would like the audio to appear it is coming from, and generating a loud impulsive noise, like a "snap" or loud "click". You can make a stereo recording of this "click" from any location within the cathedral. If you used this recording as an impulse, then convolution with it will sound as if the listener were in the exact position of the recording equipment, and the audio being convoluted were at the location of the "click".

To send any portion of unprocessed "dry" signal back out, simply add a full spectrum echo at 0 ms. The Left and Right volume percentages will be the resulting volume of the dry signal in the left and right channels.

Another interesting use for convolution is to generate an infinite sustained sound of anything. For example, one singing "aaaaaah" for one second could be turned into thousands of people singing "aaaaaah" for any length of time by using some dynamically expanded white noise (which sounds a lot like radio static).

Create Custom Impulse

Load: Press this button to load a previously saved impulse.

The sample rate of an impulse will affect the outcome of convolution. For example, if an impulse was created at 44100Hz, and reloaded and used on a 22050Hz file, then everything will be stretched out 2:1. All filtered echoes will be at half the frequency, and all delays will be twice as long.

Save: Press this button to save an impulse for later recall.

Clear: This clears the impulse completely.

Mono/Stereo: Mono impulses work with both mono or stereo data (the left and right channels will be convoluted with the same impulse). Stereo impulses convolute the left and right channels separately.

Highlighted Selection

Any audio can be added to an impulse directly. The scaling factor to use when adding a highlighted selection to an impulse determines its volume. By default, Cool E automatically estimates good starting value for you. Lower this value to increase the amplitude of the impulse.

Bandpassed Echoes

Use these options to add any echo at any frequency to the impulse.

Minimum: When adding bandpassed echoes, this is the lower cutoff frequency of the echo. For example, to echo just the range from 500Hz to 1000Hz, enter 500Hz for the minimum value.

Maximum: When adding bandpassed echoes, this is the upper cutoff frequency of the echo. For example, to echo just the range from 500Hz to 1000Hz, enter 1000Hz for the maximum value.

FIR Size: This parameter sets the size of the FIR filter to use to generate the filtered echo.

Cool Edit Pro recommends a minimum delay (below the FIR entry box) when you add this echo. If you use a smaller delay than that suggested, the echo may contain more frequencies than you want. You can ignore this delay for full spectrum echoes, as they are just single sample ticks in the impulse.

Add to Impulse

Add Sel.: This setting adds the currently highlighted selection to the impulse at the delay and left/right volumes specified. You can add as many selections of actual audio as you like.

You can make any audio data part of an impulse by first highlighting the audio and then using the Add Sel. button. Ordinarily, you should first scale down any such selection to a lower volume; otherwise the convolution will come out extremely loud.

Add Echo: This will add the bandpassed echo to the impulse at the delay and left/right volumes specified. You can add as many echoes as you like.

To add a tick at any volume, enter the volume percentages after the Left and Right settings, and the Delay at which the tick should appear. This will cause an echo of the specified volume to occur at the given delay after convolution. Besides just echoes, you can add filtered versions of echoes by entering the minimum and maximum frequencies to echo. To echo all frequencies outside the range, add a full spectrum echo (e.g. from 0Hz to 22050Hz) at a specific delay, then add another echo at the same delay, but with different minimum and maximum values, and the opposite Left and Right percentages (e.g. at -100% instead of 100%).

Delay

This parameter sets the amount (in ms) by which samples are delayed. For pre-echoes, place at least one echo that is full spectrum (Minimum=0Hz, Maximum=22050Hz) at a larger delay (like 1000ms). Then any echo placed before 1000ms will be a pre-echo.

Left

This setting specifies the left volume (in percent) at which to make the echo or selection being added to the impulse.

Right

This setting specifies the right volume (in percent) at which to make the echo or selection being added to the impulse.

Global Settings

These settings effect how the given impulse will be used during convolution.

Volume: If the convoluted audio comes out too soft or too loud, enter an appropriate value to amplify or attenuate the audio.

Shift: This value is generally set to ½ the FIR Size when building impulses from scratch, to compensate for the delay incurred when the minimum delay used was only ½ the FIR Size. Adjust this shift setting if you find that the convoluted audio is migrating too far to the right with respect to the original audio.

View Settings

These settings affect the impulse display.

Left: Check this to enable display of the impulse for the left channel.

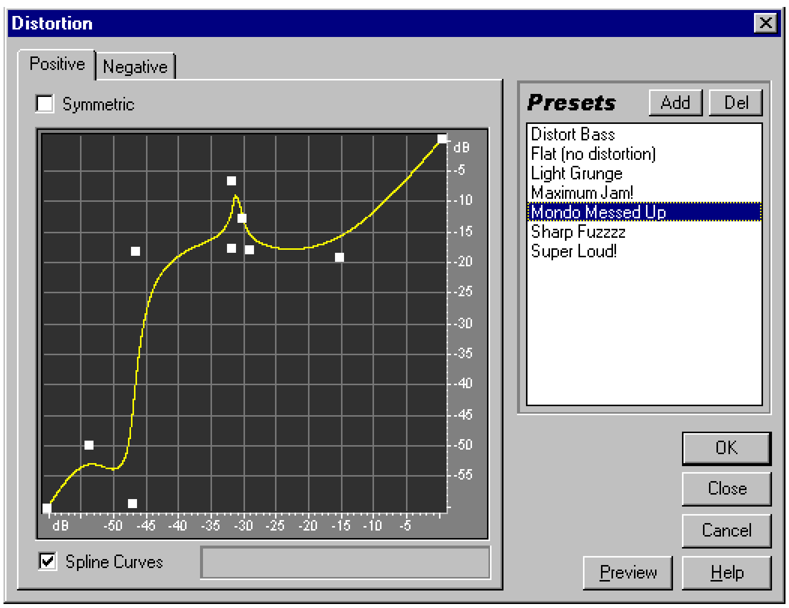
Right: Check this to enable display of the impulse for the right channel.

Normalized View: If Normalized View is checked, the impulse's amplitude is displayed so that it exactly fits in the display vertically.

An "impulse" is the data by which every other sample in your waveform will be multiplied. If the impulse is a single sample of a full volume "tick", then the convolution of that impulse with any audio data will just be that audio data itself. If that "tick" is at half volume, then the original audio data will be reproduced at half volume. If there are several ticks, descending in amplitude over time, such as one tick every 100 milliseconds, and half as loud as the previous tick, then the result of convolution with some audio will be that sound echoed with 100ms between each echo, and each echo at half the volume of the previous echo..

Distortion

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



Distortion

Positive/Negative Tabs

You can specify separate distortion curves for positive and negative sample values. Select the tab corresponding to the distortion curve you want to display. If Symmetric is checked, these tabs become inactive (and are marked "Symmetric").

Copy from Positive

Press the Copy from Positive button (available when viewing the Negative distortion curve) to copy the positive curve to the Negative window.

Symmetric

If Symmetric is checked, the positive and negative curves are identical. To specify separate Positive and Negative curves, turn OFF Symmetric and choose the appropriate tab.

Distortion Graph

This graph depicts the input sample value in dB along the x-axis (left and right) and the output sample value in dB along the y-axis (up and down). A line that flows directly from the lower-left to the upper-right (default) depicts a signal that has been left untouched, since every input value goes to the exact matching output value. Adjusting the shape of this line will adjust the input to output assignments. The readout below the graph displays the current x, y position of your mouse.

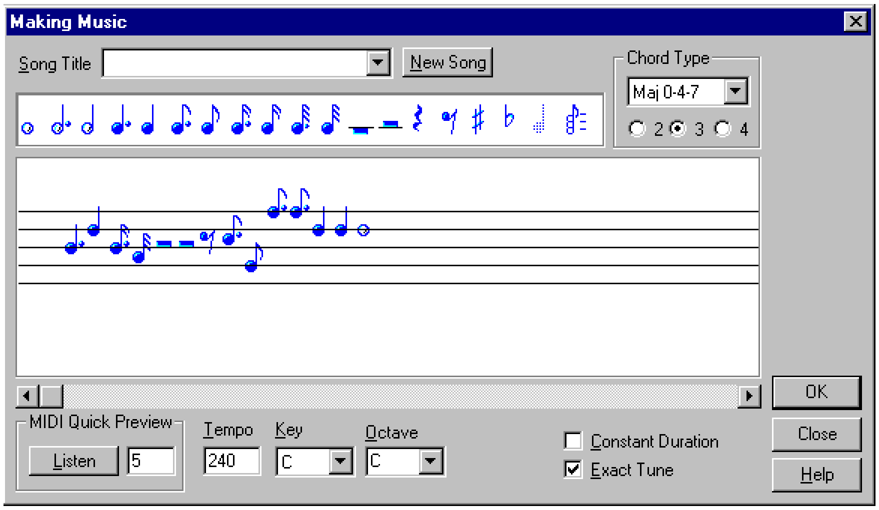
- To add a point to the graph, click in the grid at the location where you want to place the point.
- To move a point on the graph, click and hold on the point and drag it to a new location.
- When the mouse cursor is located over a point, you will see it change from an arrow to a hand.
- To numerically enter input and output signal levels for an Edit Point, right click on the point to bring up the edit box.

Spline Curves

Check this option to generate a smoother, **best fit** curve instead of a straight line between control points on the graph. When you use spline curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Music

Cool Edit Pro's Music feature can use any highlighted selection as a "voice" to synthesize music or 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, Cool E will use the data on the clipboard. Note that the clipboard data will be filled with your sample automatically once music is generated, thus selecting music a second time will automatically use your last sample.



Making Music Dialog

- Simply drag the notes and rests you desire to the staff.
- To sharpen or flatten a note, drag the sharp (#) or flat(b) symbol on top of the note you wish to transpose. 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. You can move notes up or down after they have been placed, or pick them up to insert in a new position. To remove a note, pick it up and drop it off away from the bar.
- Use the scroll bar to work on individual portions of the song at a time. You can scroll to write a piece as long as 256 notes.

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.

Song Title

If you wish to keep a sequence of notes you've created, give it a name in the Song Title box. You can choose saved song from the drop down list of song titles you've created. The actual song data is saved in the file SONGS.INI in your Windows directory.

New Song

Press this button to clear the current song/note sequence.

Notes

Drag the notes and rests you desire from here to the staff below. To sharpen or flatten a note, drag the sharp (#) or flat(b) symbol on top of the note you wish to transpose. You can move notes up or down after they have been placed, or pick them up to insert in a new position.

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.

Chord Type

Use the chord selection drop down list to determine the chord type (min/maj) and voicing you would like to use when entering a chord. You can choose to make a chord out of 2, 3, or 4 voices, then choose the chord type from the list. Then, pick up the chord object (the 3 notes on top of each other) from the Notes area 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.

Staff

Drag notes from the Notes area, and place them here. The note location (A, E-flat) will determine the playback pitch for your sample.

Listen

If you have MIDI play capabilities, you can listen to a preview of your sequence before actually applying it. Play begins at the leftmost note visible on the staff, which means play begins at the position you are scrolled to, and continues on to the end of the song. The music is played through channels 1 and 13 for Extended and Base level compatibility. The instrument can be chosen by typing its MIDI instrument number in the box to the right. 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.

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

Choose a key for the "song" from the drop down list. Only standard Major key signatures are listed, so for a minor key simply choose the relative major (B flat for C# minor, for example). The key of C is the default.

Octave

Transposes the sequence by octaves. Choosing "C" plays the notes at normal transposition.

Constant Duration

When you select Constant Duration, 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

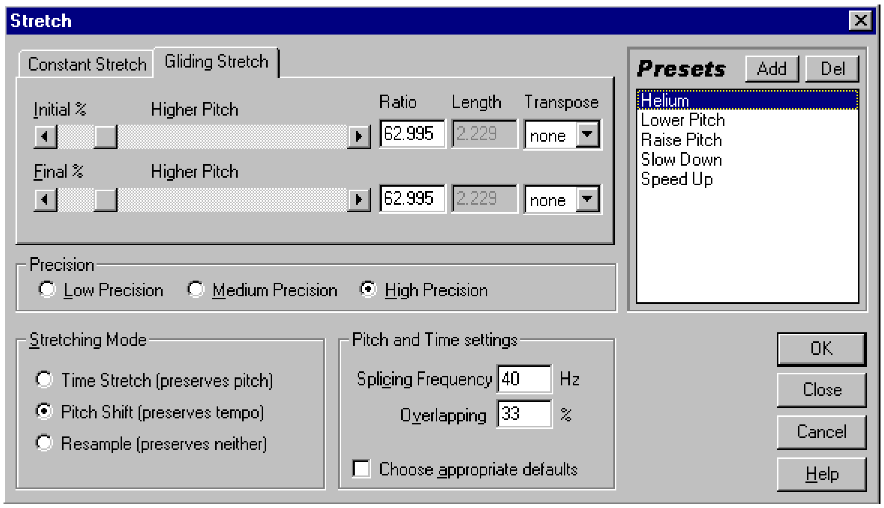
Choose Exact Tune to tune your sample so that when played at A (above middle C), the frequency of your sample is at 440Hz. If this is not checked, your sample's original frequency will be played at A (above middle C).

Time/Pitch

The options in this menu can be used to alter the time (duration) and pitch (tonality) of a audio file or selected range in the following ways:

Stretch

The Stretch function changes the pitch (frequency) and/or tempo (duration) of the audio signal. It can also change either pitch or tempo without affecting the other. For example, you can use Stretch to transpose a song to a higher key, or to slow a passage down without changing the pitch. Pitch and tempo can also be varied over the length of the audio, giving the effect of slowing down and speeding up, or raising and lowering pitch.



Stretch Dialog

Constant Stretch/Gliding Stretch

To perform a constant stretch (no variation in the amount of stretch) on the entire selection, select the Constant Stretch tab. To vary the stretch from one level to another, choose Gliding Stretch. With Gliding Stretch, you can choose the initial stretching percent (less than 100% will compress the wave), and the final stretching percent. Specifying separate values for start and end stretches the waveform in a linear fashion from one ratio to another.

Initial/Final Sliders

Adjust the slider to set the amount of stretch to be applied to the waveform. If you are using Gliding Stretch, sliders for both initial and final settings will be available. Changes in the sliders are reflected in both the Ratio and Length boxes.

Ratio

You can enter a desired Ratio (in percentage) or final Length (in time) for the stretch. 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.

Length

You can enter a desired Ratio (in percentage) or final Length (in time) for the stretch. 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

You can use this drop down list to select musical transposition amounts. The corresponding numerical values are entered into the stretch sliders automatically. For example, to transpose your sound up 1 semi-tone (one half-step on a keyboard) choose 1# for 1 sharp.

Precision

When preserving pitch or tempo, the degree of precision (overall faithfulness to the sound's quality) that an audio file should be processed with is often balanced against the time that it takes to process that file. As you might expect, the tradeoff to having a precision factor that's set too high is that it uses up a great deal of processing power (and thus time.) 8-bit or low-quality audio files can be processed in a short amount of time, using the Low Precision setting, whereas a professionally-recorded audio file may require stretching using the higher-quality algorithm. This will take longer, but the results can be worth it. A quick way to determine the quality factor to use is to process a small portion at the lowest setting and the move up the scale until you've achieved the best balance of quality vs. processing time.

Stretching Mode

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

Pitch Shift: This setting raises or lowers pitch, while the tempo or speed of play remains the same. Higher percentages will lower the pitch, while lower percentages will increase the pitch. Try using differing initial and final percentages to raise and lower the pitch without affecting the tempo.

Resample: When you use this setting, both the pitch and tempo settings are affected. When using percentages above 100, the tempo will slow, while at the same time the pitch will lower. For lower percentages, the tempo will speed up and the pitch will increase.

Pitch and Time settings

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

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

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

This determines how much the current chunk will be overlap with the previous and next chunks. The maximum allowed overlap can be as great as 1000%, in which case up to 10 sections of a waveform will be overlapped together. Whenever the overlap setting is reduced, a choppiness to the sound may appear. You might simply want to adjust this setting to your taste, so as to strike a balance between choppiness and chorusing.

If the low precision mode is used, you can improve the quality of stretched mono-tonal (pure tone) samples by choosing an Interval Rate that's evenly divisible into the frequency of the sample. Use the Frequency Analysis window to find the sample's base frequency, then divide by an integer to get the Interval Rate. For example, if the tone was reported to be 438Hz, dividing by 20 gives 21.9Hz. Thus, using 21.9Hz as the Interval Rate will greatly improve the quality by reducing phase artifacts. For non-tonal or noisy samples, the Interval Rate doesn't matter as much.

Choose Appropriate Defaults: When you preserve pitch or tempo, the Choose Appropriate Defaults box automatically selects good default values for the Splicing Frequency and Overlapping settings.

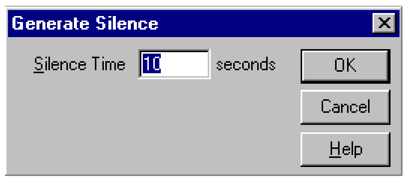
Record a piece of music that features a vocal or a guitar track and select the range that you'd like to effect. Now choose the **/Transform/Time Pitch/Stretch** option and select the "Slow Down" preset and choose **OK**. Now sit back and be amazed!

Generate Menu (Edit View)

The Generate menu displays all the options relating to waveform generation when in the Edit View Mode. These options are:

Silence

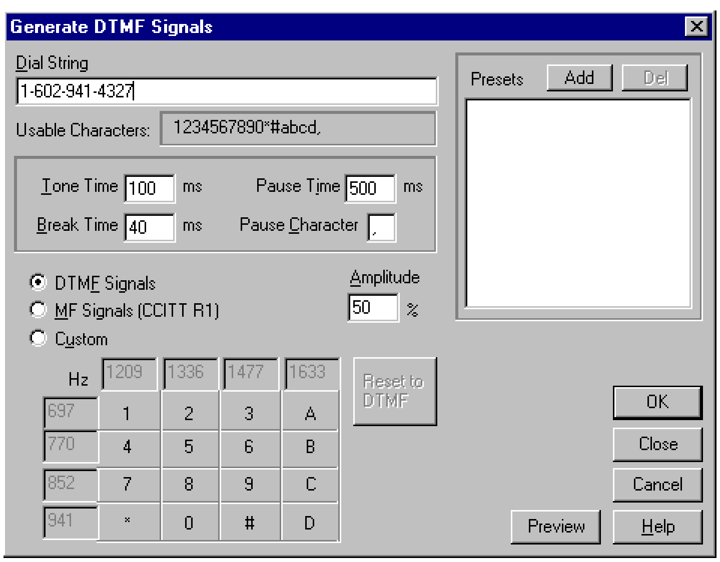
This function generates silence (entered in seconds) from the current playbar cursor position, pushing any following audio back in time, and thereby lengthening the duration of the waveform. Silence will replace any selected area.



To "silence out" a selection, use **/Transform/Silence**.

DTMF Signals

This function generates Dual Tone Multi-Frequency (DTMF) signals used for dialing telephone numbers over the PSTN (phone lines that are capable of responding to touch tone signals). 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.



Generate DTMF Signals

Dial String

Enter the phone number for which you want to generate tones. You may enter other characters such as the '*' and '#' symbols, as well as extra digits 'a', 'b', 'c', and 'd'. Entering the pause character (defined in Pause Character below) inserts a pause of a defined length.

Tone Time

Enter the milliseconds for which the tones will last. The standard time for DTMF tones is 100ms.

Break Time

Milliseconds of silence between successive tones.

Pause Time

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

Pause Character

When you enter this character in the Dial String, *Cool Edit Pro* interprets it as a pause, and will insert silence for the duration specified in Pause Time.

DTMF Signals

Cool Edit Pro generates DTMF (normal push-button telephone type) signals using combinations of the frequencies 697Hz, 770Hz, 852Hz, 941Hz and 1209Hz, 1336Hz, 1477Hz, and 1633Hz.

MF Signals (CCITT R1)

Cool Edit Pro generates MF (internal to telephone networks) signals using paired combinations of the frequencies 700Hz, 900Hz, 1100Hz, 1300Hz, 1500Hz, and 1700Hz.

Custom

Allows you to specify the combinations of frequencies to be used in generating signals. Press this option to enable the frequency entry boxes on the keypad chart below. Click in an entry box to change that frequency.

Amplitude

This determines the volume level (in percent) of the tones generated, where 100% means maximum volume without clipping.

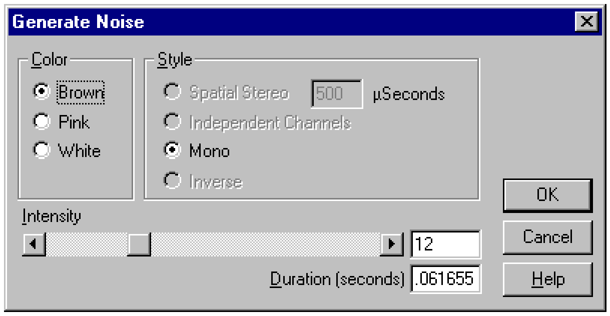
Reset to DTMF

Clears any custom frequency entries and replaces them with the standard DTMF frequency combinations.

The presets in this function save everything, including the dial string. To see how effective these tones are, try entering 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. Your phone will dial the number you entered!

Noise

This function generates random noise in a variety of colors. **Traditionally, color is used to describe the spectral composition of noise.** Each color has its own characteristics. *Cool Edit Pro* inserts noise from the current playbar cursor position, pushing any following audio back in time, and thereby lengthening the duration of the waveform. Selecting an area and generating noise replaces the selection (deleting existing audio).



Generate Noise

You can use noise as the basis for creating weird SFX (sound effects), for creating soothing sounds like waterfalls (great for use with the Brainwave function), or for generating signals that can be used to check out the frequency response of a speaker, mic, etc.

The Cast of Colors:

Brown: Brown noise has a spectral frequency of 1/f^2. This means, in English, that there is much more low-end, and there are many more low-frequency components to the noise. This results in thunder- and waterfall-like sounds. Brown noise is so called 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. When graphed, this waveform looks like a mountain range. The wave pattern is very predictable.

Pink: 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 the sounds, you can generate rainfall, waterfalls, wind, rushing river, and other natural sounds. 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: 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, white noise sounds very "hissy".
Cool Edit Pro generates white noise by choosing random values for each sample.

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

Spatial Stereo: *Cool Edit Pro* generates Spatial Stereo noise 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 you listen to it with stereo headphones, your 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: *Cool Edit Pro* generates this noise 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: *Cool Edit Pro* generates Mono noise by using 1 noise source, with the left and right channels set equal to the same noise source.

Inverse: *Cool Edit Pro* generates Inverse noise 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 you listen to it with stereo headphones, the effect is that of the sound coming from the center of your head instead of out in space somewhere.

Intensity (2 to 40)

With higher intensities, the noise becomes more erratic, and sounds harsher and louder. You can adjust the slider, or enter the value numerically in the entry box.

Duration

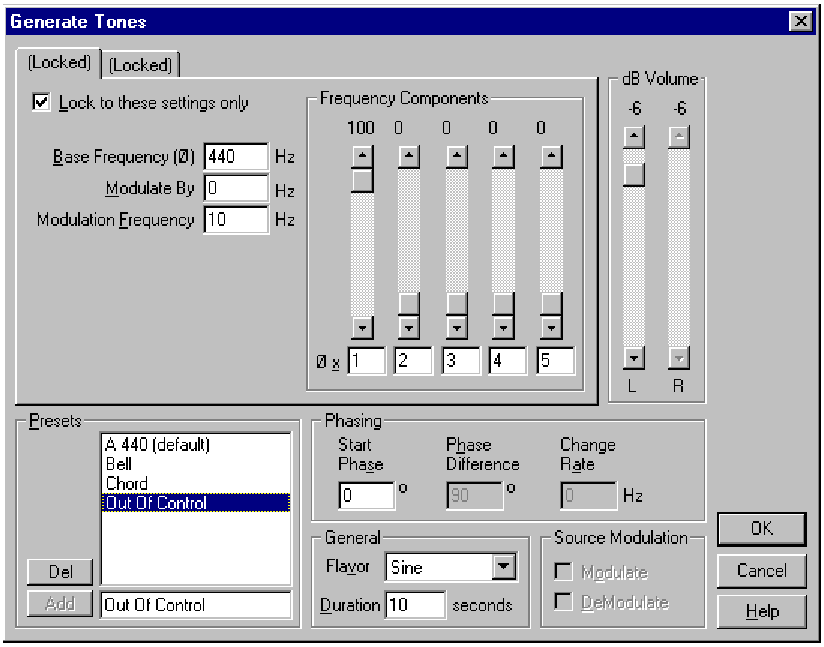
Enter here the number of seconds of noise you want to generate.

For very long periods of noise, it is faster to generate a shorter period of about 10 to 20 seconds, delete excess noise at the beginning and end so that the waves are starting and ending at the midpoint, then copy, and loop (using **/Edit/Mix Paste**) as many times as needed.

Open a new waveform and select the **/Generate/Noise** option and select the default setting (12 seconds of spatial, brown noise) Now sit back and be relaxed!

Generate Tones

Generate Tones will create a simple waveform, with control over numerous amplitude- and frequency-related settings. Generating tones is a great way to provide a base sound to create spectacular special sound effects.



Generate Tones

Initial Settings

When **Lock to these settings only** is off, choose this tab to select the initial tone settings. The tone generated will gradually go from the initial state to the final.

Final Settings

When **Lock to these settings only** is off, choose this tab to select the final tone settings. The tone generated will gradually go from the initial state to the final.

Lock to these settings only

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

Copy from Initial Settings

Press this button (available when viewing the Final Settings) to copy the initial tone settings to the Final Settings window for editing.

Base Frequency

Enter here the main frequency to be used for sound generation.

Modulate By

This setting modulates the Base Frequency in pitch over a user-defined range. For example, a 100Hz setting modulates the original frequency by +- 100Hz (i.e.: a 1000Hz tone would modulate between 950Hz and 1050Hz).

Modulation Frequency

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

Frequency Components (Overtones)

Up to 5 overtones can be added to the fundamental frequency (Base Frequency). You can enter the multiplier for each overtone below the slider (the actual frequency will be this many times the fundamental.) You can mix each of the individual components (0 to 100%) in proportion to one another, and you can adjust the overall gain (signal level) via the stereo sliders. If Lock is not checked, all of these can change over the duration of the audio file, so that they morph from the initial to final settings.

You can generate many really great effects with just these 5 overtones. Just experiment and have fun!

dB Volume (-80 dB to 0 dB)

Use the Volume sliders to select the overall gain for each channel. You can control both channels independently when generating stereo tones.

Start Phase

This is the starting location in the cycle that will be produced. If you start at 0 degrees phase, waves will start at the baseline. If you start at 90 degrees, the wave will start at full amplitude (generating a noticeable click as well). If you are working in great detail with tones and need to have the phase "just so", this option allows you to control that.

Phase Difference

This setting purposefully allows the left channel to be out of phase with the right channel. A value of 0 will be completely in-phase, and 180 will be completely out of phase.

Change Rate

Use this setting to dynamically change the relative phase between the two channels of a stereo audio file over time at a given rate. For example, if you enter 1Hz, the phase difference will cycle through 360 degrees each second.

Flavor

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

Sine	Fundamental, no harmonics (pure tone)
Triangle	Odd harmonics with amplitude of 1 to itself (squared)
Square	Odd harmonics with amplitude of 1 to itself
Sawtooth	All harmonics with amplitude of 1 to itself

Duration

This setting lets you enter the length of the generated tone (in seconds).

Modulate

Modulate requires a highlighted selection. Instead of generating new tones, the currently highlighted wave data is be "ring modulated", or multiplied, by the current tone settings. This is great for adding really weird special effects.

Demodulate

Demodulate requires a highlighted selection. Instead of generating new tones, the currently highlighted wave data is be "ring modulated", or multiplied, by the current tone settings.

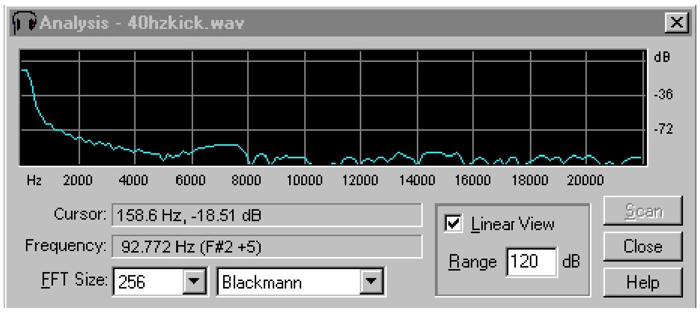
Open a new waveform and select the **/Generate/Tones** option and select the "Out of Control" preset. Now sit back and be spaceialized!

Analyze Menu (Edit View)

The Analyze menu contains functions that give you information about the file or selection being viewed when in the Edit View mode. These functions are:

Frequency Analysis (Alt+Z)

The Frequency Analysis window contains a graph of the frequencies at the insertion point (yellow arrow cursor) or at the center of a selection. This window "floats", meaning that you can click in the waveform on the main *Cool Edit Pro* window to update the analysis while the Frequency Analysis window is on top. The Frequency Analysis performs a 2048 point Fast-Fourier-Transform to determine the frequencies.



Frequency Analysis

The information in this dialog is like one "slice" or line in the Spectral View of the waveform. The most prominent frequency is interpolated and displayed in a window below. You can move the mouse over the graph area to display the frequency and amplitude components of that frequency.

When you set the FFT size to 1024 or lower, the Frequency Analysis window updates in real time 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 Pro* displays the spectral information in the Analysis window.

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

When you view stereo data, the left channel is shown in Cyan, and the right shows in Magenta.

Graph

Graph depicts the frequency spectrum along the x-axis (left to right), and amplitude, in dB, along the y-axis (bottom to top). The Cursor readout below the graph shows the current x, y position of your mouse.

Cursor

Displays a running update of your current x, y mouse position in the graph, in Hz and dB.

Frequency

This is the most prominent frequency present at the insertion point (yellow arrow cursor), or at the center of a selection.

FFT Size

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

When you set the FFT size to 1024 or lower, the Frequency Analysis window updates in real time 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 the Right Arrow key. As the cursor scrolls across the display, *Cool Edit Pro* displays the spectral information in the Analysis window (try this zoomed in all the way down to the sample).

FFT Window Type

Different window types display different frequency graphs. The Triangular window gives a more precise frequency estimate, but is also the noisiest, meaning that 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 broader frequency band which isn't as precise, but the sidelobes are very low, making it easier to pick out the major frequency components.

Linear View

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.

Range (24 dB to 240 dB)

Enter the range (from 24 to 240dB) for the vertical scale (the y axis) of the graph.

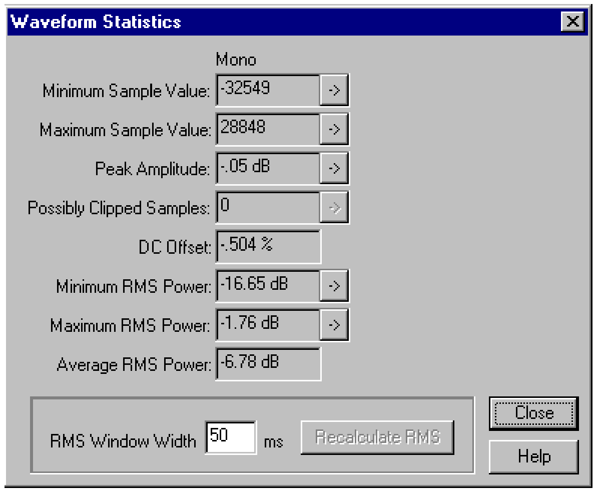
Scan

Clicking Scan will scan the highlighted selection and show all frequencies present in that selection.

This is a really cool one to check out! Simply record a waveform and then call up the Frequency Analyzer by selecting **Alt+Z**. You can view the analysis at a specific point (cursor location) or you can view your frequency balance in real time, during playback. If your computer isn't fast enough to view the spectrum in real time, simply choose a smaller **FFT Size**.

Statistics

Use the Statistics dialog to get the following details about the current waveform:



Waveform Statistics

You can jump to the exact location in the waveform for certain values by clicking on the appropriate arrow to the right of that value.

Minimum/Maximum Sample Value

Shows the maximum and minimum sample values in the range. Press the [>] button associated with the value to place the cursor at that location.

Peak Amplitude

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

Possibly Clipped Samples

Counts the samples which may be clipping; at -32768 or 32767 (for 16-bit), for example. Press [>] to jump to the first such sample.

DC Offset

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

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

Average RMS Power

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

Cool Edit Pro takes the RMS over a window of this size when calculating the RMS minimum and maximum values.

Recalculate RMS

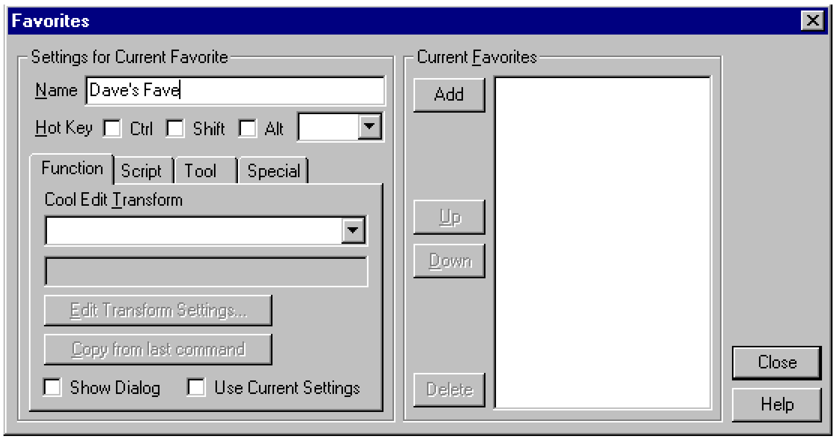
After adjusting the RMS Window Width, click this button to recalculate the RMS values based on the new window size.

Favorites Menu (Edit View)

The Favorites menu allows you to create, customize and save your favorite *Cool Edit Pro* Transforms, Generate effects, Scripts, and even 3rd-party tools (using command line executables). You can also categorize the menu items you create into hierarchical sub-menus for easy organization.

Edit Favorites

Use Edit Favorites to create, delete, edit, and organize items appearing in the Favorites menu. Edit Favorites can instantly call up any customized *Cool Edit Pro* Transform or Generate effect, Script, or even 3rd-party tool (command line executable). The menu can also contain sub menus for easy organization.



Favorites Dialog

Settings for Current Favorite

Enter here the name that will appear in the Favorites menu, and assign all of the properties to that item.

Name

The name of the item to appear in the Favorites menu. Create hierarchical menus by using a backslash. For example "Effects\Hall Reverb" places the item under "Hall Reverb" in the "Effects" sub menu.

Hot Key

You can assign a key combination to serve as a shortcut to any of your Favorites. Place a check next to Ctrl, Shift, Alt, or any combination of the three to act as the first part of your shortcut. Then select a key for the second part of your shortcut (the key to press while holding the checked combination), and type the letter or number (no symbols) in the box to the right. If you want to use Function keys, choose them from the drop-down list.

Function

***Cool Edit* Transform**: Choose the particular Transform or Generate item you would like to add from the drop-down list. A list of the last used settings for the item chosen appears in a window just below the Transform list.

Edit Transform Settings: Click on this button to call up the settings window that corresponds to the particular Transform function. You can then change the settings to be used when this item is chosen from the Favorites menu.

Copy from last command: Copies all the settings from the last function that was started successfully.

Show Dialog: Check Show Dialog to bring up the settings dialog for the particular Transform when you call the Favorite. The settings dialog will show the settings you chose in Edit Transform Settings. Un-check Show Dialog to automatically use the chosen settings to run the function.

Use Current Settings: Check this option to apply to the Favorite the settings last used for that particular Transform.

Script

Script Collection: Displays the current script collection in use.

Choose Script File: Brings up a browse dialog to choose a script collection (*.scp) file.

Script: This drop-down list contains all scripts available in the selected collection. Choose the particular script you want to run.

Pause at Dialogs: If checked, the script stops at each settings dialog to allow you to modify the values. Otherwise, the entire script will run to completion before control returns.

Tool

Command Line: Type the command line for the tool you want to run here, including any command line switches the particular tool may need.

Browse: Click to browse for a particular tool.

Special

Use this option if you'd like to enter a separator bar into the Favorites list to separate one function type from others. You can do this by entering a series of dashes (e.g. "------") into the Name box. If you want more than one separator, enter a different number of dashes, or change the text so that it doesn't match something already in the list (e.g. "------2" is considered a valid separator; the 2 after the dashes will be ignored). To create a separator bar for a sub menu, enter the sub menu path first (e.g. "Effects\------"). You can also enter any text you would like the Favorites menu to display. The text will be nonfunctional, of course.

Current Favorites

Once you fill in the name and attributes for the item, it can be added to your Favorites list. All items in this list will appear in the Favorites menu.

Add/Update

Add the Favorite currently being edited to the list, or update an item in the list if its name already exists. If you modify the settings for a current favorite, you must press Update to make the changes permanent. To add a completely new item to the list, just type a new name in the Name field, and choose the appropriate settings.

Delete

Removes the selected item from the Favorites list.

Up

Moves the selected item up in the list. The order of this list is the same as in the Favorites menu.

Down

Moves the selected item down in the list. The order of this list is the same as in the Favorites menu.

Favorites List

This list displays items currently in the Favorites menu. Choose any item in the Favorites list to call up the settings for that item. If you change some of those settings, be sure to press Update to permanently change the settings in the list. The order of this list is the same as in the Favorites menu.

Let's add a **Normalize** function to the **Favorites List**. First, select **Edit Favorites..** from the Favorites menu, then let’s name the function "Normalize"... select the **Amplitude\Normalize** from the Function Tab. By pressing the Edit Transform Settings, we can tailor any variables that we'd like. Next, press **Add** to add "Normalize" to the list and press Close. Next time you want to normalize a waveform, you can simply call it up from the Favorites List.

Options Menu

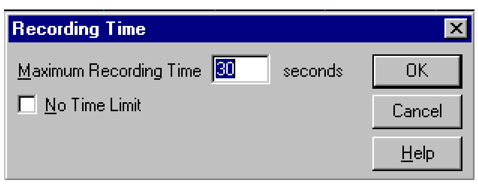
The Options menu gives you access to the system's configuration options and batch-processing functions (for processing multiple audio files from a single program application.)

Loop Mode

The **Loop Mode** option switches toggles the play button between Loop and Play states. When Loop Mode is checkmarked, the Play button will continuously loop the current selection (or the entire file, if nothing is selected); when it is off, it the Play button plays the selection (or file) once and then stops playback.

Timed Record (Edit view only)

This option can be used to set a finite time limit allowed when recording. When your time limit is up, *Cool Edit Pro* will automatically drop you out of record mode. With the **Timed Record** option check-marked, pressing the record button on the transport toolbar will bring up a dialog box prompting you to enter the total recording time (in seconds). Upon pressing OK, *Cool Edit Pro* will be placed into record mode, either until the stop button is pressed, or the time length has expired. By default, the record time is set at 30 seconds.



Time Record

Monitor VU Level

This option will activate the Level Meters and start monitoring the recording source, which is useful for setting recording levels before recording. To stop monitoring, press the Stop console button.

Cool Edit Pro's **Level Meters** are found at the bottom of both the Waveform and Multitrack Windows. The incoming signal from your selected sound card(s) is represented as the peak amplitude in decibels, where a level of 0dB is the absolute maximum before clipping occurs. If clipping does occur, the clip indicator to the right of the meter will light up, and remain lit. Clicking on the clipping indicator at any time will reset it. Yellow peak indicators will stick for 1.5 seconds before resetting to allow for reading of the peak amplitude. When displaying stereo audio, the top meter represents the left channel, and the bottom, the right.

Monitoring may also be started and stopped by double-clicking directly on the Level Meters.

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

To access the Meter Level configuration menu, right-click in the meter area. The following options are available:

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

Activates the meters while playing and recording in *Cool Edit Pro*.

Clear Clip Indicators

The box(es) at the right will light up if audio is **clipping**. Click on the box or choose this option to reset the indicator. 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 higher range to see the quieter portions for very high dynamic range audio. Note: You may find that when you think your sound board is recording pure silence, you will see the meters fluctuating between points around -87dB up to -60dB instead of going all the way down. This is because of noise in the sound board. Some sound boards have higher signal-to-noise ratios than others. Generally, the higher quality the sound board, the lower the meters will go down during pure silence. To quickly see how noisy your own sound card is, choose **/File/New** and create a new 44.1Khz 16-bit file. Then start the level meters, and choose the 90dB Range. This test only works for 16-bit sound cards, since 8-bit sound cards have a maximum dynamic range of around 45dB.

Dynamic / Static Peaks

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

MIDI Trigger Enable

By check-marking MIDI Trigger Enable, any of *Cool Edit Pro*'s Shortcuts that have been assigned to MIDI events can be called from a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command. Disable this option if you don't wish to have Shortcuts respond to MIDI events.

Before attempting to enable MIDI triggering, you will need to choose a device for MIDI In that is recognized by Windows (such as a sound card's built-in MIDI interface, a MIDI interface card or other hardware device options.) To do this, go to **/Options/Settings/Devices** to bring up the devices window. Set MIDI In (Sync/Trigger) to the source to be used for Triggering (commonly YourDevice: In 1).

SMPTE Slave Enable

With this option turned on, *Cool Edit Pro* can synchronize its playback to SMPTE time code (MTC) generated from another device, such as a MIDI sequencer, or with the appropriate hardware, a VCR or tape deck. This is commonly referred to as slaving. With SMPTE Slave enabled, *Cool Edit Pro* will update its current sync status in the lower left of the Status Bar, with **Opened MIDI Input Device** when waiting, **Synchronizing** when establishing lockup, and **Playback Synchronized** when sync is actually established.

To establish sync, you must:

First, set a SMPTE offset (a time code location for *Cool Edit Pro* to begin playback/record) if this value is other than 00:00:00:00: In Multitrack view select **View -> Info**. Enter a SMPTE location where you would like *Cool Edit Pro* to begin playback, in the format hrs:min:sec:frames.

Next, tell *Cool Edit Pro* to wait for SMPTE, choose **Options -> SMPTE Slave Enable**. Start playback on the "master" device and *Cool Edit Pro* should pick up the time code and give a running update in the SMPTE readout at the lower right of the main window. Playback in *Cool Edit Pro* will then begin at your SMPTE location.

Before attempting to enable SMPTE sync, you will need to choose a device for MIDI In that is recognized by Windows (such as a sound card's built-in MIDI interface, a MIDI interface card or other hardware device options.) To do this, go to **/Options/Settings/Devices** to bring up the devices window. Set MIDI In (Sync/Trigger) to the input and to be used for SMPTE sync (commonly YourDevice: Sync).

Cool Edit Pro requires a "preroll" of about 5 seconds of time code before the actual program begins. This is to allow the system time enough to establish "synchronization lock". The readout in the lower left corner of the main window will read **Synchronizing** when establishing lockup, and **Playback Synchronized** when sync is actually established.

Synchronize Cursor Across Windows (Edit view only)

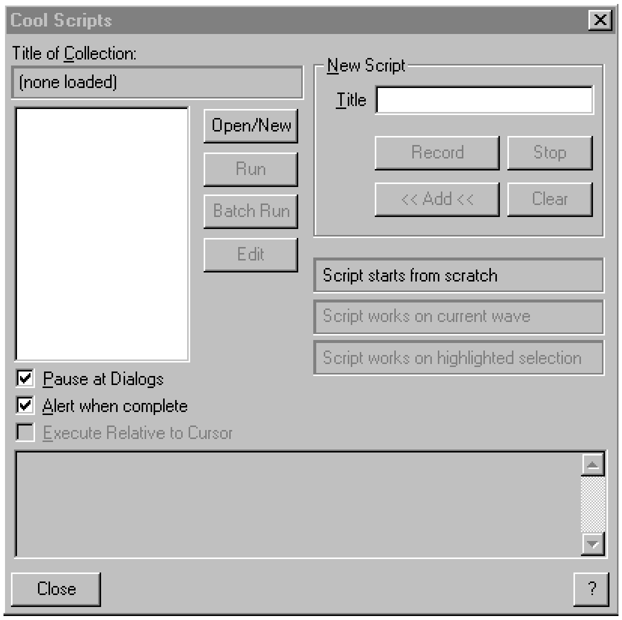
With this command check-marked, the playback cursor bar or currently highlighted selection will retain its position when switching between open waveforms within *Cool Edit Pro*'s Edit View Window. For example, if you are looking at waveform and highlight from 0:01:00 to 0:02:00 and then switch to another open waveform, the second waveform would also be highlighted between the 0:01:00 to 0:02:00 points. This can be useful if are switching between different versions of the same audio file during editing. When unchecked, the positions of each of the waveform's previous cursor or range areas will be retained and re-displayed. If the waveform that you are switching to does not contain the currently selected time location (as in a .5 sec waveform, in our example) then switching acts this way, as well.

When working in the Multitrack Mode, this option should be turned off, as the Edit-View and Multitrack-View display modes will rarely have any timelimes or markers in common, and will often only serve to confuse your relative cursor/marker points.

Scripts and Batch Processing (Edit view only)

Have you ever wished that you could perform a series of tasks on your editor (no matter how simple or complicated they are), and then save these steps, so that they can be repeated whenever you needed them. Well, using Script and Batch files, you can do just that! Basically a *Cool Edit* script lets you record any number of processing steps (such as normalizing, equalizing and then saving a file in a specific format). Once recorded, the script can be saved as a file for later use on another file, at any time. Batch processing of scripts simply means that you can perform all of the steps that have been recorded into a script and then automatically perform them upon any number of files... at the simple touch of a button. In addition, either of these functions can be added to the Favorites Menu or be assigned to simple keyboard Hot Key commands.

For example, suppose you have a combination of effects with particular settings (an EQ setting, a Hall reverb, etc.) that you like to apply often and in combination to achieve a certain sound. You can record these steps, and specific settings, to be carried out again by simply calling the script.



Scripts Dialog

There are three types of scripts. The type created depends upon your state when you initiate the recording:

- Scripts that start from scratch. These scripts start with no waveform opened, and their first command is /**File/New**.
- Script that works on the current wave. This type of script operates on the entire waveform. It requires a file to be opened, but with no selection made. Actions begin at the playback cursor position in the waveform, and will affect any data present at that point.
- Scripts that work on a highlighted selection. This type requires a highlighted selection to be made. All actions in the script pertain only to the portion that is highlighted, leaving the rest of the waveform untouched.

Again, to create a particular type of script, simply match your current state with that of the target for the script. For example, if you wish a script that starts from scratch, then close all open waveforms before recording the script.

While all scripts will be displayed in the script list, only scripts whose type matches your current status (open waveform with a highlighted selection, for instance) will be allowed to run.

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

Title of Collection

This text area displays the title of the currently opened script collection. If the collection in use has not yet been named, New Collection will be displayed. To name a script, press the Edit button and look for the **Collection:** entry (the first line). Overwrite New Collection with your title.

Scripts List

This area lists the names of all scripts in the currently opened collection. Highlight a script in the list to edit or run it.

Open/New

Pressing this button brings up a standard Open dialog with which you can browse your directories for script (.SCP) files. Double click an existing .SCP file to open that script collection. To create a new script collection, navigate through the dialog to the directory that you would like to save the .SCP file, and enter a name for it in the File Name: field. Next, press Open to save the file (it will then open for use as the current script collection).

Run

Runs the currently selected script. If the script type is not compatible with your current state (i.e. trying to run from scratch script with a file open) it will not run until your condition matches the requirements (in this case, closing all open files).

Batch Run

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. Pressing this button will bring up the Batch Process dialog, allowing you to select the files to process.

For more information, see the section on **Batch Processing**.

Edit

This will open the currently selected script as a text file for editing.

New Script

Title: Before creating a new script, you must first supply a title here.

Record: After titling your soon-to-be script, hit the Record button to begin capturing the steps to be included.

Stop: After completing all of the steps to be included in the script, return to the Cool Scripts dialog and press Stop.

Add: This button will add the script you have just created to the currently open Script Collection. Remember that if you wish to supply a description for the script (in the Description text area below) do so before hitting this button.

Clear: This option allows you to clear the script you have just recorded, if you do not wish to add it to the collection. This button is only available immediately after pressing Stop. If you wish to remove a script after it has been added, you will need to edit the script by pressing the Edit button.

Script Type

These three fields indicate what type of script is currently highlighted in the Scripts List. There are three possible script types, with each operating upon either the entire waveform, a selection, or from scratch. The type of the script currently selected will darken, leaving the other two "grayed out".

Pause at Dialogs

Check this option to make the script stop at each dialog, allowing you to modify the values of that function. Pressing Cancel in any dialog will stop the script, pressing OK will continue it.

Alert when complete

A dialog box will signal the completion of the script if this option is checked.

Execute Relative to Cursor

When using "Works on Current Wave" script types, you can have all script operations performed relative the original positioning, as opposed to at the original position. For example, suppose you have a script which was originally recorded with a cursor position of 0:10:00. Checking this option will apply the script at the current cursor location, plus 10 seconds (if your cursor is at 0:05:00, the results would be at 0:15:00). With this option unchecked, the script will be performed at the original location (0:10:00 in our example).

To make a script which you will execute at the current cursor position, record the script at a 0:00:00 position, and check this option.

Description

Immediately after recording a script, you may enter a description in this text area for the script you have just recorded. This description will appear (in a non-editable form) when the user of the script highlights the script to run.

The only time you can enter a description in this box is after recording, not before, and not after it has been added to a script collection file. If you wish to enter or edit a description thereafter, you can do so by pressing the Edit button to edit the text file directly.

Batch Processing

A single script can be run repeatedly over a group (batch) 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.

You can customize *Cool Edit Pro*'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.

Source Files

This is the list of files to be included in the batch for processing. Press the Add Files… button to add a file to the list, or Remove to remove a file from the list. To remove multiple files from the list, hold down on the SHIFT key for contiguous selections, or the CTRL key for noncontiguous selections.

The wave files can be in different formats if desired.

Add Files…

This button brings up a standard Open dialog, allowing you to add files to the batch for processing. To add multiple files to the list, hold down on the SHIFT key for contiguous selections, or the CTRL key for noncontiguous selections.

Remove

This button removes any selected files from the Source Files list.

Destination Directory

This is the directory to which the processed files will be saved, after the script has been run on them. You can select the Destination Directory by typing in the path here, or by pressing the Browse button.

Output Format

This selects the file format that all processed waveforms will be saved as. You may choose from any of *Cool Edit Pro*'s supported file types, as well as enter any appropriate options for the target format by pressing the Options button, if available.

Output Filename Template

The names of files in your batch can be modified before being saved to the Destination Directory. When running a batch, the processed file's extension will automatically change to that of the format chosen in Output Format (e.g. *.AIF for Apple AIFF). You can, however, force another extension, or alter the filename itself (portion before the "."), by using the filename template. The are two characters to use when altering the Output Filename Template:

- A question mark '?' will signify that a character does not change.
- A star '*' will 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

Disable Undo

This will disable the saving of undo information for the duration of 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 since undo information does not continually need to be saved.

Overwrite Existing Files

With this option checked, files will always be saved to the Destination Directory after performing the script, regardless of whether a file of the same name already exists at this location. 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.

Open Raw PCM As…

If a source file is unreadable, or in a RAW format without any header information, then the batch needs to know what file type to assume for the data. Pressing this button allows you to define a global file type for interpretation in such cases; you will need to set the sample rate, channels, resolution, and formatting. This ensures that the batch will run continuously without interruption by dialogs asking for input data formats on header-less files.

Begin Processing

Press this button to begin batch processing all files in the Source Files list.

If you need to convert multiple files from one format (like .WAV) to another (like .AU), you can use *Cool Edit Pro*'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.

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.

Settings/General

Hilight after Paste

When performing a Paste operation (including Mix Paste), you can elect to have the inserted selection automatically highlighted, or to have the cursor placed at the end of the pasted selection. Check this box to enable highlighting.

Leave this option unchecked for easier multiple pastes, one after the other.

Use old style file open/save dialogs

Check this box to have *Cool Edit Pro*'s File Open and File Save dialogs similar to those used in 16-bit Windows applications, as opposed to the Windows 95 style. You may wish to use this older style if you are used to them and do not want the extra features of Windows 95 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

When launching *Cool Edit Pro*, this option will automatically play a file that has been specified on the command line. For example, if you choose Start - Run and enter "c:\cool\coolpro.exe thisfile.wav" as your command line, then *Cool Edit Pro* will open, and begin playing thisfile.wav.

Play From Cursor

When no selection is highlighted, you can have playback begin from either the current playback cursor location, or from the left edge of the waveform display. Check this box to start playback from the current cursor position.

Live update during record

This enables live waveform drawing while recording. On faster machines, you can have the waveform displayed in real-time as audio is being recorded. However, if you find the recorded audio becoming choppy, leave this option disabled. In Spectral View mode, and at lower spectral resolutions (around 256) a nice scrolling spectral plot can be performed while recording with this option on.

Auto-scroll during Play and Record

If enabled, the waveform display will scroll in sync with playback. This only affects when you are zoomed in on a portion of a waveform, and play past the viewed portion (by pressing Shift+Play, for example).

Show center line on top

Check this box to have the center line (zero amplitude) for each channel displayed on top of the waveform itself. If disabled, the waveform is displayed on top of the center line.

Beat Sensing

Decibel Rise (dB): This is rise in amplitude needed to constitute a beat when using the Find Beats function. The waveform must rise by this amount within the specified *Rise Time* to be considered a beat.

Rise Time (ms): This is the amount of time (in milliseconds) in which the amplitude must rise by the *Decibel Rise* setting in order to be considered a beat when using the Find Beats function.

Maximum Display on Load (seconds)

This is the maximum number of seconds into the audio file 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 that you do not have to wait for the entire waveform to draw. Setting this value to zero means there is no limit on the initial display size.

If you have the Peak Files feature enabled (see **/Options/System**), then waveform redrawing will not be an issue; after the file has been loaded once, subsequent redraws will be instantaneous.

Custom Time Code Display (FPS)

This field is used to define the number of frames per second (FPS) assigned to the Custom time format (**/View/Display Time Format**). Some common settings are 30 (SMPTE non-drop), 24 (film sync), and 25 (EBU).

Display boundary lines at (dB)

This designates the position at which the boundary lines in the waveform display are positioned. A value of 0 dB will display the boundaries at the maximum amplitude value possible, before clipping would occur.

By setting a boundary of -1dB or so, and keeping your audio within it, you can maintain some headroom which can come in useful when applying audio transformation functions.

Minimum Preview Buffer Size

This is the buffer size to be used when sending data to your sound card for the real-time Preview found in many effect dialogs. Different sound card drivers can require different memory buffer settings. If you hear "choppiness" (skips or dropouts) in Preview, you can try to remedy it by adjusting the buffer size used (though this can also result from insufficient processing power). A greater buffer size will require more of your computer's memory.

Settings/System

Keep in mind that computers will often vary from one system to the next (depending upon their CPU speed, hard disk capacity and supporting hardware). It's the intention of *Cool Edit Pro* Edit that your data be processed as fast as possible... however, if your system simply can't keep up with the task that you've given it, you can change the system variables to optimize *Cool Edit* to your current system.

Total Buffer Size (seconds)

Different sound card drivers can require different memory buffer settings. *Cool Edit Pro*'s default settings should work fine for most sound cards, but if you hear "choppiness" (skips or dropouts) in recording or playback, you may need to adjust the buffer size or number of buffers used. Use this field to reserve memory for recording and playback by entering a buffer size (in seconds). A greater buffer size will allow for increased multitasking while audio is being played, at the expense of taking more of your computer's memory.

If you do experience break-up in your audio, or you cannot Stop after you have started recording, increase the buffer size, or switch to a faster (non-compressed) hard drive.

Number of Buffers

Different sound card drivers can require different memory buffer settings. *Cool Edit Pro*'s default settings should work fine for most sound cards, but if you hear "choppiness" (skips or dropouts) in recording or playback, you may need to adjust the buffer size or number of buffers used. Use this field to adjust the number of buffers.

If you do experience break-up in your audio, try reducing the number of buffers. Increasing the number of buffers may also help for some configurations.

Cache Size

Cool Edit Pro maintains its own data buffer, and reserves for it the amount of memory specified in this field. Recommended cache sizes are from 1024 kb to 4096 kb. Cache sizes greater than about 4096 kb will tend not to increase speed notably for most processing, and sizes below about 1024 kb will tend to slow processing down.

Use System's Cache

Check this option to let Windows handle all disk caching. Keep in mind that *Cool Edit Pro* usually handles caching better than Windows can. However, this option reserves the least amount of memory, so it 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.

Peaks Cache

This determines the number of samples per block to be used when storing peak (.pk) files. Larger values will reduce the RAM requirement for large files at the expense of slightly slower drawing at some zoom levels. If RAM is an issue on your system, and 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.

Save Peak Cache Files

Peak files store information about how to display .WAV files, and can make file loading almost instantaneous by greatly reducing the time it takes to draw the waveform (especially with larger files). If enabled, all .WAV files will have peak files saved with them (in the same directory) with the extension .PK following the original audio file name. Un-check this box if you do not wish to have *Cool Edit Pro* save peak files on your hard drive.

You can safely delete peak files (.pk) from your hard drive at any time. If the **Save Peak Cache Files** option is checked, a new peak file will be created in such a case.

Rebuild Wave Display Now

Press this button to force *Cool Edit Pro* to rescan the current file for sample amplitudes, and redraw the waveform.

Temporary Directories

Cool Edit Pro creates temporary files for use when performing edits on your audio. Use these two fields to specify the paths to the directories in which these files will reside. You will need to have enough space available in these directories to accommodate the total size of all the audio files you wish to edit simultaneously. You can also specify an amount to leave "free" for headroom purposes for both the primary and secondary directories.

For best results, select separate physical hard drives for the primary and secondary Temp Directories.

Enable Undo

This option enables/disables the Undo function. Because Undo requires extra disk space for its temporary files, and time to save them before processing, you may wish to disable the Undo feature. 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.

Undo Levels (minimum)

This number specifies how many levels of Undo (how many edits you can go back through) that *Cool Edit Pro* will save. This is a minimum figure; more Undo levels will be created if there is enough disk space available. If space is at a minimum, Undo levels will be removed as necessary. If *Cool Edit Pro* must remove Undo levels beyond this minimum setting, it will warn you about it, and give an option to cancel the operation.

Purge Undo

Press this button to erase all Undo levels below the specified minimum, freeing up the hard disk space used by them. For example, if you have 5 levels of Undo set, Purge Undo will delete 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 perform the next operation on the waveform. You can also remove all undo levels by entering 0 for Undo Levels and then pressing Purge to expunge the entire undo history.

Settings/Colors

Color Scheme Presets

Most color assignments in *Cool Edit Pro* can be adjusted to your preference, and saved as a preset for later recall. This drop down menu contains all of the available presets, to choose one simply select it from the list. The currently selected color scheme will be displayed in the Example window

Save As…

Press Save As… to save the current color scheme as a preset.

Delete

This button deletes the currently selected preset.

Display Element

This list contains all elements which can be can be given custom colors. Choose a display element and press Change Color… to choose a new color for that element. The example to the right will change to reflect the newly chosen color.

Example

This area shows the elements within the waveform display, and their current color. The example will change to reflect the current preset, or changes made to any display elements.

Change Color

This button brings up a standard Windows color selection dialog. Use this dialog to choose a color for the currently highlighted item in the Display Element list.

Settings/Spectral

Windowing Function

This is the function that will be used to window the data before being displayed. In general, you can keep this at Blackmann or Blackmann-Harris. The windows are listed in order from the narrowest frequency band/most noise to the widest frequency band/least extra noise.

Resolution (bands)

This setting specifies the number of vertical bands to be used in drawing frequencies. Values of around 256 to 512 give good resolution while not taking too long to draw.

Window Width

This is the width of the window used in plotting the spectral data. Generally this is kept at 100%.

Spectral Plot Style

Logarithmic Energy Plot: In this mode, colors change with the decibel value of the energy at any particular time and frequency. More details in the very quiet ranges can be seen in this mode, especially if the Range is quite high (above 150dB). Use the range value to adjust the sensitivity in plotting frequencies.

Linear Energy Plot: In the Linear mode, colors are chosen based on percentage of maximum amplitude instead of decibel amplitude. Linear Energy Plot can be useful for viewing the general overview of a signal without getting bogged down by detail at much quieter levels. The scaling factor can be adjusted to highlight audio of different intensities, and can be thought of as a sensitivity value.

Reverse Color Spectrum Direction

Check this box to reverse the "direction" in which the display changes colors from low to high frequencies.

Settings/Data

Auto-convert all data to 32-bit upon opening

In this mode, data is converted to 32-bit when a file is loaded, and all subsequent operations will keep the data in the 32-bit realm. Data will be converted back to 16-bit when saved.

Dither Transform Results

Check this option to enable dithering when processing effects and transforms, such as FFT Filter or Amplify. Most processing done by *Cool Edit Pro* uses arithmetic greater than 16-bit, with the results converted back to 16-bit when complete. During this conversion, dithering provides a higher dynamic range and cleaner results, with less distortions and negative artifacts. If this option is disabled, the results are truncated to 16 bits when converting back, thus losing the more subtle information. When enabled, the addition of dither retains this subtle information. The drawback is that with each operation a small amount of white noise is added at the quietest volume level. 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 another 10dB or so, allowing signals as quiet as -105dB.

Smooth Delete and Cut Boundaries

Check this box to have Cut and Delete operations smoothed at the splicing point. This will prevent audible clicks at these locations.

Smooth all edit boundaries by crossfading

When applying a transform, enable this option to automatically apply a crossfade to the starting and ending boundaries of the selection. This will smooth any abrupt transitions at these endpoints. You can enter a value (in milliseconds) in Crossfade Time to specify the crossfade duration to be applied.

This option prevents audible clicks when filtering small portions of audio.

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.

Downsampling Quality Level: Enter a value (30 to 1000) for downsampling quality. Higher values retain more high frequencies while still preventing the aliasing of higher frequencies to lower ones. A lower quality setting requires less processing time, but will result in certain high frequencies being "rolled-off", leading to muffled-sounding audio. Because the filter's cutoff slope is much steeper at higher quality settings, the chance of ringing at high frequencies is greater (frequencies just below the Nyquist may be abnormally boosted in level). Usually values between 100 and 400 do a great job for most conversion needs.

Pre Filter: To prevent any chance of aliasing, the pre-filter on downsampling will remove all frequencies above the Nyquist limit, thus keeping them from generating false frequencies at the low end of the spectrum. In general, this option should be enabled for best results.

Upsampling Quality Level (30 to 1000): Enter a value (30 to 1000) for upsampling quality. Higher values retain more high frequencies while still preventing the aliasing of higher frequencies to lower ones. A lower quality setting requires less processing time, but will result in certain high frequencies being "rolled-off", leading to muffled-sounding audio. Because the filter's cutoff slope is much steeper at higher quality settings, the chance of ringing at high frequencies is greater (frequencies just below the Nyquist may be abnormally boosted in level). Usually values between 100 and 400 do a great job for most conversion needs.

You should use high quality settings whenever you downsample from a high sample rate to a low rate. When upsampling, a low quality setting will sound almost identical to a high quality setting. The difference lies in the larger phase shift that exists at higher frequencies, but since the phase shift is completely linear, it's very difficult to notice. Downsampling, at even the lowest quality setting, generally won't introduce any undesired noisy artifacts. Instead, it may just sound slightly muffled because of the increased high end filtering.

Post Filter: To prevent any chance of aliasing, the post-filter on upsampling will remove all frequencies above the Nyquist limit, thus keeping them from generating false frequencies at the low end of the spectrum. In general, this option should be enabled for best results.

Dither amount for saving 32-bit data to 16-bit files (0 to 1)

This option enables/disables dithering when saving 32-bit audio to 16-bit. A value of 1 (bit) will enable dithering , while a value of 0 will disable dithering. For semi-dithering, choose a value of 0.5. With dithering, you get almost 24-bit sample performance in only 16-bits, as the dynamic range is increased by another 10dB or so, allowing signals as quiet as -105dB.

Settings/Multitrack

Playback Response Time (buffer size)

This is the buffer size to be used when sending data to your sound card when playing back a multitrack Session. Different sound card drivers can require different memory buffer size settings. *Cool Edit Pro's* default settings should work fine for most sound cards, but if you hear "choppiness" (skips or dropouts) in multitrack playback, you can try to remedy it by adjusting the buffer size used (though choppiness in multitrack playback can also be attributed to the *background mixing* process not being far enough ahead). A greater buffer size will require more of your computer's memory. The default setting is .08.

Playback Buffers

Different sound card drivers can require different memory buffer settings. *Cool Edit Pro's* default settings should work fine for most sound cards, but if you hear "choppiness" (skips or dropouts) in recording or playback, you may need to adjust the buffer size or number of buffers used. Use this field to adjust the number of buffers used for playback in the multitrack environment. If you do experience break-up in your audio, try reducing the number of buffers. Increasing the number of buffers may also help for some configurations. The default setting is 10.

Recording Buffer Size

Use this field to reserve memory for recording in a multitrack Session by entering a buffer size (in seconds). Different sound card drivers can require different memory buffer size settings. *Cool Edit Pro's* default settings should work fine for most sound cards. If you experience dropout while recording in multitrack (especially when playback seems fine), try increasing this setting (first be sure the *background mixing* process is sufficiently complete when you go to record as this may cause the same symptom). A greater buffer size will require more of your computer's memory. The default setting is 2.

Recording Buffers

Different sound card drivers can require different memory buffer settings. *Cool Edit Pro*'s default settings should work fine for most sound cards, but if you hear "choppiness" (skips or dropouts) in recording or playback, you may need to adjust the buffer size or number of buffers used. Use this field to adjust the number of buffers used for recording in the multitrack environment. If you do experience break-up in your audio, try reducing the number of buffers. Increasing the number of buffers may also help for some configurations. The default setting is 10.

Background Mixing Priority Level

This setting assigns a level of priority to the background mixing process done in a multitrack session. Lower values indicate a higher level of priority above other system events. You can enter fractional numbers, as in 0.8, the default setting.

Delete old takes after merging

Check this box to have *Cool Edit Pro* automatically delete any unused takes created during a Punch-In when you select a take to go with (this is done by choosing Merge Current Take from /Edit/Take History in the Multitrack View). If this setting is off, unused takes remain available to the Session (in the Insert menu) and occupy hard drive space.

Crossfade time (ms)

This is the amount of time over which crossfading occurs when a take created using Punch-In is merged back into the surrounding waveform.

Playback Mixing

This is the bit size used for the background mixing process. Best quality is achieved by leaving this at the default 32-bit setting, however if using multiple sound cards, it may be advantageous to choose 16-bit for Playback Mixing as less data will be transferred across the hard drive(s), speeding things up. For single output device situations, or faster hard drives, 32-bit is better as it provides optimization at mix down.

Mixdowns

When performing a Mixdown (merging the selected, or all, waveforms into a single waveform by choosing /Edit/Mixdown from the Multitrack View), this is the bit-resolution that will be used. Regardless of the session format (16 or 32 bit), Mixdowns can be generated at either 16-bit or 32-bit quality with this option. The default is 16-bit.

Minimum track height for full waveform display

This setting determines the minimum track height at which waveform data will still be drawn in the Multitrack View. When viewing a large number of tracks in the waveform display, or zooming out on a session, the height of individual tracks diminishes. At a certain point, trying to display the waveform data in each track may slow things down or be undesirable (and if zoomed out far enough, will not be possible). This is the minimum height, in pixels, at which attempt to display waveforms.

Settings/SMPTE

Lead Time (ms): This sets is the amount of time *Cool Edit Pro* has to establish sync with the incoming Time Code. Lower settings (200ms) will result in faster response (when pressing Play), but at the risk of true sync not being established. 500 to 1000 ms should provide plenty of Lead Time to establish lock-up.

Stopping Time (ms): Stopping Time is the amount of time *Cool Edit Pro* will continue playing if a drop-out in the Time Code is encountered.

Lag Time (samples): This setting fine tunes the relationship between Time Code coming in and audio data coming out, and can be used to make up for any discrepancies introduced by your sound card buffers.

Slack (frames): This sets the number of frames *Cool Edit Pro* can periodically be "out of sync" with the time code before either repositioning the playback cursor to match the code, or perform a full re-sync. Generally a setting of about 2.5 frames will do, as there are occasions when the time code frame location numbers may be incorrect, usually corrected on the next frame sent. This will not cause a sync problem in the audio sense.

Reposition playback cursor when slack used: If checked, the playback position will readjust if the sync is off by the Slack amount.

Full re-sync when slack used: If checked, a full re-sync will be performed if the sync is off by the Slack amount.

Settings/Devices

If you have multiple sound cards, or a single card that has multiple outputs, you can use the Devices tab to choose the input and output devices you would like *Cool Edit Pro* to be assigned to. If your system is equipped with MIDI devices, you can also choose the MIDI in, and MIDI out sources, using this setup option. The capabilities of the recording and playback devices will be displayed in the given tables.

These settings will be remembered in the [*Cool Edit*] section of Window's WIN.INI file, which means if you install a new sound driver or card, Cool will not access it until you choose it from this dialog.

MIDI In (Sync/Trigger)

This setting is used to assign a MIDI source to *Cool Edit Pro*'s MIDI In Sync/Trigger input. The Sync source will provide a MIDI Time Code source that *Cool Edit* can synchronize to, while a trigger source can be used to transmit MIDI not on messages for triggering individually assigned waveforms.

MIDI Out (Music preview)

This setting is used to assign *Cool Edit Pro*'s Music Preview function (see **/Transform/Special/Music**) to a MIDI destination (such as a sequencer or MIDI instrument).

Waveform Playback

Use this determines the sound card, or outputs to assign to *Cool Edit Pro* for playback.

Waveform Record

Specifies the device to be used when recording sound data.

Selecting a mapper device such as the Wave Mapper or Sound Mapper allows Windows to select an appropriate device that is to be used for the current sound data.

Play 16-bit files as 8-bit

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

Shortcuts (Keyboard and MIDI Triggers)

Shortcuts are used to speed up the editing process by allowing you to execute almost any command in *Cool Edit Pro* from your computer and/or MIDI keyboard. For example, instead of using your mouse to go to the Edit menu and selecting Cut to cut the highlighted portion of a waveform, you can simply press Ctrl+X (this labeling technique translates to “hold the first key down [Ctrl], and press the second [X]). Likewise, you can execute commands from a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command. This type of Shortcut is referred to as a MIDI Trigger. An example of this would be assigning the Play command in *Cool Edit Pro* to the C4 note on your MIDI keyboard.

Before attempting to enable MIDI triggering, you will need to choose a device for MIDI In that is recognized by Windows (such as a sound card's built-in MIDI interface, a MIDI interface card or other hardware device options.) To do this, go to **/Options/Settings/Devices** to bring up the devices window. Set MIDI In (Sync/Trigger) to the input and to be used for Triggering (commonly YourDevice: In 1).

When using *Cool Edit Pro* with a MIDI sequencer, you can trigger audio from your MIDI sequence by embedding a controller event at a specific location, and assigning playback in *Cool Edit Pro* to that event. This works well for shorter samples, but for longer audio segments you should use SMPTE to ensure synchronization.

This list displays all of the assignable commands in *Cool Edit Pro*. Click on a command name in the list to see or change its' Shortcut.

Shortcut Key

Use this section if you wish to be able to execute the currently highlighted command from you computer's keyboard. You can select from the Ctrl, Shift, and Alt keys, or any combination of the three, to be used as the first half of the Shortcut combination (the part to hold down), and the second half of the Shortcut (a single key) can be typed in or selected from the drop down list.

MIDI Trigger

Checkmark Enable Trigger to have the currently highlighted command assigned to a MIDI event. You can select the MIDI channel to receive the trigger on from the drop down list to the right (default is 1). The type of MIDI event used for the Trigger is selected by choosing either Note or Controller. For Note events, you can select the note number (middle C is C4 for example) from the drop down list. For Controller events, type in the Controller number and Value to the right (for on/off controllers, such as Hold Pedal [64], values of 0 and 127 represent on and off, respectively).

Conflicting Keys

If a command's Shortcut key combination is already in use, you will be informed here.

Toolbars

The Toolbar options allow for only the toolbars that relate to your everyday work to be displayed within *Cool Edit Pro*s main screen. These toolbars include: **File**, **Edit**, **View** and **Options** toolbars.

Windows Menu (Edit View)

When you are working in Edit View mode, the Windows menu gives you a simple way to navigate between audio file waveforms and to open new waveforms in *Cool Edit Pro*.

Waveforms List

When you are working in the Edit View window, the Waveforms List works as a simple navigation tool for opening, closing, and switching between audio files. You can also use Waveforms List to convert audio files into the appropriate format for the current project.

Waveform List options include:

Waveform List

This displays the audio files or waveforms that are currently open in *Cool Edit Pro*. Click on any item in the list to select it, or double-click to switch to that waveform in the Edit View window. To select more than one item in the list, hold down the SHIFT key for contiguous selection, or the CTRL key for non-contiguous selection.

Switch To

Actively places the selected waveform into the Edit View window.

Close Wave

Closes the selected waveform and removes it from the Waveforms List.

Insert

Inserts the selected waveform(s) into the Multitrack View window, so that it can be mixed within the multitrack environment. *Cool Edit Pro* places the selected items into the next available track(s). If the selected audio file is not of the same sample type as those in the current session, the Convert Sample Type dialog box will pop up to allow you to convert the file to the appropriate sample rate/file type.

Open

Lets you browse for and open a previously recorded audio file.

Full Paths

Displays the full DOS path for files in the Waveform List.

Close

Closes the Waveforms List dialog box.

File List

This list simply provides a quick way to switch between audio files that are currently open in *Cool Edit Pro*. You'll find a check mark next to the file that's currently loaded in the Edit View window.

Open a series of Audio files and skip between open windows using this function. Easy, Huh?

Help Menu

Use this menu to access *Cool Edit Pro*'s on-line help.

Contents

Click on this topic to bring up *Cool Edit Pro*'s Help Index.

Quick Reference…

This Quick Reference **Index** gives you quick access to most of *Cool Edit Pro*'s operating features. Using the **Find** dialog box, you can search for information on topics using a single key word.

Overview…

Choose this option to see a general description of *Cool Edit Pro* and its features.

Multitrack Editing…

This Help topic will give you tips and guidelines on operating in a multitrack environment.

Search for Help On…

Use this option to search for a specific Help topic by keyword.

Tip of the Day

You can access *Cool Edit Pro's* "Tip of the Day" feature through this option.

Syntrillium on the Web

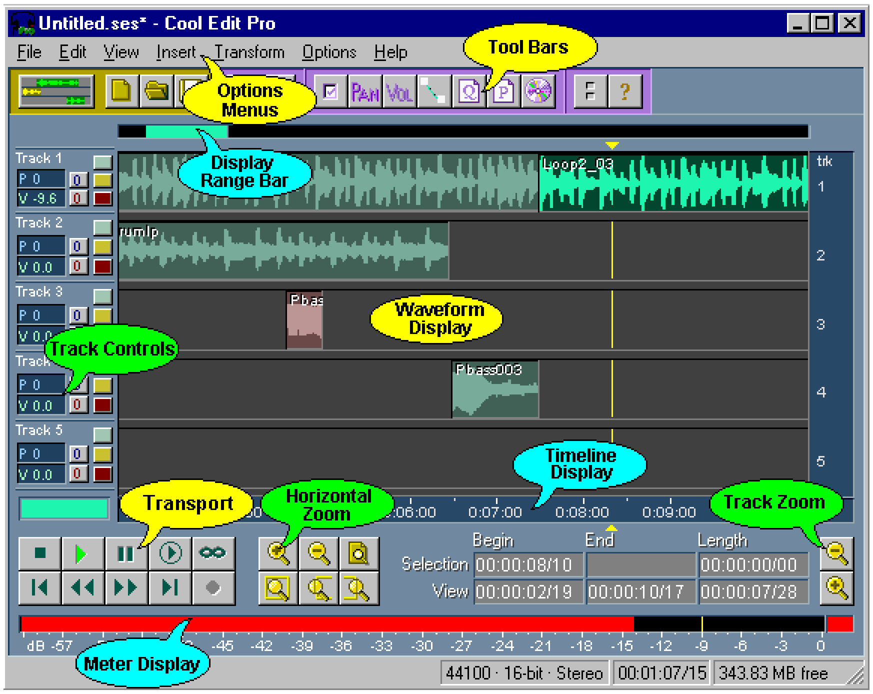
Use this option to access Syntrillium's web site at http://www.syntrillium.com and find out what's new!

About Cool Edit Pro

This dialog lets you know what software version of *Cool Edit Pro* is currently loaded onto your computer and to whom it is licensed, as well as how to contact us (the folks at Syntrillium).

Navigating the Multitrack View screen

The following is a typical layout for the *Cool Edit Pro*'s **Multitrack View Screen**:



Multitrack View

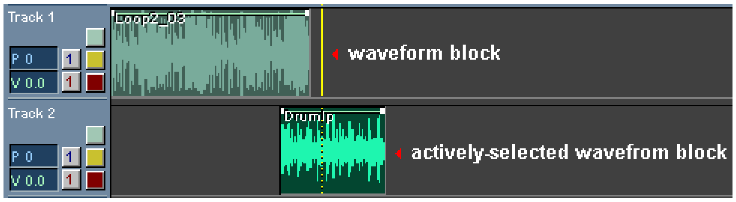
What's an Image…Image…?

When you are working in the Multitrack Mode and you move a waveform from one track and place to another, or create a series of duplicates for looping, you aren't actually moving the original audio file data in any way. In fact, you are manipulating a virtual "image" of that waveform or waveform range in a non-destructive fashion that doesn't effect the original audio file in any way.

Essentially, an image is a set of playback instructions that are tagged to the original audio file(s). For example, creating a series of 16 loops within the Multitrack View window doesn't actually waste disk space by making sixteen copies of the original waveform. Instead, the session file will instruct the program to playback the same portion of the original audio file (the image) 16 times in a repeating fashion. This concept lies at the heart of modern-day hard disk editing.

What's a Waveform Block?

Waveform blocks are actually images that are graphically displayed within the Multitrack Waveform window that allow you to mix, move, copy, loop and alter volume and pan levels in a non-destructive fashion (non-destructive meaning that you can alter image parameters without effecting the original waveform data in any way.)



Waveform Blocks

You can right-click on a waveform block to call up a menu that contains shortcut commands relating to multitrack editing.

Waveforms List (Insert Menu)

The first thing that you want to do when starting a session is to begin loading waveforms into the Waveform Display. What's a session, you ask?... Basically, a "session" uses the (*.ses) file structure to save all of the information that relates to a current multitrack project (such as audio file placement, mix/pan info, mute/solo info… you name it!)

When working in the **Multitrack View** window, the **Waveform List** works as a simple navigation tool for loading audio files into a multitrack session. Basically, it can be thought of as a central "holding tank" from which audio files can be easily selected and placed into the **Multitrack View** window as a waveform block.

Looping

/Edit/Loop Duplicate will duplicate a specified number of copies of the waveform block that has been selected in a consecutively repeating fashion. Alternately, you can specify a spacing (like every second) to place the block, so that specifying 9 copies with a spacing of 10 seconds will copy the selected wave block and paste it in 9 times, with each being spaced 10 seconds apart.

Punch Ins

When a particularly difficult passage is being recorded, it's not uncommon for a note or words to be missed or totally screwed up (to err is human)... no big deal! You can either stop and pick up from before the mishap, or you can continue on (as though nothing had happened) and record over the mishap at a later time (a process known as a **punch in**.)

Punching in using *Cool Edit Pro* is particularly easy. Simply highlight the area that you would like to record over, select the **/Edit/Punch In** option from the menu (or right click on the waveform block and choose **Punch In**. By placing he desired track into the Record Ready mode, placing the playback cursor at a convenient point before the "punch" is to occur, you're ready to record the proper notes over the previous mishap.

Whenever one or more **Punch Ins** have been performed within a waveblock, a **Take History** option will be activated (both **Punch Ins** and **Take History** are available by right-clicking on the waveform block.) The **Take History** option lets you revert the waveform block back to any previously-available take level (essentially acting as a selective undo function -

allowing you to choose any change level between its original un-punched state and the current waveform level.

Splice

The **Splice** option creates a non-destructive "break" within a waveform block at the current cursor point. Once the block has been "spliced", each portion of the newly created blocks can be moved, deleted, slid in time/track, etc. with complete freedom.

Snapping

Often, when working with multiple waveforms in a multitrack computer-based editing environment, there is the need to place waveforms into a session with extreme accuracy. For example, let's say that we have a series of audio files that we have inserted into a session, and these waveforms have to be consecutively placed (one after the other) so that no gaps exist between them. We could zoom out and place each splice point close together and then zoom in on each edit and manually "butt" each file against the other... one at a time! A process that would not only be time consuming, but a bit frustrating... or we could simply use *Cool Edit Pro*'s waveform "**snap**" function.

In this case, the term "snap" simply means that whenever a waveform is placed within a certain distance of a defined boundary, the waveform's beginning point will "jump" or "snap" precisely to that boundary point. So, getting back to our consecutive files, all we'd need to do is place the files into the **Multitrack View** window, turn on one of the snap to waves option (if its not already on) and then butt each file up to the previous one. As the two files approach each other's boundaries, you'll see then visually snap to each other (as though they were magnetized). All that's needed is to play the combined results of your quick-n-simple edit session.

Track/Wave Properties (Right-Click)

Right-clicking on either the track or on a waveform within the Multitrack View window will respectively invoke either the **Track Properties** or **Wave Properties** dialog box. The property options include:

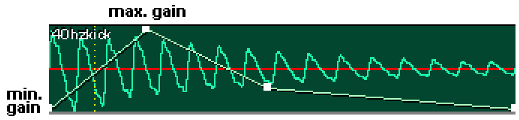
Edit Envelopes Mode

The **Edit Envelopes Mode** allows *Cool Edit Pro* to non-destructively re-calculate audio file volume changes in real-time, allowing for Level and Pan positioning to be changed as session audio files are being played. This feature allows for waveforms within *Cool Edit Pro*'s **Multitrack View** window to be mixed in a virtual fashion (either to a single sound card, or to a number of sound card output destinations.) The **Edit Envelopes Mode** offers two types of real-time envelope drawing: **Volume Envelopes** and **Pan Envelopes**.

Volume Envelopes

This option lets you draw volume envelopes that range from minimum gain (bottom portion of the audio file's waveform display) to maximum gain (top portion of the waveform display). When activated by the **Enable Envelope Editing** option, volume changes can be drawn into each waveform window by simply clicking anywhere on the light green volume line and moving the line to the desired gain position. All volume changes are totally non-destructive (meaning that the original audio file data isn't effected by these calculations) as they are processed in real time as the session track is being played back.

It's important to remember that real-time volume changes will require some number-crunching on your computer's part. Mixing a large number of tracks in real-time may "bog down" your computers main processor, causing "jumps" or "digital glitches" during playback. If this happens, you may want to use the **/Edit/Mixdown** function to create a submix that effectively "bounces" several tracks down to a single track or stereo pair of tracks.



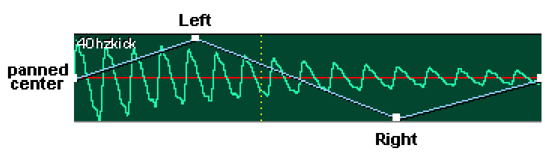
Amplitude Envelope

Pan Envelopes

This option lets you draw panning envelopes that range from Left (top portion of the audio file's waveform display) to the right (bottom portion of the waveform display). When activated by the **Enable Envelope Editing** option, pans can be drawn into each waveform window by simply clicking anywhere on the light blue pan line and moving the line to the desired L/R pan position. All pan calculations are totally non-destructive (meaning that the original audio file data isn't effected by these changes) as they are processed in real time as the session track is being played back.

Pan changes can be directly entered into the track by holding down the left and right mouse buttons and then sliding the mouse vertically.

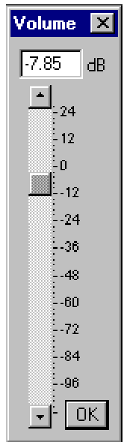
It's important to remember that real-time pan calculations will require some number-crunching on your computer's part. Mixing a large number of tracks in real-time may "bog down" your computers main processor, causing "jumps" or "digital glitches" during playback. If this happens, you may want to use the **/Edit/Mixdown** function to create a submix that effectively "bounces" several tracks down to a single track or stereo pair of tracks.



Pan Envelope

Track Volume Control

This option allows you to set the overall volume (gain) for the selected track, ranging from infinity (-100 dB) to +20 dB. The Track Volume fader can be easily called up by left-clicking on the appropriate track within the waveform display and selecting the **Volume...** option, or by right-clicking on the **Track Volume** control icon V0.0 (which will display the track's volume fader).



Volume Dialog

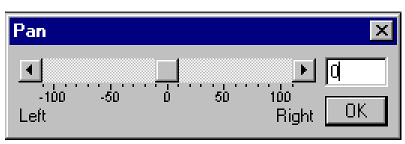
Volume changes can be directly entered into the track by holding down the left mouse button and then sliding the mouse vertically.

Whenever the numeric volume readout is highlighted, the track volume can be easily set to "unity gain" by simply pressing "0".

Keep in mind that real-time volume changes will require some number-crunching on your computer's part. Mixing a large number of tracks in real-time may "bog down" your computers main processor, causing "jumps" or "digital glitches" during playback. If this happens, you may want to use the **/Edit/Mixdown** function to create a submix that effectively "bounces" several tracks down to a single track or stereo pair of tracks.

Track Pan Control

This option lets you vary the overall L/R pan controls (for a mono waveform) or stereo balance (for a stereo waveform) for the selected track, and ranges from hard-left (-100 dB) to hard-right (+100 dB). The Track Pan control can be easily called up by left-clicking on the appropriate track within the waveform display and selecting the **Pan...** option, or by right-clicking on the **Track Pan** control icon P 0 (which will display the track's pan positioner (also known as a pan pot).



Pan Dialog

Pan changes can be directly entered into the track by holding down the left mouse button and then sliding the mouse vertically.

Whenever the numeric volume readout is highlighted, the track's pan position can be easily set to the center position by simply pressing "0".

Keep in mind that real-time pan calculations will require some number-crunching on your computer's part. Mixing a large number of tracks in real-time may "bog down" your computers main processor, causing "jumps" or "digital glitches" during playback. If this happens, you may want to use the **/Edit/Mixdown** function to create a submix that effectively "bounces" several tracks down to a single track or stereo pair of tracks.

Track Controls

Here's a brief description of the controls in the Multitrack window:

Mute Track

Solo Track

Record Enable Track

Playback Device - left click to choose device

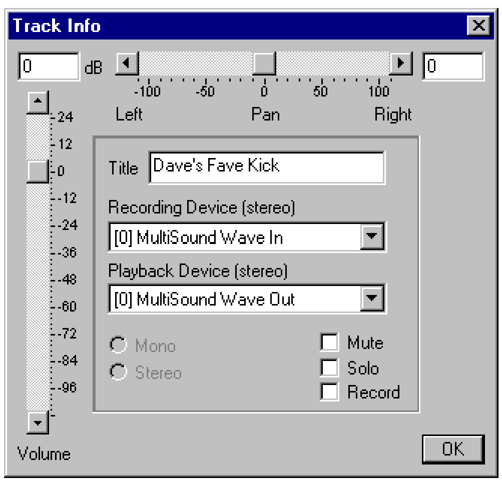
Record Device - left click to choose device

Track 1 **Track Name** - left click for track information

P 0 **Track Pan** - right click for slider or left-click and hold for direct adjustment (adjustments are made by sliding the mouse vertically)

V 0.0 **Track Volume** - right click for slider or left-click and hold for direct adjustment (adjustments are made by sliding the mouse vertically)

 Left-click within the **blue Track Name** area to call up the **Track Info** dialog, which allows you to directly assign such information and gain-change variables to a track as:



Track Info Dialog

Mix Gauge

The **green** Mix Gauge (which is located at the left-hand side of the Multitrack View screen) indicates how much background processing *Cool Edit Pro* has completed. In a sense, it can be thought of as a fuel tank gauge, except that it measures how much processing needs to be carried out by *Cool Edit Pro* behind the scenes. The further that the Mix Gauge's **green** bar progresses to the right the more processing has been done to complete its current mixing tasks. Once the processing has been completed, the bar lights up to a bright shade of **green** (meaning the entire session is fully mixed).

Different States of Mix Gauge

Some notes on the background mixing process: Mixing involves combining all of the placed waveforms into two (or more) channels for output. Because you can edit, add, and subtract waveforms in the multitrack environment, *Cool Edit Pro* must constantly watch for changes to the multitrack session, such as a moved or deleted waveform, a volume change, or new material recorded into a track. When something is altered, *Cool Edit Pro* must immediately work that change into the mixed output, and it does so through background mixing. Background mixing occurs behind the scenes much of the time that you are in the multitrack environment, and is generally very fast. The faster your system—especially your CPU and hard drive(s)—the faster *Cool Edit Pro* can mix in the background. The mix progress meter constantly shows how far along *Cool Edit Pro* in the process of mixing your session. However, you need not wait for the meter to reach completion entirely before beginning playback. As stated above, *Cool Edit Pro* continuously mixes, and will continue to do so while playing, so you can safely begin playback when the mix progress meter is about half-way up. In general, if the background mix is not sufficiently completed, you will hear a break-up or skipping when playing back the mix. If that happens, just wait a few seconds; *Cool Edit Pro* will usually "catch up" very quickly, and you can begin playing again.

Track Display Ruler

The **Blue Track Display Ruler** displays the track number (or range of track numbers) for easily placing waveforms into the current session. The range of tracks that can be visible at any one time can easily be changed, using the following mouse commands:

Left-click and dragging the little hand vertically over the **Blue Track Display Ruler** will allow you to scroll through *Cool Edit Pro*'s visible track ranges (from tracks 1 - 64!)

The number of tracks that will be visible at one time will remain the same, however the track ranges being viewed will be shifted either up or down.

Right-click and dragging the magnifying glass over the **Blue Track Display Ruler** will automatically zoom the waveform window to show only the specified tracks.

Whenever the waveform display is zoomed out to a level where your monitor's resolution cannot accurately show waveform with sufficient detail, the waveform itself will not be displayed. Instead, the waveform's title will be clearly displayed within its range boundaries. The waveform's graphic detail will again re-appear whenever the track display is Zoomed in (to display fewer tracks at any one time.)

Right-click on the **Blue Track Display Ruler** to call up a pop-up dialog window that lets you to quickly and easily select various Zoom options.

File Menu (Multitrack View)

The File menu offers functions relating to opening, saving, and otherwise manipulating session files.

What's a Session?

Before we even begin, you'll need to know that a "session" uses the (*.ses) file structure to save all of the information that relates to a current multitrack project (such as audio file placement, mix/pan info, mute/solo info... you name it!). All of your multitrack masterpieces are saved to and recalled from disk as a session (*.ses) file.

Keep in mind that your session files save information that relates to a current multitrack project and doesn't save the waveform (i.e.: *.wav) files within these files. It's important to keep your waveform files intact and within their original directory... And remember, whenever possible, **BACK YOUR IMPORTANT DATA UP!**

New Session (Ctrl+N)

New Session… creates a new, empty **Session,** where you record or insert audio into multiple tracks. When creating a new Session, you will be prompted to specify its properties: sample rate, and bit-resolution. Any audio files to be used in a session must share these properties, or be converted upon inserting them.

Sample Rate

The sampling rate you choose will determine the overall bandwidth (how many frequencies can be encoded within the audio signal), with higher sampling rates yielding a wider bandwidth. You may enter any sample rate directly, or choose a common sample rate from the list. Note that most sound cards support only certain sample rates.

Resolution

Select the bit resolution to use for the Session:

8-bit:	for use in telephony/Internet applications
16-bit:	standard professional/CD quality
32-bit:	this signal processing mode is supported by some high end sound cards and other equipment

To achieve higher quality, you may want to work at the 32-bit level while processing your audio. Since current sound cards cannot playback audio that has been stored and processed using the full 32-bits, this rate must be converted downward for playback. It's best to use this mode when using audio cards that have resolutions greater than 16-bit (20-bits as of this writing).

Certain combinations of sample rate, channels, and resolution may not be available on your system. To see the capabilities of your system, check the Devices tab under **/Options/Settings**. Although *Cool Edit Pro* can create and edit those files, your sound card may not be able to play them properly.

Open Session

Opens an existing Session file (.SES), along with all audio files used in the Session.

Recent Directories

This displays the most recently used directories, for quicker access to common locations.

Look In

Navigates through all available drives and directories.

File Name

You can specify the name of the file you wish to open here, if it resides in the current directory.

Files of Type

Select from the list of supported file types to display any files with a corresponding extension.

Append to Session

This opens an existing Session and all associated audio files, and begins track placement at the end of the currently open Session. For example, if the currently open session has material in tracks 1-4, then using **Append to Session**… to open another four track session would place the new material into tracks 5-8. The selected Session must have the same file properties (sample rate, bit resolution) as the Session to be appended to.

Recent Directories

This displays the most recently used directories, for quicker access to common locations.

Look In

Navigates through all available drives and directories.

File Name

You can specify the name of the file you wish to open here, if it resides in the current directory.

Files of Type

Select from the list of supported file types to display any files with a corresponding extension.

You should note that the appended tracks are loaded into the session at the beginning of the project's timeline (i.e.: all of the tracks will load in at the project from its beginning point.) Once loaded, the appended waveforms can easily be moved into place by actively selecting the desired waveforms and by right-clicking on any one of them and then dragging them into position.

If you'd like to add a previously created session to your current session, this can easily be done using the **/File/Append to Session** command. For example, lets say that we'd like to insert an existing multitrack composition that's been saved as a session. Appending the desired session into our current workspace (which will place them at the beginning of the timeline, onto the next available set of open tracks) we can now select and move the appropriate session waveforms into place (often using the **Ctrl-Right Click** mouse command to select multiple waveforms)... and Voila... you've integrated two sessions into a whole new composition.

Close Session

Closes the Multitrack Session currently in use. You will be asked if you would like to save the file if not saved since the last modification.

Close Session and Waveforms

Closes the Multitrack Session currently in use, and also closes all wave files in that session. You will be asked if you would like to save the file if not saved since the last modification.

Save Session

Saves the current Session back to disk, overwriting the original without confirmation. If the Session does not yet exist as a file (not previously saved), you will presented with the **Save Session As**… window.

Save Session As

Saves the current Session to a new name or location, along with all wave files used in the Session.

Recent Directories

This displays the most recently used directories, for quicker access to common locations.

Look In

Navigates through all available drives and directories.

File Name

Specify a name for the Session.

Files of Type

Select from the list of supported file types to display any files with a corresponding extension.

Most Recently Used File List

The Most Recently Used (MRU) file list displays the eight files that were most recently opened by *Cool Edit Pro*. You can quickly re-open any of these files by selecting one from the list with the mouse or by typing the number associated with the desired file.

Exit

Closes *Cool Edit Pro*. At close time, *Cool Edit Pro* asks if you want to save any unsaved files that were modified. Any temporary files that were created will be removed.

Edit Menu (Multitrack View)

The edit menu displays all of the options that relate to basic waveform editing in the Multitrack View mode. These options are:

Wave Block Info...

Selecting a Waveform Block from within the Multitrack Waveform View window and selecting Edit/Wave Block Info... will inform you as to the source audio file (from which the waveform block derives its audio data), as well as allowing you to define certain placement and mixing parameters that relate specifically to this soundblock.

Volume Levels

Allows you to set volume levels for the selected waveform block.

Pan Position

Allows you to set relative L/R pan positions for the selected waveform block.

File Name

Displays the waveform block's original source audio file (from which the waveform block derives its audio data).

Time Offset

Allows you to place the waveform's beginning point at a specific point within the session's timeline.

Mute

Mutes the selected waveform Block.

Lock in Time

This command "locks" any selected waveform (or combination of waveforms) to its current time location in the multitrack session. Locking a waveform is useful when you have decided upon its position in time, and do not want to accidentally move the waveform when right-clicking on it, etc. While a waveform which has been locked cannot be moved in time (left or right), it may be vertically repositioned, allowing you to move it between tracks. To unlock a waveform, simply select it and choose Lock in Time to free it for horizontal movement.

Lock to Play Only

Lock for Play Only will set the selected waveform block so that it may not be recorded into. The Locked waveform will not become red when positioned in a record-enabled track. To re-enable recording for a waveform, uncheck this box, or choose **/Edit/Lock for Play Only**.

Crossfade

Crossfades are used to create smooth transitions from the end of one audio segment to the beginning of another. They do so by creating a fade out and a fade in over the transition region. Waveforms to crossfade between should be positioned in different tracks (or can be in the same track, in a back-to-back manner), adjacent if possible.

To crossfade, make a selection with your mouse and include in it the beginning and ending sections of any waveforms you wish to include in the crossfade. The highlight should extend a bit beyond the end of the waveform(s) to fade from, and should start a bit before the beginning of the waveform(s) to fade to. Or, if **Edit/Snapping/Snap to Waves** is enabled, the highlight will easily adjust itself to the beginnings and endings of the waveforms. (You can also adjust the highlight by holding Shift when doing a left-click near the edge of the selection.) After establishing a highlighted area, you need to select the waveforms to include in the crossfade by using ctrl+left-click. Next, choose Crossfade to fade out the endings of the selected waveforms, and then have the beginnings fade in.

You can choose between Linear and Sinusoidal crossfades. Choose Linear for an even grade, and Sinusoidal for a curved, sine-like slope to the fade.

The fade curves created with Crossfade can be modified by enabling choosing Show Volume Envelopes and Enable Envelope Editing from the View menu.

Try highlighting the crossfade area in one of the waveforms, then using **Ctrl+left-click** to activate other waveforms to be used in the crossfade. You can also adjust the highlight by holding Shift when doing a left-click near the edge of the highlight. If Snap to Waveforms is enabled, the highlight will adjust itself very nicely to the beginnings and endings of the waveforms. When Crossfade is chosen, then the endings of all selected waveforms will fade out, and the beginnings will all fade in linearly. The fade curve can be modified by enabling the Volume envelope and Edit Points feature.

Punch In

When a particularly difficult passage is being recorded, it's not uncommon for a note or words to be missed or totally screwed up (to err is human)... no big deal! You can either stop and pick up from before the mishap, or you can continue on (as though nothing had happened) and record over the mishap at a later time (a process known as a **punch in**.)

Punch in using *Cool Edit Pro* is particularly easy. Simply highlight the area that you would like to record over, select the **/Edit/Punch In** option from the menu (or right click on the waveform block and choose **Punch In**. By placing he desired track into the Record Ready mode, placing the playback cursor at a convenient point before the "punch" is to occur, you're ready to record the proper notes over the previous mishap.

Take History

Whenever one or more **Punch Ins** have been performed within a waveblock, a **Take History** option will be activated (both **Punch Ins** and **Take History** are available by right-clicking on the waveform block.) The **Take History** option lets you revert the waveform block back to any previously-available take level (essentially acting as a selective undo function - allowing you to choose any change level between its original un-punched state and the current waveform level.

Merge Current Take

Choose this to Merge a take created with Punch In back into the surrounding material.

Delete Current Take

Deletes the most recent punch in and reverts to the previous punch state (basically an undo function that lets you easy revert to the previous take.)

Mix down

The Mix down function creates a two-track stereo mix that is the summation of the virtual tracks that are contained within an entire session or selected portions of the current session. Of course, the resulting mixdown track will contain all of the session's mix parameters (including volume and pan changes, loops, muting, etc.)

All Waves

Selecting **All Waves** will mix the entire session to a stereo track pair and enter the results into the Edit View window.

Selected Waves

Choosing Selected Waves will mixdown only those waveform blocks that have been highlighted and will enter the enter the results into the Edit View window.

This function comes in handy for "bouncing down" tracks to a single, stereo track pair. In the recording studio, the tracks on an analog multitrack tape machine (and even on Modular Digital Multitracks) are often bounced from various recorded tracks to a single track or pair of tracks. This process frees creates a single "submix" of the previous tracks, so that they can be freed up for the recording of new material.

There are two possible reasons for mixing down several tracks to a single stereo pair. The **first** would be to create a final stereo mix of your session for saving your creation to disk as a stereo .Wav file. **Secondly**, you can use this feature to free up hard disk access and CPU processing time. As some of you may have found out, not all computers are made to be as fast as lightning and have hard limitless hard disk space. By bouncing down tracks to a single track or set of tracks, you can drastically reduce the amount of hard disk space and time that is taken by the CPU to number crunch large numbers of tracks (especially, if there's lots of real-time mixing going on.) So, we've created this handy-dandy function for you folks that have run up against the limits of your PC... (and few of us haven't at one time or another.)

Once you've created a session, you can choose the **Edit/Mix Down (All Waves)** function to create a stereo mixdown of your entire session. Once created, simply save the resulting mix as a file. That's all there is to it. If you'd like to create a submix of a several vocal overdubs (for example), you can simply highlight the waveform blocks that you'd like to be mixed down from the desired tracks and choose the Selected Waves function. Once the waves have been mixed, the results will appear within the Edit View window. You can then re-enter the mixdown back into your session by selecting the entire waveform and choosing the **Insert in Multitrack** function within the Edit

View menu. Once you've switched back to the Multitrack View window, you can now place the mixed tracks at the proper beginning point and mute (or remove) the previous tracks from the session.

Loop Duplicate

Loop Duplicate will duplicate a specified number of image copies of the waveform block that has been selected in a consecutively repeating fashion. Alternately, you can specify a spacing (like every second) to place the block, so that specifying 9 copies with a spacing of 10 seconds will copy the selected wave block and paste it in 9 times, with each being spaced 10 seconds apart.

Duplicate Waveform

This is the number of images that you would like to created.

Waveform Spacing

This determines the spacing at which to place the duplicated Image(s). The **No Gaps** setting, places each subsequent image directly after the preceding block, for a continuous loop. Choose Evenly Spaced if you would like to define a spacing to use. The entry box to the right will display the spacing in the current Time Display format. It defaults to the length of the selected waveform block, which function identically to **No Gaps**. Entering a greater value will put space between each waveform block, a lesser value will cause them to overlap.

You can add some cool effects to an existing audio file in the Multitrack View window by **Shift-right clicking** on the waveform (possibly containing vocals, a guitar lick, claps... you name it) and creating a duplicate image. The duplicate waveform can then be placed onto a new track (possibly below the existing track)... and then shifted in time (relative to the original file), to create cool phase shifts (small time offset), delays (15-50 ms offsets) or discrete echoes (50 ms or greater offsets)... without having to alter the original audio file waveform. Go ahead... create a number of image waveforms and see just how wild of an effect you can come up with... You might want to reduce the volume levels slightly for each image, as the combined waveforms will increase the track's overall level in the mix.

Using the **Show Volume-** and **Show Pan Envelopes**, you can also change the relative volume and pan levels so that the effect will change over time or stop/start at a particular time.

Record or open up an audio file that has a beat to it. Use the **Edit/Find Beats** option within the Edit Menu to create a loop that doesn't have any breaks or gap points (if you've done it, congratulations, you're now a loopologist!) Now choose the **Insert in Multitrack** function within the Edit View menu. Once you've switched back to the Multitrack View window, you can now select the loop's waveform block and either right-click on the block and choose Loop Duplicate or choose it from the Edit menu. and choose the number of times that you'd like to loop the segment. Now press play and listen to your cool groove.

Convert to Unique Copy

This command will take the selected waveform block and create a new audio file from it. This is useful for when you have a waveform block in a track that has been edited in some way (trimmed, spliced, etc.), and you now wish it to represent its own audio file.

For example, let's say you have a guitar part, guitar.wav, placed in Track 1. You decide to use the end of this part again later, in Track 2. To accomplish this, you create an Image of the waveform using **Shift+Right-Click**, and drag the Image to Track 2. After highlighting the end portion of the Image that you want played, you then use /Edit/Adjust Boundaries to trim it. Now, because the material in Track 2 is an Image, any changes to the end of the waveform in Track 1 will be reflected in the Image, at its location in Track 2. However, selecting the trimmed Image and choosing Make Unique will create a new audio file with the material displayed in the Image as its contents, thus avoiding its being affected when editing guitar.wav.

A way to add effects to an existing audio file in the Multitrack View window is by creating an actual copy of the waveform (this can be done using the **Convert to Unique Copy** command in the Multitrack View mode or by saving the file under a new name using the **Save As** command within the Edit View Mode). Once a copy of the original file has been made, it can be processed in any way you'd like and then be placed onto another track, such that it will simultaneously play back with the original file. Using this method, you can then vary the effected track's volume and pan controls (in a non-destructive, automated fashion) without altering the original audio file in any way.

Although the Transform function doesn't exist within the Multitrack View, you can simply double-click on a waveform block (which will open up the selected waveform into the Edit View window)... From here, you can use the **/Edit/Save Selection** option to create a copy of the file that can be effected using any Edit View Transform function... Once done, the newly processed file can be placed into the Multitrack View window.

Mute

Mutes and Un-mutes the selected Waveform, effectively allowing a specific waveform within a session to be ignored during playback.

Lock in Time

This command "locks" any selected waveform (or combination of waveforms) to its current time location in the multitrack session. Locking a waveform is useful when you have decided upon its position in time, and do not want to accidentally move the waveform when right-clicking on it, etc. While a waveform which has been locked cannot be moved in time (left or right), it may be vertically repositioned, allowing you to move it between tracks. To unlock a waveform, simply select it and choose Lock in Time to free it for horizontal movement.

Locked waveforms are identified by a **TAN** colored border at the top and bottom of the waveform.

Lock for Play Only

This command marks the selected waveforms as play only. Lock for Play Only disables the ability of a waveform block to be recorded into, so while the track they reside in may be record enabled (have the red button on the Track Console depressed), the marked waveform will not be recorded over. This is signified by the waveform not turning red in color when the track is record enabled.

Allow Multiple Takes

With this option enabled a Take History is preserved for the selected waveform in the event that it is recorded over. With Allow Multiple Takes, material is not overwritten when recording into an existing waveform block, instead a new instance, or take, of the block is created and recorded into, preserving the previous waveform. With this disabled, recording is done into the same spot each time. The *Punch In* function will automatically enable Allow Multiple Takes for the selected Punch area.

Splice

The **Splice** option creates a non-destructive "break" within a waveform block at the current cursor point. Once the block has been "spliced", each portion of the newly created blocks can be moved, deleted, slid in time/track, etc. with complete freedom.

To save time, simply right-click on a waveform block and choose **Splice**.

Merge/Rejoin Splice

Merge/Rejoin Splice will recombine a spliced segment of a waveform block with the original surrounding material, provided that the spliced segment sits at the same relative location as where it was spliced from, and in the same track as the original segment(s). The boundaries of the segments are joined together using a 30 millisecond crossfade. This command can also be used to rejoin a take created using Punch In to the rest of the waveform.

This merge operation is destructive in that it does alter your sound file (when rejoining a Punch In segment, the file is rewritten with the new material, for example).

Select All Waves (Ctrl-A)

This command selects all waveforms within the current session. Selected waveforms will appear brighter in color than their non-selected counterparts; record enabled waveforms (**red**) will become bright red when selected, regular unmuted waveforms (**green**) will become bright green whenever they are selected.

Trim / Expand

This function will adjust the boundaries of the selected waveform block either inwardly or outwardly. After highlighting to define an area to adjust to, any material in the waveform block that lies within this highlight will be retained or exposed, removing that which is outside of the highlight (or including material inside the selection that is not currently revealed). For example, with a 5 second waveform block, highlighting the middle second and choosing Trim/Expand will trim off the surrounding material, leaving just the middle second. If you decide you should not have trimmed off so much, you can highlight the middle second plus a little to the right, for instance, and again choose Trim/Expand to bring back that little bit to the right, thereby lengthening the waveform block.

Unlike the Trim command in the Edit Waveform View, using Trim/Expand to trim does not remove any data from the actual waveform. Trimming in Multitrack View simply tells *Cool Edit Pro* to “disregard this material” when playing back. You may return any trimmed waveform to its original state at any time by selecting it and choosing /Edit/Full.

To fine tune your adjustments in this example, you could also have double-clicked on the trimmed block to bring up the Edit Waveform View, double-click to highlight the displayed area, then zoom out to adjust the highlight boundaries to your satisfaction. When finished, switch back to multitrack view (the highlight will be retained) and choose Trim/Expand to have the waveform block re-adjust to match your highlight.

Cut

Cut will remove any highlighted portion of a selected waveform. Once a waveform range has been selected, the **Cut** function will remove only the selected range and will leave those areas that haven't been selected intact. This is also useful for eliminating extraneous noises, breaths, talking... basically anything that you would like to have removed. For example, if track 3 contains a waveform that is 5 seconds in length, selecting it, highlighting the middle 1 second, and choosing Cut will remove the middle 1 second, leaving the 2 second portions on both sides. Use Trim to remove unwanted material around a highlighted region.

Unlike **Cut** in **Edit View**, this Cut command does not remove any data from the actual waveform, and places nothing on the clipboard. Using Cut in Multitrack View simply tells *Cool Edit Pro* to "disregard this material" when playing back. You may return any Cut waveform to its original state at any time by selecting it and choosing **/Edit/Full**.

Full

This option returns any selected waveform to its original state (the full waveform). Any material that has been removed using Trim or Cut (while in Multitrack View) will be replaced.

Snapping

Enable Snapping to have the edges of any waveform "snap" to certain time locations when being dragged. This allows you to easily line up any number of waveforms at a specific start time, without having to zoom in and manually find the location. The Snap To locations (which vary depending on the Snapping option you have selected) act as magnets for any waveforms dragged in their proximity, and have the effect of pulling the waveform toward them. For example, if your vertical ruler's format is hours:minutes:seconds, with seconds showing as the division at your current zoom level (1.0, 2.0, etc.), then dragging a waveform which starts at 1.6 seconds to the right will result in it snapping to the 2.0 location as it nears it. You can, of course, keep dragging beyond 2.0 in this case, if you wish. You can choose among three options in determining the time locations to Snap To.

Open a single waveform block into the waveform window and perform a "Splice" (which effectively splits it into two blocks). Turn the Snapping option off. By right-clicking on the second block and moving it, try to manually rejoin the blocks. You'll notice that it may not be so easy to re-join them, so that playback is continuous. Now, turn the **Snap to Waves** option on and try to rejoin them... you'll find that the second block "magnetically" snaps to the first block and will play back continuously, with little effort on your part.

Refresh Now

Forces *Cool Edit Pro* to start the background mixing process "from scratch".

Remove Waves

Removes any selected waveforms from a multitrack Session. Removed waveforms will still be available in Edit View, and to the Session through the Insert menu until the audio files are Closed.

Waveforms can be selected individually or they can be selected into groups by holding down the Control (Ctrl) key and then left-clicking on the waveforms that you would like to select. De-selection is done by keeping the Control key held down and then left-clicking on the waveform to be de-selected again.

View Menu (Multitrack View)

The View Menu displays options that relate to changing the display or viewable area when in the Multitrack View Mode.)

Edit View/Multitrack View

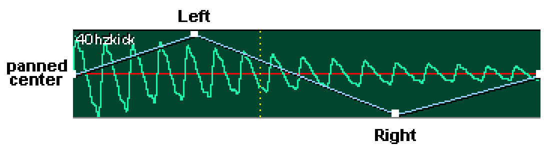
Cool Edit Pro offers two main edit modes: **Edit View** and **Multitrack View**. To choose between these options, select the one you want from the View menu or click-on the icon to toggle between the two operating states. A checkmark will appear next to the view mode that's currently selected.

The Edit mode icon that is currently displayed on the screen represents the alternate operating mode that the system will "jump" to when pressed. For example, when working in the Edit View mode, the Multitrack View icon "" will appear. Conversely, when working in the Multitrack View mode, the Edit View icon "" will appear.

Show Pan Envelopes

This option lets you draw panning envelopes that range from Left (top portion of the waveform block's display) to the right (bottom portion of the waveform display). When activated by the **Enable Envelope Editing** option, pans can be drawn into each waveform block by simply clicking anywhere on the light blue pan line and moving the line to the desired L/R pan position. All pan calculations are totally non-destructive (meaning that the original audio file data isn't effected by these changes) as they are processed in real time as the session track is being played back.

It's important to remember that real-time pan calculations will require some number-crunching on your computer's part. Mixing a large number of tracks in real-time may "bog down" your computers main processor, causing "jumps" or "digital glitches" during playback. If this happens, you may want to use the **/Edit/Mixdown** function to create a submix that effectively "bounces" several tracks down to a single track or stereo pair of tracks.



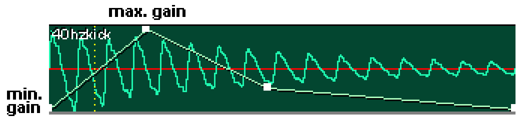
Pan Envelope

Place a waveform block within the waveform window and select the **Enable Envelope Editing** option and then activate the **Enable Pan Envelopes** option. Click anywhere on the light blue pan line several times to create some wild L/R pans. Then sit back and be amazed!

Show Volume Envelopes

This option lets you draw volume envelopes that range from minimum gain (bottom portion of the audio file's waveform block) to maximum gain (top portion of the waveform block). When activated by the **Enable Envelope Editing** option, volume changes can be drawn into each waveform window by simply clicking anywhere on the light green volume line and moving the line to the desired gain position. All volume changes are totally non-destructive (meaning that the original audio file data isn't effected by these calculations) as they are processed in real time as the session track is being played back.

It's important to remember that real-time volume changes will require some number-crunching on your computer's part. Mixing a large number of tracks in real-time may "bog down" your computers main processor, causing "jumps" or "digital glitches" during playback. If this happens, you may want to use the **/Edit/Mixdown** function to create a submix that effectively "bounces" several tracks down to a single track or stereo pair of tracks.



Volume Envelope

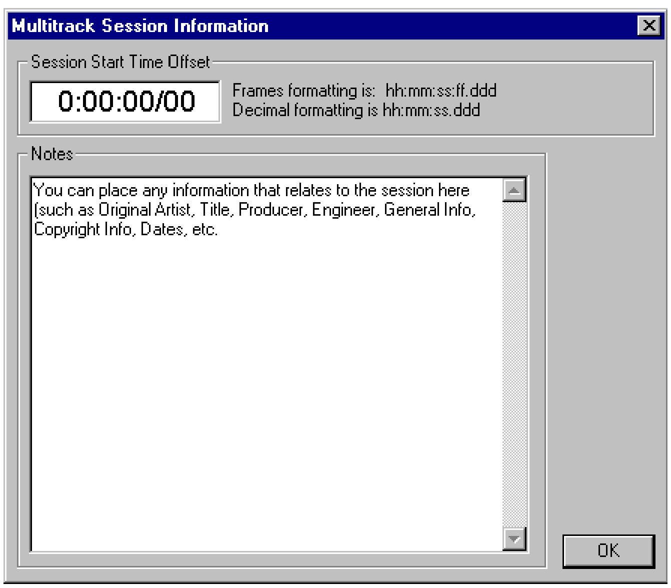
Place a waveform block within the waveform window and select the **Enable Envelope Editing** option and then activate the **Enable Volume Envelopes** option. Click anywhere on the light green volume line several times to create some wild volume moves. Then sit back and be amazed!

Enable Envelope Editing

This option simply enables or disables the drawing of both Volume and Pan Envelopes within the waveform block's of a session, when in the Multitrack Edit View.

Info (Alt+I)

Extra user-information can be directly imbedded within your session's *.ses file. This information should stay with your session file through its lifetime.



multitrack Session Information Dialog

Session Start time Offset

This denotes the time in frames (frames formatting is hh:mm:ss:ff.ddd and decimal formatting is hh:mm:ss.ddd) that the SMPTE start time will be offset by. This offset number is often useful when the session start time is to be triggered at a number other than 00:00:00:00. For example, if you are wanting to sync vocal tracks to a song that begins at 00:05:00:18 on an ADAT digital multitrack recorder, you'll want to set your SMPTE offset time to this number.

Notes

You can place any information that relates to the session here (such as Original Artist, Title, Producer, Engineer, General Info, Copyright Info, Dates, etc.

Time Window

This windows is used to provide an easy-to-read time display that can readout the current cursor position in decimal, SMPTE drop, samples, bars and beats and custom SMPTE frame rates.

Time Window

The standard method of interlocking audio, video, and film transports makes use of synchronization code known as SMPTE time code (SMPTE stands for the Society of Motion Picture and Television Engineers.) The use of time code makes it possible for sequential, time-based location points (known as addresses) to be encoded either onto a track of a video or audio tape recorder, or imbedded within the digital code of a MIDI sequencer, DAT or digital multitrack recorder. This address code cannot slip and always retains its original location, which allows for continuous monitoring of tape position to an accuracy of between 1/24th and 1/30th of a second. Each time code "frame" is tagged with a unique identifying number, known as a time code address. This eight-digit address is displayed in the form 00:00:00:00, in which the successive pairs of digits representing Hours:Minutes:Seconds:Frames. This window is used to display a quick-n-easy time code readout that can be resized to your liking.

Insert List (Multitrack View)

When in the Multitrack View mode, the Windows menu gives you a simple way to navigate between audio file waveforms and to place waveforms into *Cool Edit Pro*'s Multitrack View window.

Waveforms List

The first thing that you want to do when starting a session, is to begin loading waveforms into the Waveform Display.

When working in the **Multitrack View** window, the **Waveform List** works as a simple navigation tool for loading audio files into a multitrack session. Basically, it can be thought of as a central "holding tank" from which audio files can be easily selected and placed into the **Multitrack View** window. You can also use Waveforms List to convert audio files into the appropriate format for the current project.

Waveform List options include:

Waveform List

This displays the audio files or waveforms that are currently open in *Cool Edit Pro*. Click on any item in the list to select it, or double-click to switch to that waveform in the Edit View window. To select more than one item in the list, hold down the **SHIFT** key for contiguous selection, or the **CTRL** key for non-contiguous selection.

You can quickly insert waveforms at specific locations through drag-and-drop. Left-click on a waveform in the list and drag it (keep the mouse button held) into the Multitrack View window at the desired track and time placement, and then simply release the mouse button. It's that easy!

Switch to

Places the selected waveform into the Edit View window.

Close Wave

Closes the selected waveform and removes it from the Waveforms List. If the file being closed is currently being used in the Multitrack environment, all references to this file (Images, etc.) will be removed from the current mix. A dialog box will appear as a fail-safe, asking you if this is what you really want to do.

Insert

Inserts the selected waveform(s) into the Multitrack View window, so that it can be mixed within the multitrack environment. *Cool Edit Pro* places the selected items into the next available track(s). If the selected audio file is not of the same sample type as those in the current session, the **Convert Sample Type** dialog box will pop up to allow you to convert the file to the appropriate sample rate/file type.

Open

Lets you browse for and open a recorded audio file.

Full Paths

Displays the full DOS path for files in the Waveform List.

Close

Closes the Waveforms List dialog box.

List of loaded waveforms

This list displays all currently open waveforms. Simply select the waveform name to insert it into the Multitrack View window, so it can be mixed within the multitrack environment. Inserted waveforms will be successively placed into the next available empty track, beginning at the left-most point of the window's visible timeline. Files names are given as full paths.

Synchronization Using SMPTE Time Code

By David Miles Huber

Assigning a SMPTE Source to Cool Edit Pro

If your system is equipped with MIDI devices, you can also choose the MIDI in source that will be used to deliver MIDI Time Code to *Cool Edit Pro*. This is done using the **MIDI In (Sync/Trigger)** dialog box within the **Devices** setup tab of the **/Options/Settings menu**.

*What is SMPTE?**

The standard method of interlocking audio, video, and film transports makes use of synchronization code known as **SMPTE time code** (SMPTE stands for the **Society of Motion Picture and Television Engineers**.) The use of time code makes it possible for sequential, time-based location points (known as addresses) to be encoded either onto a track of a video or audio tape recorder, or imbedded within the digital code of a MIDI sequencer, DAT or digital multitrack recorder. This address code cannot slip and always retains its original location, which allows for continuous monitoring of tape position to an accuracy of between 1/24th and 1/30th of a second. Each time code "frame" is tagged with a unique identifying number, known as a time code address. This eight-digit address is displayed in the form 00:00:00:00, in which the successive pairs of digits representing Hours:Minutes:Seconds:Frames. This window (which can be accessed using the **/View/Time Window** command) is used to display a quick-n-easy time code readout that can be resized to your liking.

*What is MIDI Time Code?**

For decades, SMPTE time code has been the standard timing reference within audio and video production. This is due to the fact that it is an absolute timing reference that remains constant throughout an entire program.

In order for MIDI-based devices to operate on an absolute timing reference that is independent of tempo, Chris Meyer and Evan Brooks of Digidesign created **MIDI time code** or **MTC**. Basically, MIDI time code provides a cost-effective and easily implemented means for translating SMPTE time code into MIDI messages. It also allows for time-based code and commands to be distributed throughout the MIDI chain to those devices or instruments that are capable of understanding and executing MTC commands.

MTC makes use of a reasonably small percentage of the available MIDI bandwidth (about 7.68% at 30-fr/second). Although it's able to travel the same signal path as conventional MIDI data, it is recommended, wherever possible (within a 32- or more MIDI channel system), that the MTC signal path be kept separate from the MIDI performance path in order to reduce the possibility of data bottlenecking or delay.

* Excerpted with permission from The MIDI Manual - David Miles Huber, Focal Press (Boston/London), www.bh.com/focalpress

Understanding MIDI

By David Miles Huber

Assigning a MIDI Source/destination to Cool Edit Pro

If you have multiple sound cards, or a single card that has multiple outputs, you can use the Devices tab to choose the input and output devices you would like *Cool Edit Pro* to be assigned to. This is done using the **MIDI In (Sync/Trigger)** and **MIDI Out (Music Preview)** dialog boxes within the **Devices** setup tab of the **/Options/Settings menu**. If your system is equipped with MIDI devices, you can also choose the MIDI in, and MIDI out sources, using this setup option.

These settings will be remembered in the [*Cool Edit*] section of Window's WIN.INI file, which means if you install a new sound driver or card, Cool will not access it until you choose it from this dialog.

MIDI In (Sync/Trigger)

This setting is used to assign a MIDI source to *Cool Edit Pro*'s MIDI In Sync/Trigger input. The Sync source will provide a MIDI Time Code source that *Cool Edit* can synchronize to, while a trigger source can be used to transmit MIDI not on messages for triggering individually assigned waveforms.

MIDI Out (Music Preview)

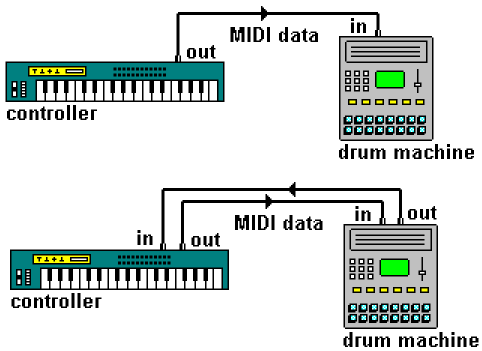
This setting is used to assign *Cool Edit Pro*'s Music Preview function (see **/Transform/Special/Music**) to a MIDI destination (such as a sequencer or MIDI instrument).

*What is MIDI?**

MIDI (short for **Musical Instrument Digital Interface** is a digital communications "protocol". That is to say, it is a standardized control language and hardware specification that allows multiple electronic musical instruments and devices (that can transmit and/or respond to this control language) to communicate real-time and non real-time performance and control data.

MIDI is a specified data format that must be strictly adhered to by those who design and manufacture MIDI-equipped instruments and devices. In this way, performances and task-related automation functions can be communicated between devices with relative transparency, speed and ease. Thus, when performing a task (such as controlling multiple instruments from a keyboard controller or transmitting a patch bank from a patch librarian to a synthesizer), the user need only consider the control parameters of the involved devices and not those of the transmission medium itself (in this case, MIDI). This could be likened to our English language, through which ideas can be transmitted from one person to an audience. We, as English speaking people, are able to concentrate wholly upon the content of a lecture without having to think about the language medium itself. Similarly, performance and control data can be easily communicated through the standard medium of MIDI.

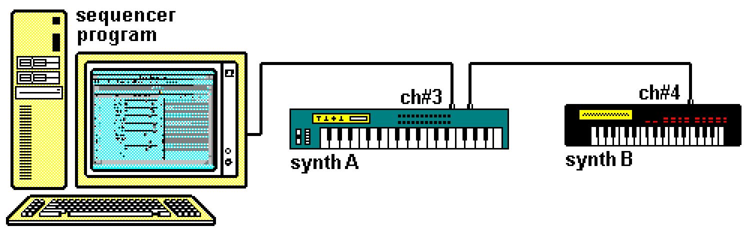
One of the major distinction between MIDI and speech is the fact that MIDI (as defined in the 1.0 specification) is not bi-directional. Data within a single MIDI line can only travel in one direction, from a single source to a destination. In order to make two-way communication possible, a second MIDI data line must be connected from the external source back to the original destination device.



Example MIDI Configurations

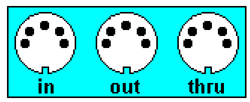
Just as it is possible for a public speaker to single out and communicate a message to one individual within a crowd, a single MIDI line is able to communicate MIDI messages over 16 channels to a specific device or range of devices that are instructed to respond to these messages.

Whenever a MIDI device is instructed to respond to any of the 16 channel numbers, it will ignore Channel messages that are transmitted upon any other channel. Likewise, any device that is selected to respond to a specific MIDI channel will only respond to messages that are transmitted upon that channel (within the capabilities of the device). As an example, let's assume that we have two synthesizers and a MIDI sequencer (a device that is capable of recording, editing and outputting MIDI data) with which to create a short song. We might start off by playing a melody line on synthesizer A into our sequencer, which is set to transmit and respond to data upon MIDI channel #3. Having done this, we can then decide to play background chords upon synthesizer B, which we shall set to MIDI channel #4. Even though the system is connected by one MIDI line, it would be a simple matter for our sequencer to output the previously recorded MIDI data on channel #3 (which will still be played by synth A), while our synth B simultaneously responds to our live playing on the keyboard. Upon playing back the finished MIDI sequence, both of our instruments will respond only to their assigned MIDI channels and will reproduce their individual sounds as they were originally recorded.



Using MIDI with a Sequencer

Most MIDI devices have three types of ports that make use of 5-pin DIN jacks in order to provide interconnections between MIDI devices within a network; these are: the MIDI in, MIDI out and MIDI thru ports.



MIDI Ports

MIDI IN

The MIDI in port receives MIDI messages from an external source and communicates this performance, control and timing data to the device's internal microprocessor.

More than one MIDI in port can be designed into a system that is capable of providing MIDI merging functions or for devices that can support more than 16 channels. Other devices (such as a MIDI controller) may not require the use of a MIDI in port at all.

MIDI OUT

The MIDI out port is used to transmit MIDI messages from a single source device to the microprocessor of another MIDI instrument or device.

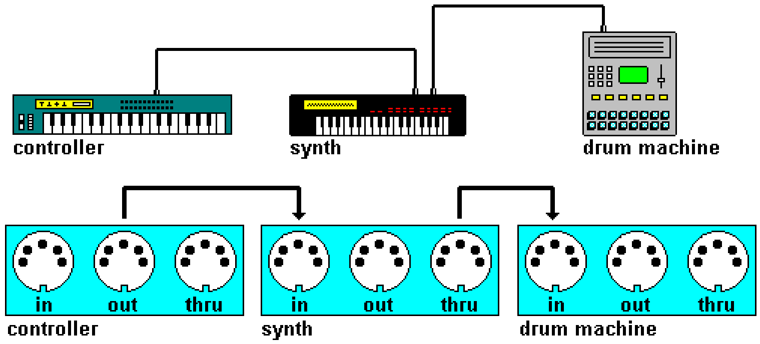
More than one MIDI out port can be designed into a system for the simple purpose of providing multiple MIDI outs (providing distribution of the same data stream to a number of devices). Alternatively, devices that can support more than 16 channels often have the ability to rout individual MIDI data channel information to more than one isolated MIDI port. This has the advantage of providing a system with greater channel capabilities, providing data isolation between MIDI ports (thus reducing possible data clogging) and allowing the user to filter MIDI data on one port while not selectively restricting the data flow within another port.

MIDI THRU

The MIDI thru port provides an exact copy of the incoming data at the MIDI in port and transmits this data out to another MIDI instrument or device that follows within the MIDI data chain. This port is used to relay an exact copy of the MIDI in data stream to the thru port and is not merged with data that is transmitted at the MIDI out port.

THE DAISY CHAIN

One of the simplest and most commonly used methods for distributing data within a system is the MIDI daisy chain. This method is used to distribute a single MIDI data line to every device within a system, by transmitting data to the first device and subsequently passing an exact copy of this data through to each device within the chain. This is done by sending the MIDI out data from the source device (controller, sequencer, etc.) to the MIDI in of the second device. By connecting the MIDI thru port of the second device to the MIDI in of a third device, this last device will receive an exact copy of the original source data at its input. This process may then continue throughout a basic MIDI system until the final device is reached.



A MIDI Daisy Chain

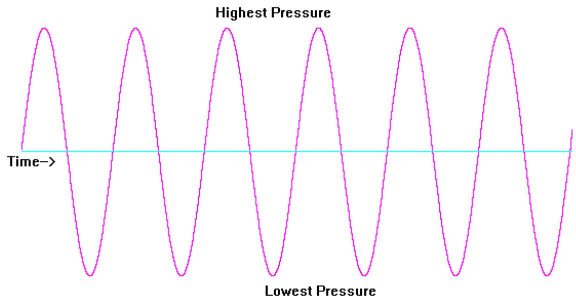
As the MIDI thru port will only pass data that is an exact copy of the data at its MIDI in port, the signal can be traced back through each device to a single master controlling device (such as keyboard). In most cases, this is acceptable as the controller is used to transmit data over one or more MIDI channels that will, in turn, be individually responded to by devices that have been assigned to these channels. Any device within a chain may be used as a controlling source by simply plugging its MIDI out port into the MIDI in of any device that follows within the MIDI chain.

* Excerpted with permission from The MIDI Manual - David Miles Huber, Focal Press (Boston/London), www.bh.com/focalpress

A Short Course in Digital Signal Processing

By David Johnston

Signals (or Waves)



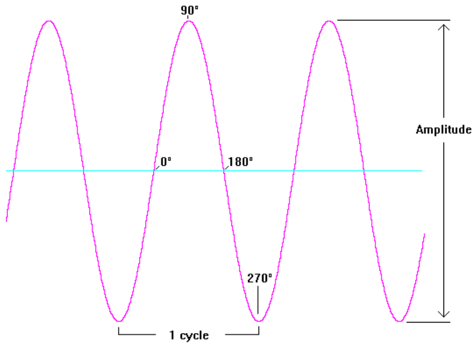
Waves

Waves in the context of *Cool Edit* are sound waves. A sound wave can be written as how the air pressure on your ear changes over time. When you hear a loud sound, the pressure on your eardrum is greater, and it vibrates harder. Soft sounds affect the eardrum very little, and thus are noticed as softer sounds. So a wave is a convenient representation of how the sound level varies over a time interval. The illustration is a sine wave of a constant pitch. It shows the sound pressure oscillating from low pressure to high pressure and back. In natural systems, this motion follows the path of a sine wave when graphed. The wave here is of a constant frequency and constant amplitude. Choose any time along the Time axis going from left to right, draw a vertical line up and down, and there will be exactly one spot where the wave crosses this vertical line. This is because a wave can have only one value at any instant in time. For example, there can not simultaneously be two different pressure levels on the eardrum at once. If two sounds are heard at the same time, the pressure levels from both of them are simply added, and a single resultant pressure is observed (and at that instant, it is impossible to tell exactly which role each sound played in creating this single value). So a waveform is depicted as a line that can vary up and down freely, going from left to right, but with no "backtracking" (e.g. a graph of a waveform will never look like a circle, or a "U" on its side).

Waves in the natural world are continuous, which means that no matter how much you "zoom in" to the waveform, or no matter how small of a time interval you look at, there are an infinite number of values needed to represent the progression of the waveform during that interval. Other types of waves exist besides just sound waves. Seismic activity can also be viewed as a wave - as in the shock wave during an earthquake. The Richter scale graphs (the familiar graphs of earthquake activity they show on television after a quake) are a prime example. There is a single needle that sways back and forth leaving a mark on a slowly turning cylinder. As the seismic receptors placed into the Earth pick up vibrations, the electrical impulses are sent to the device, causing the needle to sway in response to the movement of the earth.

Cool Edit's normal Waveform View displays waveforms just as described, as a plot with the time going from left to right, and at each instant in time there is exactly one value for the waveform's instantaneous amplitude, or pressure level.

Amplitude



Wave Amplitude

The amplitude of a sine wave is the difference between the highest part of the wave and the lowest part. The difference between the high and low pressure parts. A low amplitude, quiet wave would be one that would vary much less up and down, while a louder waveform would vary much up and down. Amplitude is generally measured in decibels, although the decibel (dB) itself not an absolute measurement like Fahrenheit is for temperature, but instead is a measurement of ratio. If one decibel is the quietest sound someone can hear, then the loudest sound one can hear without damaging the ears with prolonged exposure would be about 100 dB. Normal speaking would be at about 20 dB. Ten decibels is an increase in volume of 10 times. The decibel scale is not linear, but logarithmic, which means that 20 dB is not 20 times louder, but instead 100 times louder (10 times louder than a 10 dB increase).

Frequency

The frequency of a wave determines the pitch we perceive, and is measured in cycles per second, or Hertz (Hz). As seen on the graph, the time it takes the wave to complete one cycle is the time it takes to go all the way from the point of lowest pressure, on up to highest pressure, and then back to lowest pressure where it started. A cycle can start anywhere, not just at the bottom of the wave. The cycle will always end at the same pressure level it began. For a male voice, you may count about 180 complete cycles in one second of audio, which would give the speaker's voice a pitch of 180Hz. A female singing voice may attain 600Hz. The key "A" below middle "C" on the piano is about 440Hz. Each time a frequency doubles, it is said to raise an octave. So, if 440Hz is "A" below middle "C", then 220Hz is "A" the next octave lower, and 110Hz is still "A", yet another octave lower. The high pitched ringing you may hear emanating from television sets is around 17,000 Hz. The human ear can perceive frequencies up to about 20,000Hz.

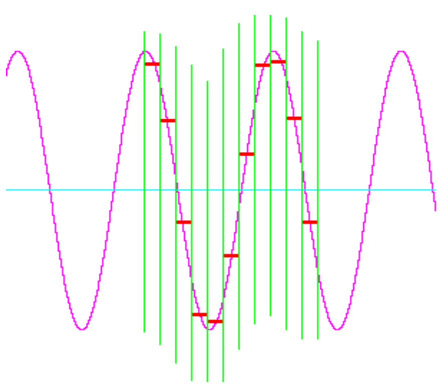
Phase

As a wave cycles through at its particular frequency, it can be thought of as passing through all the degrees of a circle, from zero to 360 degrees. Each part of the cycle can be referred to by its phase, with zero degrees being the midline value as the pressure level is increasing, or as the wave is on its up-swing.

Then 90 degrees would be the peak, and 270 degrees the valley. At 180 degrees, the wave is back at the midline, but this time on the down-swing. At any instant in time, a sine wave can have only one phase value. Phase is the one component that human ears can not discern very easily. For example, a bell rang at exactly 12:00pm, we would not be able to notice at exactly one second past 12:00pm whether the phase of the sound we are hearing is at 0 degrees, or 90 degrees, or anywhere. If the bell rang at 3 milliseconds past 12:00pm, the phase at exactly one second past 12:00pm would certainly be different (depending on the pitch of the bell), but we still would not notice it was different.

The sounds we hear in everyday life and when we listen to music, are generally not just a pure sine waves as exemplified above, but a collection of an infinite number of sine waves, each at its own varying amplitude, and its own varying phase. At any instant in time, what we truly hear is the sum of all of the frequencies present, each at their own amplitude, and each at a particular phase during their cycle. Summing all these will give exactly one value, or one pressure level that is present at the eardrum at any particular instant. Because of this, a wave can be graphed as the instantaneous amplitude (or pressure level) during an interval of time. *Cool Edit*'s Spectral View will display all the frequencies present at any instant in time (almost), and their amplitudes at that instant. The louder a particular frequency is, the brighter the point will be. The higher up on the display the point is, the higher the frequency is that the point represents. Time is still represented as going from left to right. The reason we say the Spectral View *almost* displays all frequencies present at any instant in time is because it is impossible to determine all frequencies at any instant that went into producing the pressure level that was measured during recording, but it can be estimated from the audio before and after that point. This is why, as you zoom in more and more, the particular frequencies begin looking very smeared from left to right.

Sampling



Wave Sampling

In the computer, it is impossible to work with an infinite amount of data, which is what would be required if a continuous wave were to be represented inside the machine, so at every possible instant in time we would have the value of the waveform at that instant. For this reason, it is necessary to *sample* the data. Sampling consists of breaking up the waveform at constant intervals, and representing all values in that interval by a single value. By dividing the waveform up like this, one second of audio can now be represented by a finite number of values. The *sample rate* is the number of divisions taken per one second of audio. As you can see by the graph, the sampled waveform (horizontal dashes) contains much less information about the shape of the wave than the original continuous wave itself. The highest frequency that can be represented by this method is exactly one half the sample rate. So, if a sample rate of 22,050 slices per second was used, the highest frequency that can be represented would be a tone of 11,025 Hz. This frequency is known as the Nyquist frequency. The higher the sample rate, the higher the frequencies are that can be represented by sampling. The human ear can hear frequencies up to about 20,000 Hz (or 20 kHz), so to represent all sounds that humans can hear, a sample rate of at least 40,000 Hz must be used, which would yield 40,000 values for the computer for each second of audio. This is why CD players sample at 44,100 Hz, to be enough above twice the highest frequency anyone can hear.

So far we have broken *time* into discrete intervals. What about the actual values, or amplitude, at each of those intervals? The wave above can be thought of as going from values between 100 (at the top) to -100 (at the bottom). This would cause the short sequence of 11 samples to be given to the computer as roughly 93.7, 51.5, -22.1, -89.4, -97.6, -48.0, 25.7, 92.1, 93.9, 54.5, and -21.1. The same problem as with having continuous time exists with the values at each interval as well. There are an infinite number of possible values the wave can have during each of these intervals. For example, instead of the first value being 93.7, it could just as easily have been 93.716746352231... These values must also be broken up into intervals (which may or may not be evenly spaced) so that computers can deal with them. This breaking up of continuous values to a specific number of values is known as *quantizing.* An 8-bit sound card, for example, can have any one of 256 unique values, where the continuous range of real numbers has been quantized to 256 steps. The reason 8-bit sound has 256 possible values is because 2 to the 8th power (2 x 2 x 2... eight times) is 256. The 16-bit sound cards can have any one of 65,536 values for each interval. The quantization level, directly relates to the *dynamic range* (difference between the softest and loudest sound) that can be represented (if the range has been evenly divided into equal spaced steps). With only 256 levels, the sound quality is comparable to that of AM radio. With the 65,536 levels possible with 16-bit audio, the quality is comparable to that of compact discs, which can have much louder 'louds', and much softer 'softs'. So, where the sample rate (divisions in time) determines the highest frequency sound that can be represented, the quantization level (divisions in amplitude) determines the highest dynamic range that can be represented. If the divisions in amplitude are not evenly spaced, but there are more divisions with lower values, and fewer with higher values, softer sounds can be represented with fewer volume levels, at the expense of less precision at higher volume levels. The A-law and m-law file formats do just this -- they save only 256 unique volume levels, but have more levels at the lower volumes. The result is the dynamic range equivalent to having 4096 volume levels (comparable to a high quality cassette), but with some distortion (like distortion heard on broadcast FM radio, but a little worse).

Sound Reproduction

The hardware in your sound card has a special chip for converting the *analog* (continuous time) signal to *digital* (discrete or sampled) signal called an A/D converter. It also has the reciprocal chip, which converts the digital signal from the computer to an analog signal that can be piped to a pair of headphones, or to your stereo system which is called a D/A converter. Even though the sampled signal has far fewer samples than a continuous time analog signal, the D/A convert can reproduce the original analog signal perfectly. This means that the only loss in going from the analog domain to digital, then back to analog again is the loss in higher frequencies (those above 1/2 the sampling rate), and loss in the dynamic range (depending on the quantization level used). Of course, with lower quality components, there will be loss in the A/D converter by the converter not perfectly converting the original analog signal to digital. Also, there can be loss in the pre-amplifier of the sound card after the signal has been converted from digital back to analog. What this means, though, is that it is perfectly safe to take any audio signal, convert it to digital, work with it in the digital domain (on the computer using *Cool Edit*), and convert it back to analog (play the sound). This means we are no longer restricted to the limitations of electronic components in filters, special effects boxes, etc.

Quality Issues

As alluded to before, there are many sacrifices, gives, and takes that must be considered when going from the analog domain to the digital. With enough money, you can purchase a 'high quality' sound card that has excellent A/D and D/A converters, and high quality pre-amplifiers for getting the signals from the real world into the computer, and back out again. But no amount of money in the world will make a 16-bit sample have a higher dynamic range than 96 dB, or a wave sampled at 22,050Hz able to contain frequencies above 11,025Hz. The current sound cards, though, are about matched in quality between their analog components and digital specifications. That is, the components can handle frequencies up to 24 kHz, and have a signal to noise (S/N) ratios approaching 96dB, which matches the digital specifications of a 48 kHz sample rate, and 16-bit word size. Some high quality amplifiers on the market can amplify signals upwards of 30 kHz, and also have S/N ratios of around 110 dB, perhaps more. Even though the digital portion may be trivial (sampling at 60 kHz, and using a 20-bit word size), a sound card of this caliber may be larger than your PC! If the components in the sound card can not handle these higher specifications, then the extra space required by the PC to store larger samples (20 bits instead of 16 bits) and more samples per second (60,000 as opposed to 48,000) would just be wasted.

Noise

In general, noise is the opposite of pure tones or sine waves. Instead of sound at exactly one frequency, noise consists of random collections of all frequencies. At any instant in time, any number of frequencies may be present, at any volume, at any phase. The effect is that of the static you hear when tuning between FM radio stations, or the sound of your TV when it is not tuned to any broadcast. See the Generate Noise function for examples of types of noise, and try generating some yourself. Notice that in the Spectral View, the entire screen is filled with colors -- with random splotches all over. Noise can be colored by filtering it, which reduces the occurrence of some frequencies, or increases the occurrence of others.

Filtering

Filtering is one of the most popular uses for digital signal processors, or programs that modify sound data. Simply put, a filter just adjusts the volumes of specific frequencies, or adjusts the phases of specific frequencies. Many of the functions in *Cool Edit* are based on the Filter. The Filter function itself allows one to choose exactly which frequencies should be boosted, or cut. The Noise Reduction function dynamically cuts frequencies by differing amounts depending on how much of that particular frequency is present. It analyzes the frequencies present in the unwanted portion, and attempts to cut out these, and *only* these frequencies while leaving all the rest untouched. The Quick Filter function boosts or cuts a large range of frequencies, that range from generally low frequencies to generally high frequencies. Any of these filtering functions can be tested by generating white noise (which contains equal amounts of all frequencies), filtering it, and then viewing it in the Spectral View. Just filter half the waveform and compare the difference between the unfiltered data and filtered data to see exactly which frequencies have been boosted, or cut.

The FFT

The Fast Fourier Transform is an algorithm that *Cool Edit Pro* uses to perform its filtering functions, as well as its Spectral View and Frequency Analysis functions. This transform takes data in the time vs. amplitude format (the Waveform View plot), and converts it to time vs. frequency (the Spectral View plot). It can also convert back the other direction. For more detailed information on the FFT, see any books on Digital Signal Processing.

Miscellaneous Notes on Using *Cool Edit Pro*

This chapter contains various notes and explanations regarding the way *Cool Edit Pro* works and how you can get the most out of it.

Cool Edit's Working Philosophy

To get the most out of *Cool Edit Pro*, you should be familiar with some of its general operating concepts. The information below will help you to understand *Cool Edit Pro's* working model in a few key areas.

Use of files

Being a software-based audio system, *Cool Edit Pro* deals with audio in a digital form, meaning that an analog waveform (such as your voice saying "hello") is converted, or sampled, into a binary representation. This sampled waveform then exists as a wave file on your computer's hard drive. When you put together a song or audio presentation in a *Cool Edit Pro*, audio files are your building blocks, much as text files are your building blocks when you use a word processor to compose a book.

One key point is the way in which an audio program operates on a wave file. There are two kinds of computer-based audio editing: destructive and non-destructive editing. Destructive editing means simply that edits (cut, paste, reverb, etc.) are applied to the wave file itself, so that the original audio data itself changes, while non-destructive editing implies that the file on disk is not actually altered; instead, each "edit" is stored as an instruction to be applied to the file. For example, in destructive editing, a change in audio volume would actually alter the amplitude of a wave file, whereas in non-destructive editing, the volume change would consist of instructions that essentially instruct the program to "get louder at this point".

Cool Edit Pro employs both destructive and non-destructive editing methods. However, in *Cool Edit Pro*, even "destructive" edits are not applied directly to the wave file in use until you save the file—in essence, *Cool Edit Pro* uses "delayed destructive editing". When you open a wave file in *Cool Edit Pro*, it places a copy of the file in a temporary directory, and it uses that copy for editing. The original file remains unaltered until you choose to save any changes you've made (for example, by choosing Save from the File menu). When you save your changes, *Cool Edit Pro* overwrites the original file what you've done to the copy. Therefore, destructive edits in *Cool Edit Pro* do not directly affect your original wave file. Edits performed in the Edit Waveform View, like cut, paste, and Transform effects (like reverb), are destructive in nature, so when you save your file again, those edits are applied to the waveform itself. Edits made in the multitrack View, such as a cut, splice, or volume change, are non-destructive.

This same model of delayed destructive editing also enables *Cool Edit Pro* to offer multiple levels of Undo. When you apply destructive edits to (the copy of a) wave file, *Cool Edit Pro* retains a copy of the file as it exists before the edit. It does this for each edit you perform, enabling you to travel back through your edits to previous states of the wave file. Multiple Undo you tremendous freedom in working on wave files without having to worry about "destructiveness" in the least; however, it also requires additional hard drive space to store these copies (copies are automatically deleted from you hard drive when you close a file or exit the program). You can disable the Undo feature if your disk space runs too low.

Real time preview

Cool Edit Pro offers real time preview for many of its DSP effects. This means that you can monitor the processed signal before applying the effect to the waveform (remember: the effect is actually applied to a copy of the waveform, and can be freely undone). The preview feature updates in real time, meaning that changes you make to effect parameters while in the dialog for that effect become audible immediately, while the audio is playing. Keep in mind, however, that this system performance affects the preview feature. On slower systems, some effects may tend to break or skip during preview.

Multitrack

Cool Edit Pro's multitrack environment enables you to place any number of waveforms into different tracks for simultaneous playback and mixdown. This mixing process involves combining all of the placed waveforms into two (or more) channels for output. Because you can edit, add, and subtract waveforms in the multitrack environment, *Cool Edit Pro* must constantly watch for changes to the multitrack session, such as a moved or deleted waveform, a volume change, or new material recorded into a track. When something is altered, *Cool Edit Pro* must immediately work that change into the mixed output, and it does so through background mixing. Background mixing occurs behind the scenes much of the time that you are in the multitrack environment, and is generally very fast. The faster your system—especially your CPU and hard drive(s)—the faster *Cool Edit Pro* can mix in the background. The mix progress meter in the lower left corner of the Multitrack View constantly shows how far along *Cool Edit Pro* in the process of mixing your session. The meter goes from "empty" to "full" as it mixes the session, and it turns bright green when the entire session is fully mixed. You can think of this meter as a "ready" indicator; however, you need not wait for the meter to reach completion entirely before beginning playback. As stated above, *Cool Edit Pro* continuously mixes, and will continue to do so while playing, so you can safely begin playback when the mix progress meter is about half-way up. In general, if the background mix is not sufficiently completed, you will hear a break-up or skipping when playing back the mix. If that happens, just wait a few seconds; *Cool Edit Pro* will usually "catch up" very quickly, and you can begin playing again.

Again, playback can be directed to a pair of outputs (as in a single stereo sound card) or to multiple outputs (as in multiple stereo sound cards, or a single card with multiple outputs). *Cool Edit Pro* generates a mix for each set of outputs used. If you are using one stereo sound card, *Cool Edit Pro* generates just one mix, but if you have multiple outputs, it must create a separate mix for each output device (typically a stereo pair). The additional mixing required for multiple outputs demands more processing power and therefore tends to slow down the mixing process.

Wave Files and Session Files

If wave files are the building blocks of songs, session files are the recorded songs themselves. Managing all of these files and the relationships between them can be tricky. For example, if you create a session with a file called C:\MYWAVES\HIHAT.WAV, and later use another program (like Explorer) to move DRUMS.WAV to a new subdirectory called C:\MYWAVES\DRUMS, *Cool Edit Pro* won't know you moved the file the next time you load the session. Try to keep this in mind as you manage your wave and session files so that you don't end up losing tracks or objects inside your sessions. If you want to move an entire session file and all of its embedded waves to a new directory, you can use /File/Save Session As in Multitrack View and check the box labeled "Save copies of all associated files" in the Save Session dialog.

Function Presets

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

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

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

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

If you find that your presets aren't being saved, please check to ensure that the cool.ini file is not more than 64K in size. This should never happen, but it just might be possible if you have hundreds and hundreds of presets.

Some Cool Tips to Help You Along...

Edit View

Don't forget.. Try **right-clicking on everything!!!!!!!!** You'll be surprised how many option shortcuts can be accessed from the main window in this way.

It should be remembered that computers will often vary from one system to the next (depending upon their CPU speed, hard disk capacity and supporting hardware). It's the intention of *Cool Edit Pro* Edit that your data be processed as fast as possible... however, if your system simply can't keep up with the task that you've given it, you can change the system variables to optimize *Cool Edit* to your current system using the **Systems** tab within the **Setup** menu.

Whenever multiple files are open, you can easy select between these files using the Control-Shift keys (which shifts between open files in the forward direction) or using the Control-Shift-Tab keys (which shifts between open files in the reverse direction).

Don't under-estimate the power of the Cue List within the Edit View mode. If a waveform range or cursor position that you know will be of particular interest to you in the near future, simply press **F8**. This will add the selected range to the Cue List. Calling up the Cue List will allow you to recall any range or cursor point within the session, simply by double-clicking on the cue selection.

Pressing **F8** will automatically add the currently defined range to the **Cue List**. Once a cue point has been entered into the Cue List, the **yellow** marker points will be replaced with static **blue** markers. Individual cursor positions can be added to the list and will be displayed as a **red** marker. These cue entries can easily be referenced back to by simply double-clicking on the cue entry within the list.

Right-clicking on the **yellow** marker points will pull up a pop-up menu that will let you find the waveform's **next/previous zero crossing**, **Find Next Beat**, and enter a range into the **Cue List (F8)**.

If the amplitude levels between two waveforms don't match up at an edit or loop point, you may hear audible clicking as an edit passes or a loop repeats. To adjusts for this, you can revert to the original source file and use the Zero Crossings feature to match initial and ending levels. If there is a DC offset, this DC level difference can be set to zero by highlighting it and choosing the "Center Wave" preset from the Amplify function. Because centering takes out all frequencies below about 16Hz, it is completely safe to do without any ill side effects.

Interesting effects can be generated from the **/transform/noise reduction/noise reduction** module by choosing valid "foreground" audio as your profile "footprint" rather than background noise or hiss. For example, within a vocal line, you can select the vowel "O" to be used as the profile. Processing this vocal file (remember, it's best to process a copy of the original file) will then reduce or eliminate the "O" sounds... thereby creating a rather wild effect.

Multitrack View

One of the easiest ways to open up a session is by "associating" *Cool Edit Pro* to all files with the ".ses" extension. Once a session has been save to disk, an easy way to associate the extension with the program is to open up either **My Computer** or **Explorer**, locate and double-click on a session file. An association dialog box will automatically pop up asking you for a program file to associate the session to... Click on browse and locate the coolpro directory, select coolpro.exe and click **OK**. Now every time you double-click on a session file, *Cool Edit Pro* will automatically open up with the session fully intact and ready to go.

Don't under-estimate the power of the Cue List within the Multitrack View mode. If a waveform range or cursor position that you know will be of particular interest to you in the near future, simply press **F8**. This will add the selected range to the Cue List. Calling up the Cue List will allow you to recall any range or cursor point within the session, simply by double-clicking on the cue selection.

Pressing **F8** will automatically add the currently defined range to the **Cue List**. Once a cue point has been entered into the Cue List, the **yellow** marker points will be replaced with static **blue** markers. Individual cursor positions can be added to the list and will be displayed as a **red** marker. These cue entries can easily be referenced back to by simply double-clicking on the cue entry within the list.

You can add some cool effects to an existing audio file in the Multitrack View window by **shift-right clicking** on the waveform (possibly containing vocals, a guitar lick, claps... you name it) and creating a duplicate image. The duplicate waveform can then be placed onto a new track (possibly below the existing track)... and then shifted in time (relative to the original file), to create cool phase shifts (small time offset), delays (15-50 ms offsets) or discrete echoes (50 ms or greater offsets)... without having to alter the original audio file waveform. Go ahead... create a number of image waveforms and see just how wild of an effect you can come up with... You might want to reduce the volume levels slightly for each image, as the combined waveforms will increase the track's overall level in the mix.

Using the **Show Volume-** and **Show Pan Envelopes**, you can also change the relative volume and pan levels so that the effect will change over time or stop/start at a particular time.

Another way to add effects to an existing audio file in the Multitrack View window is by creating an actual copy of the waveform (this can be done using the **Create Copy** command in the Multitrack View mode or by saving the file under a new name using the **Save As** command within the Edit View Mode). Once a copy of the original file has been made, it can be processed in any way you'd like and then be placed onto another track, such that it will simultaneously play back with the original file. Using this method, you can then vary the effected track's volume and pan controls (in a non-destructive, automated fashion) without altering the original audio file in any way.

Although the Transform function doesn't exist within the Multitrack View, you can simply double-click on a multitrack waveform (which will open up the selected waveform into the Edit View window)... From here, you can use the **/Edit/Save Selection** option to create a copy of the file that can be effected using any Edit View Transform function... Once done, the newly processed file can be placed into the Multitrack View window.

By choosing the **MixDown** option, you can mix any number of instruments down to a single track or stereo track pair. In this way, you can create a submix that will be imported into the Edit View window for further processing, or for re-insertion back into the Multitrack window. For example, you could record several backing vocal tracks and then mix them down to a stereo track pair for further processing or for reducing the strain on the CPU/hard disk of having to simultaneously mix the numerous music bed and vocal tracks at one time.

You could also use the **MixDown** option to create a single composite track of several existing takes. For example, let's say that we've done extensive editing on a guitar track (involving several takes and countless edits). Personally, I'd consider it wise to mix my efforts down into a composite track... if nothing than for safety's sake.

An easy way to enter a waveform range from the **Edit View** into the **Multitrack View** is by selecting the **/Edit/Insert In Multitrack** option. This simple-yet-powerful function simply takes the highlighted **Edit View** range and pastes it into the **Multitrack View** window (within the next available track at the beginning of the session's timeline).

Say, for example, that you wanted to loop a short percussion riff. A simple way to accomplish this would be to use the **/Edit/Find Beats** function to easy search for the loop points and the select **Insert In Multitrack**, switch to the **Multitrack View** mode and Choose the **/Edit/Loop Duplicate** function to create the number of loops that you want.

If you'd like to add a previously created session to your current session, this can easily be done using the **/File/Append to Session** command. For example, lets say that we'd like to insert an existing multitrack composition that's been saved as a session. Appending the desired session into our current workspace (which will place them at the beginning of the timeline, onto the next available set of open tracks) we can now select and move the appropriate session waveforms into place (often using the **Ctrl-Right Click** mouse command to select multiple waveforms)... and Voila... you've integrated two sessions into a whole new composition.

Spline Curves

Some DSP effects have user definable charts in them, and some of these charts offer a **Spline Curves** option. Normally, a graph is created by clicking control points on the chart, and that graph is used to control the given effect. When the **Spline Curves** option is active, instead of using straight lines to connect the control points, a curve is inserted that smoothly transitions from one point to the next is used instead. Most of the time, this curve will not go through all the control points, but rather the control points will be used to control the shape of this curve.

In order to get the curve closer to a control point, you must create more control points near the point in question. The more control points that are clustered together, the closer the spline curve will be averaged to those points.

Whenever very smooth curves are desired instead of straight lines (with their associated discontinuities at the control points), it's best to use the **Spline Curves** option.

The functions that make use of spline curves are: Envelope, FFT Filter and Brainwave Synchronizer.

Using ACM

Cool Edit Pro 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).

Using Cool Edit Pro with Cakewalk Pro Audio

When you install Cool E, it automatically registers itself for use with **Cakewalk® Pro Audio™** from Twelve Tone Systems. This means you can use *Cool Edit Pro* 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 will automatically load that wave into *Cool Edit* for editing.

You must have Cakewalk Pro Audio 5.0b or higher to integrate *Cool Edit* with it.

Mono to Stereo Conversion

Use the /**Edit/Convert Sample Type** function to convert a waveform from 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.

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, you may wish to use the Channel Mixer to mix both channels at 50% first.
- Highlight the section you wish to place on the left channel, and choose /**Edit/Copy**.
- Select the new blank stereo waveform, and choose /**Edit/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%.

Adjusting recording and playback levels

Cool Edit Pro 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:

- Click on Start - Programs - Accessories - Multimedia - Volume Control
- 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.
- 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.

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.

Spline Curves

Some DSP effects have user definable charts in them, and some of these charts offer a **Spline Curves** option. Normally, a graph is created by clicking control points on the chart, and that graph is used to control the given effect. When the **Spline Curves** option is active, instead of using straight lines to connect the control points, a curve is inserted that smoothly transitions from one point to the next is used instead. Most of the time, this curve will not go through all the control points, but rather the control points will be used to control the shape of this curve.

In order to get the curve closer to a control point, you must create more control points near the point in question. The more control points that are clustered together, the closer the spline curve will be averaged to those points.

Whenever very smooth curves are desired instead of straight lines (with their associated discontinuities at the control points), it's best to use the **Spline Curves** option.

The functions that make use of spline curves are: Envelope, FFT Filter and Brainwave Synchronizer.

Creating Sound Effects

Two types of sound effects can be created using *Cool Edit Pro*: 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 to sound sources, noise and tone, practically any effect can be created.

Noise Effects:

Waterfall, wind, and rain

Thunder, snare drum, cymbals, jet engines

Fantasy sounds such as time tunnel vortex, etc.

Tone Effects:

Siren, pipe organ, piano, and other musical instruments

Space ship sounds, whining, whistles, etc.

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

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.

Q: I can record just fine in Edit View, but when I try to record in Multitrack View, I get a flat line or nothing at all. What's wrong?

A: Your sound card and its drivers probably don't support simultaneous play and record (known as "full duplex" capability). When you're in Multitrack View, *Cool Edit Pro* is **always** in full-duplex mode, so even if you're only trying to record one track, you must still be able to record and play at the same time to do it in Multitrack View. If it doesn't work for you, use Edit View to record and then insert the recorded track(s) into Multitrack View. If you think your card ought to support simultaneous play and record, contact the manufacturer of the card to see what limitations may apply.

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.

Q: I am using *Cool Edit Pro* 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 per the standard), and by clearing out all information in the **/Options/Info dialog**. Uncheck the Fill * fields automatically checkbox to prevent *Cool Edit Pro* 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, there will be problems starting and stopping audio too quickly.

Q: Some of the features I want to use are unavailable ("grayed") 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 Reductions, and others. Try loading a file and selecting some or all of it with the mouse; you should see those features "light up" in the menus and toolbar once you've done this.

Q: Can *Cool Edit Pro* convert files, either singly or in batches?

A: Yes! To convert a single file, use **/File/Open** to open the file in *Cool Edit Pro*, select **/File/Save As**, and specify the file type at the bottom of the Save As dialog.

Q: I can't record. 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. Try adjusting your sound card's record levels in its mixer or in the one built into Windows (Start - Run - sndvol32exe).

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 you 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 is not somehow conflicting 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/Devices** 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.

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

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

Q: Why are some functions not selectable?

A: Some functions require that you select part of the waveform before they are selectable, while others 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* doesn't seem to know the format of waves that have headers.

A: If you find you cannot load or save .WAV format waveforms, if you try to load a .WAV waveform file and the "Choose Sample Rate" dialog appears, or have any trouble with loading and saving in the proper format, this problem could be the result of 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. You should have as much free space on the temporary files drive as twice the size of the largest file you will be working with, as a rule of thumb.

Brainwave Synchronization

By David Johnston

About Brainwave Files

Cool Edit Pro's Brainwave feature (**/Transform/Special/Brainwave Synchronizer**) works like many meditation tapes and light/sound devices on the market, which range in price from $200 to $500. There are even boards available with plug in glasses (which have blinking lights) for your PC in the price range of $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**.

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!

Major brainwave pattern frequencies and possible uses for brainwave synchronization

Delta	1-3 Hz	Deep sleep, lucid dreaming, increased immune functions.
Theta	4-7 Hz	Deep relaxation, meditation, increased memory and focus.
Alpha	8-12 Hz	Light relaxation, "superlearning", positive thinking.
Beta	13-25 Hz	Normal state of alertness, stress and anxiety.
Gamma	30 Hz on up	Hyper-awareness
High Gamma	200+ Hz	Various effects

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.

Sleep Replacement - A 30 minute session at 5Hz replaces about 2-3 hours of sleep, allowing one to wake up in the morning more refreshed. Try listening 1/2 hour before waking up in the morning, or 1/2 hour before going to bed.

Improved Sleeping Patterns - Any of the Alpha and Theta frequencies (8Hz to 4Hz) for 30-45 minute sessions at the same time each day.

Treatment of Insomnia - Choose between 4hz and 6hz for starters (the first 10 minutes), then go into frequencies below 3.5hz (for 20-30 minutes), settling on about 2.5hz before fading out.

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 while using visualization techniques.

Alleviation of Migraines and Headaches - Experiment with Alpha and Theta combinations. Try and visualize the pain getting smaller and smaller until it disappears.

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.

Accelerated Learning - Choose about 7hz to 9hz while playing any learning tapes, like foreign language tapes, etc. to increase comprehension. Also, while studying, take breaks every half hour and listen to 10 minutes of Alpha (10Hz) while reflecting on the material you just learned.

Subliminal Programming - Choose 5hz to 7hz while playing your favorite subliminal tapes, or make your own by recording some affirmations, and mix pasting (Edit:Paste Special) them from the clipboard at barely audible volumes.

Improve Intuition (or ESP?) - Theta frequencies help in this area, 4hz to 7hz.

Reaching Higher States of Consciousness - Theta again, with daily half hour minimum sessions. Give at least a month for results.

Quick Refresher on long days - Low Alpha 8hz to 10hz for about 15 minutes works well. Sort of induces a cat-nap.

Increased Immune System - Relaxing to Alpha and Theta combinations daily. Learning how to relax, and relaxing more often can lower blood pressure and increase the body's natural defenses. Using Alpha Synchronization (8Hz to 12Hz), expect similar increases in the neuro-chemical levels of Norepinephrin (11%), Serotonin (21%) and Beta-Endorphins (25%).

DISCLAIMER

> By using this program, you agree that the author will not be responsible for any damage as a result, direct or indirect, of using this program. The author makes no claims about the effectiveness of these sounds for any particular

purpose. The user is encouraged to do his/her own research into the area of brainwave synchronization via auditory stimulation.

WARNING

Sounds generated by the wave function may not suitable for epileptics or persons undergoing psychiatric treatment.

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. These sounds may be recorded using any stereo cassette recorder and played back on any stereo cassette player without losing effectiveness. 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, and your tape deck does not need to be able to record frequencies this low either. The brain *does* however respond better to the lower frequencies because of the nature of the synchronization algorithm, so the better the headphones you buy, the more dramatic the results may 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. The active ingredient, so to speak, are the frequencies from about 40Hz up to about 2khz depending on the frequency being encoded and the intensity.

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. The channel mixer will also allow you to put in just the amount of correlation you desire (for example, a 20% mixture of both channels, leaving the rest untouched.) 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, Paste Special (with overlap) one tone on top of the other. Use low frequency tones, like 50Hz to 120Hz for best results. These tones, by themselves, will help coerce the mind into the state associated with the difference between the frequencies. For example, for a theta state of 6Hz, use a 70Hz and a 76Hz tone. Combining this tones sample with an existing brainwave file, by overlap pasting at a quiet volume (20%) is even more effective.

Generate Noise - Use the Generate Noise function (pink and brown work best) in any of the modes: mono, inverse, or spatial stereo (independent channels noise will **not** work as a carrier for brainwave frequencies at all, since there is no correlation between the left and right channels). 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. Inverse works 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 rearranged, otherwise it would sound discontinuous. Creating brainwave files using the flat file method will be discussed first, since it is more straightforward.

Flat Brainwave File Generation

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

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

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

Cued Brainwave File Generation

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

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

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

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

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

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

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

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

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

Authoring Brainwave Files

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

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

Frequencies of 8-10Hz correspond to an alpha state -- light relaxation, like a quick afternoon siesta. Frequencies of 6-7Hz correspond to a theta state -- meditation. 4-5Hz correspond to deep relaxation. You can create a session that is constant, in one of these states, or create a session that dynamically flows from one to the other. When going down in frequency, give the listener about one minute to *catch up*, and stay in sync with the wave. Going up in frequency does not require the listener to catch up. In other words, if you go from 6Hz down to 4Hz over a 20 second timespan, 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 awakened 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.

High Gamma frequencies of 200Hz or more seem to help in relaxation, and do something no doubt. 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

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.

Choose **Generate:Noise**. Select **Pink, Spatial Stereo** (500 µSeconds) 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.

Choose **Edit:Copy**. From now on, we will paste the noise in when we need it!

Make sure the noise is highlighted. If it is not, select all by double-clicking on the waveform until it is highlighted.

Choose **Add** in the Cue list, and give the entry a **Label** of **10Hz to 8Hz**, and a **Description** of "**Warm-Up**"

Choose **Transform:Wave** to bring up the brainwave dialog box. Enter **10** for the **Highest Frequency**, and **8** for the **Lowest Frequency**, and an **Intensity** of **35**. On the graph above, click the leftmost dot, and drag it to the top of the graph. Click the rightmost dot, and drag it to the bottom of the graph. This will product a frequency encoded at 10Hz at the beginning, and glide down to 8Hz by the end. Choose **OK** to generate the encoding. This shouldn't take nearly as long as it did to generate the noise.

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.

Choose **Edit:Paste** to insert another copy of the original noise that we had copied originally.

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
(Graph should go from left=highest to right=lowest)				
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
(Graph should look like an upside-down "V" for Spike)				
5-8-5	Spike	8	5	50
(Graph should go from left=lowest to right=highest for Awake)				
5 to 12Hz	Awake	12	5	40

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.

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

How to use brainwave synchronization files

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 you have mixed two different (but similar in frequency ranges) brainwave files together, you may notice 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.

References on Brainwave Synchronization

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

Adrian, E. D. and 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 and 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 and 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. and Fine, T. H. (Eds.), *Restricted Environmental Stimulation*, New York: Springer-Verlag.

Cade, C. M. and 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., and 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 and Sons.

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

Donker, D. N. J., Nijo, L., Storm Van Leeuwen, W. and 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. and 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. and Shor, R. E. (1970). "Verbally-induced behavioral response during sleep." *Journal of Nervous and Mental Disease*, 1, 1-26.

Evans, C. and Richardson, P. H. (1988) "Improved recovery and reduced postoperative stay after therapeutic suggestions during general 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. and 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., and Green, A. M. (1986). "Biofeedback and States of Consciousness." In B. B. Wolman and M. Ullman (Eds.). *Handbook of States of Consciousness.* New York: Van Nostrand Reinhold.

Harding, G. F. and 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. and 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. and 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 and 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 and 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., and Lankton, C. H. (1983). *The Answer Within: A Clinical Framework of Ericksonian Hypnotherapy.* New York: Bruner/Mazel.

Leman, K. and 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 and Wilkins.

Mavromatis, A. Hypnagogia: *The Unique State of Consciousness Between Wakefulness and Sleep.* New York: Routledge and 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. and 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., and Meichenbaum, D. New York: John Wiley and Sons, pp. 49-87.

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

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

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

Sittenfeld, P., Budzynski, T. and 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., and 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. and 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. and 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.

More Information About Filters

by David Johnston

Filters are used to modify a signal's frequency content. As such, most filters allow some frequencies to pass unchanged (The Passband), while some are reduced or eliminated (the Stopband). This, however, is not the only effect a filter can have. Infinite Impulse Response (iir) filters also affect the relative time-delay at various frequencies differently. This is known as Phase Shift or Group Delay, and it can be a desirable effect, an undesirable effect, or something you don't care about. The phase shift "curve" of a filter defines its ringing and overshoot properties, which may or may not matter in a given application. The design of a given filter is a problem with many variables, which can be traded off against each other depending on what's important to the designer at the time. Several "standard" approaches exist, and these are named after the person who first invented or popularized them. There is nothing particularly magic about any of them, it's just that each has been designed to maximize one variable at the expense of the others and is therefore in some sense ideal if what you want is what was optimized in the given type. See below for more information on each of the filter types available in *Cool Edit Pro*.

Bessel: Bessel filters have the best phase response possible for an analog filter, with almost a constant time-delay at all frequencies, and virtually no ringing or overshoot. This comes at the expense of a "flat" passband, which is droopy near the corners, and high rejection in the stopband. Bessel filters are used most often where preserving the shape of pulse-type signals is of paramount concern. To accomplish this, the rejection in the early part of the stopband is the poorest of the types. Bessel filters eventually reach a falloff rate of 6 dB/octave per order, but this doesn't happen until well into the stopband. The nearly flat-phase property of Bessel filters only somewhat translates into the digital version, so these are of questionable use. If you need "perfect" phase, there is another way to get this in a digital filter, with an FIR filter, provided in the FIR Filter or Graphic Equalizer functions.

Butterworth: Butterworth filters are optimized for a flat passband at the expense of the other variables. As it turns out, phase shift, ringing and overshoot are still not too bad-- usually under 10% for most designs. Due to other issues beyond the scope of this discussion, Butterworth filters are also "computer friendly" in the sense that they move well into the digital domain with their properties intact, and are among the easiest to compute at high orders with good precision. Butterworth filters have much better stopband rejection than Bessel filters, and only slightly worse than the Chebychev types, making them probably the most used types in either analog or digital domains. "Try Butterworth first" is a good rule, as this compromise handles more cases well than any other. The stopband rejection for Butterworth types is 6 dB/octave/order like the Bessel, but this value is reached much sooner in the stopband.

Chebychev type 1: for this type, passband "flatness" is traded off to get the best possible rejection in the first part of the stopband. Phase response is the worst of the types, and ringing and overshoot goes with this. They are used where rejecting the stopband seems almost more important than keeping the passband accurate. In the golden analog days, Chebychev filters were used whenever you needed to build a lot of them and the fact that you could sometimes do a "good enough" Chebychev with a lower order than a Butterworth saved some money. Since this is now all in software, and quite fast, the cost of additional filter sections is minimal, and there is less need for this type. Also, because Chebychevs tend to have higher "Q" internal sections, they are more prone to overflow troubles at high orders. Like Butterworth and Bessel, the ultimate slope between the pass and stop bands is 6 dB/octave/order. All you get with Chebychev over Butterworth is passband ripple, worse phase response, and somewhat quicker initial falloff. If you design a Chebychev with 0 passband ripple, you get a Butterworth.

Chebychev type 2: this is also sometimes called a Cauer filter though that is not quite correct. A "true" Cauer has more ripple in the passband and better rejection--but much worse phase, ringing etc. *Cool Edit*'s version of Chebychev type 2 consists of a Butterworth filter for the passband, with notches (or zeros) added to the stopband for additional rejection. In between the notches, the signal "bounces back" somewhat, so there is now ripple in the stopband instead of the passband. This is still a useful compromise in the digital domain. You get most of the "goodness" of Butterworth with potentially great stopband rejection. You must specify how much the signal is allowed to "bounce back" for this type. You are basically trading off good "initial" rejection for worse "ultimate" rejection compared to types that have no "bounce". This is a decent compromise for many problems. In addition, since the passband is Butterworth-like, most of the phase shift or group-delay distortion falls into the stopband, where presumably you don't care about it. With this type, you can build a quite impressive bandstop filter that leaves the passband almost totally unaltered. Because the passband is Butterworth-like, this type also has fewer internal overload problems than the Chebychev type 1. Passband "ripple" is reported or specified for this type as well. In this case, the passband has some "droop" near its edge due to the complete cancellation occurring just after it. This "droop" is lessened by going to higher orders.

Which filter type should I use? You can construct even a Bessel filter with decent rejection if the order is high enough. But it will still have *some* phase error, and this goes up with increasing order, as do errors due to non-infinite-precision math. Conversely, the phase response and ringing of a low order Chebychev aren't too bad. But for the *huge* majority of cases, the Butterworth is simply the best compromise you can make. This is definitely helped by the low "cost per order" in this domain. In analog, when you need more than about a 6_{th}-order filter, most authors question whether you are trying to do with a filter what you should be doing with some other technique, such as a phase locked loop. This is partly because higher order filters are hard to make in analog due to the necessity for accurate tuning etc., and extremely accurate components are not readily available. Contrast that with the fact that this package can do a more than 60_{th}-order Butterworth filter with good accuracy. However, an order that high is rarely if ever needed for any "real" problem, though it sure looks impressive on a plot! Most audio uses of this will never need more than about 6^{th}-10^{th} order if even that. You'll find that really steep rejection, and the phase shift associated with that, sounds "fakey" or "overprocessed" -- which of course *could* be what you're after. For reference, most tone controls are first order filters, most equalizers (like the

Parametric Equalizer) are second order. So if you're using these in a production environment, you will probably want to use the lower order filters.

As you might guess, a lowpass filter passes the low frequencies, and stops the high frequencies. The highpass is just the reverse. A bandpass filter rejects frequencies outside some "band", while a bandstop filter rejects frequencies inside some band.

To specify a lowpass filter some numbers are needed. The most important one is the "cuttoff" frequency. This would be the highest frequency that you want the filter to pass -- or the "end" of the passband. In many cases you will also specify another frequency for the "beginning" of the stopband. This is done in conjunction with a dB number to specify how much attenuation or rejection must be achieved by this frequency. In our package, this is what is going on when Transition Bandwidth is chosen (and order is filled in automatically). You specify what you want, and *Cool Edit Pro* figures out how big a filter that would take for you. For a highpass filter, it's the same, except that the lowest frequency is now the stopband ending, and the next frequency is the start of the passband. You may well be able to figure out how this all translates to the band-types, but now you need twice as many frequencies to specify the filter.

Cool Edit Pro attempts to give as much flexibility as possible when designing filters. You can specify pass and stop band frequencies and an attenuation dB, and *Cool Edit Pro* will do the rest. However, advanced users may want to set the order of the filter for a number of reasons. *Cool Edit Pro* lets you do that too when Order is chosen. With this option, you specify passband corner frequency and order, and take the stopband you get. You might want to do this for several reasons. First, high-order filters have lots of phase shift, and if you are going to use the filtered signal in conjunction with the original signal or other filtered signals, you can encounter problems with this: things no longer happen at the same time as they did before filtering. Another reason you might want to specify the order yourself (other than speed) is for the Chebychev type 1, which has passband ripple. For a 3 dB ripple lowpass filter of this type, the gain at DC is -3dB for even order filters, and exactly 0 dB for odd order. So if DC gain is important, you will want to specify order yourself for this type. There is also an option 2 for this type. For this, you specify everything but passband ripple, and *Cool Edit Pro* picks that. This can lead to some pretty strange-looking filters, but is good to give you an idea of the tradeoffs involved if nothing else. This is a holdover from the bad old days when filters were expensive and one was always trying to push what could be done with a low order filter. Now it's just there as a learning tool.

Why is -94dB SNR better than infinity?

By David Johnston

Pure digital silence (that generated with the Generate Silence function in *Cool Edit*) is actually at a signal-to-noise ratio of negative infinity (digital zero's all the way). The quietest, most faint signal that could appear on a CD in 16-bit is around -94dB (closer to -93.86). With very faint signals around the -90dB range, if the samples were simply truncated (those between -1.0 and 1.0 get set to zero) then very noticeable unwanted artifacts would be introduced. These would be harmonic distortions that would be correlated with the signal being truncated which is 99% of the time unwanted in the audio business. It may sound great for making distorted guitar effects, but not great if your piano sounds like that same distorted guitar. Also, as the audio gets quieter, as in doing a fade out, straight truncation will cause the audio to flutter between total silence and -94dB without any amplitudes in between. So generally, truncation is bad because it introduces this quantization noise.

So why is -94dB better than -infinity, especially when doing Fades? When doing fades in *Cool Edit*, it is possible to fade out a signal well below -94dB and still hear the signal if dithering of transform results is enabled under Settings/Data! This is because the noise itself still contains audio information that is audible. Signals as quiet as -118dB (that's 22dB below the -96dB so-called "limit") can still be heard, even though the S/N ratio is still -94dB. In other words, the S/N ratio is 94dB while the dynamic range is closer to 118dB. So to fade out completely, you want the loudest audio near the end of the fade when fading out (or at the quietest portion of the fade) to be faded out to at least -118dB.

To test this theory, go to a regular section of music, use Analyze/Statistics and note the maximum RMS power. Then subtract it from -104 (for example, the max RMS power was -10dB, so -104 - (-10) = -94. Go to Amplify and amplify by this amount (-94dB in the example). This will make the maximum RMS power -104dB. When you use Analyze/Statistics, you see a reading of about -93.8dB though, but the audio is really still down there. Now Amplify it back up about 80dB and listen to it. You will be able to hear the music still, behind the background noise! If you did not have "Dither Transform Results" checked, you would have gotten digital silence, and would not hear anything when boosting the result back up 80dB. With "Dither Transform Results" enabled, the music you were listening too had a power no greater than -104dB, which is 8dB quieter than that -96dB limit of 16-bit audio.

Now, why is -83dB better than -94dB? You probably will at first think this sounds nuts, but a -83dB SNR really can be perceived as higher quality than a -94dB SNR. Above, *Cool Edit* uses a straight "triangular" probability distribution function (p.d.f.) for dithering, which is flat across the spectrum (there is just as much noise at 500Hz as there is at 10KHz). If that background noise is dithered using a different p.d.f. and shaped such that most of the noise is pushed into the higher frequencies, above the range of human hearing, and above which most analog equipment will cut off due to filtering, then the perceived noise level will go down. In *Cool Edit*, using a shaped triangular p.d.f., and the "C3" noise shaping curve, the S/N ratio (as seen in the minimum RMS power under Analyze/Statistics) will increase to about 11dB to -83dB. But, since most of this noise is above 15.5KHz, and because the ear itself is already much less sensitive to audio above 15.5Khz, we should only analyze the noise below 15.5Khz. Doing this reveals that the S/N ratio from 0Hz to 15.5Khz is 94.5dB. If just we look at the range in which most music resides, 50Hz to about 8KHz then the S/N ratio is a whopping 109dB! And around 3Khz to 4KHz, where the ear is extremely sensitive, the S/N ratio is 135dB. This means that a lot of the music below the 96dB limit can be heard clearly, if the analog equipment's D/A converters and amplifiers own noise doesn't mask it out.

This can be demonstrated using *Cool Edit Pro* by taking a short 10 second snippet or so of your favorite music, and converting to 32-bit (Edit->Convert Sample Type). Then reduce the amplitude as in the previous example so the peak is no greater than -104dB. Then go to Edit->Convert Sample Type again, and convert to 16-bit (always working at 44.1Khz) and choose the Shaped Triangluar p.d.f., and Noise Shaping C3 curve. If you used the same music for both examples, you'll notice that the noise-shaped version here sounds much clearer, and much fainter sounds can be heard than before.

Different noise shaping curves are provided for use at different sample rates, and quality levels that will depend on the source audio being used.

Using this method, one can trade off frequency response for signal-to-noise ratio also. With 48KHz sampled audio at 20 bit resolution for example (which is converted to *Cool Edit*'s internal 32-bit float format) dithering and noise shaping can be used to boost the SNR while sacrificing some of the frequency response - bringing the frequency response down from a maximum of 24KHz to about 20KHz. The loss in frequency response will not be noticeable, but the gain in SNR will be noticeable. Using noise shaping curve "D", convert the 32-bit 48KHz audio to 16-bit 48KHz. Most of the quantization noise will be pushed up above 19KHz, well out of the range of most ears. If converting to 44.1KHz, use curve "D" or curve "C". The 44.1KHz audio will retain the subtler audio information that was present at the original 20-bit resolution at 48KHz sample rate. For most practical purposes, you will be getting 48KHz 20-bit quality in 44.1Khz 16-bit audio.

These same principles can be applied when going to 8-bit audio as well with some playing around with the parameters. At 8 bit though, the type of hiss may not be acceptable when doing noise shaping because it is so loud at 8 bits - again, it will be a matter of individual taste and the material being converted.

About *Cool Edit Pro*'s Creator

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 II'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 became some of the most popular shareware programs available: *Cool Edit*, upon which *Cool Edit Pro* is based, and *Kaleidoscope*, a screen saver for Windows that responds to music.

In 1995, David 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 a 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. It can also sound like a pianist playing with a tune, a violin section working on theme variations, or even a gunfight.

You can find out more about David's and Syntrillium Software's creations on Syntrillium's web site at **http://www.syntrillium.com**.

About the author

Official bio

David Miles Huber is widely acclaimed in the recording industry as a digital audio consultant, author, engineer, university professor and guest lecturer. In addition, he's also a professional musician (having performed, produced, and engineered over 15 of his own CDs.)

David received a degree in recording techniques (I.M.P.) from Indiana University and was the first American to be admitted into the prestigious Tonmeister program at the University of Surrey in Guildford, Surrey, England. He has authored numerous books on the subjects of recording and electronic music. His most prominent book Modern Recording Techniques (Focal Press - Boston/London - www.bh.com/focalpress) has been an industry standard text worldwide for over 16 years.

un-Official bio

Where's he from? -	Born in Connersville, Indiana to Oliver Wendel and Myrtle May Huber.
What were his first words? -	Whaaah! Seriously, folks, I've been told by my family that my 3rd word was "record!"
Where does he live? -	In the hills of Western Washington.. Don'tcha just love the mountains, the water and (ahem...) the rain!
What does he do for a living? -	He writes books (about seven, so far, including Modern Recording Techniques, The MIDI Manual (Focal Press), Hard Disk Recording for Musicians (Music Press), etc. He's an electronic musician (had to turn out 15 of his own CD's before he had the guts to say... Yea, I'm a musician!), He's writes for EQ magazine (www. eqmag.com) and teaches recording at the University of Washington (cause he's a natural-born ham). Any more questions?
Favorite drink? -	Scotch or diet Coke...
Astro statistics? -	Sagittarius sun, Gemini moon with a Capricorn ascendant.
Newest toy? -	A 1981 Suzuki GS450t motorcycle with less than 6k miles on it (not any more, I've been having too much fun with it!...) and not a scratch!
Favorite movies? -	Young Frankenstein, Betelgeuse, Hairspray and anything to do with Star Trek.

Contacting Syntrillium Software

If you have any comments, or a question to which you cannot find an answer in our documentation, please feel free to contact us. Syntrillium Software can be reached during the hours of 9AM - 4PM PST (Pacific Standard Time) or by Fax or email at any time. Our web site is also available at any time for the latest information and updates, and for technical support. We have at our disposal, all of the popular communication technologies of the day:

Phone 9AM - 4PM PST (Pacific Standard Time)

+1 (602) 941 - 4327

Email

General: cepro@syntrillium.com

Technical support: prosupport@syntrillium.com

Fax

+1 (602) 941 - 8170

Web

http://www.syntrillium.com

Technical Support

Cool Edit Pro is designed to be easy to use, but on occasion you may need some explanation or help with an issue. In most cases, you'll find the help you need right in your computer; this Help file, and the *Cool Edit Pro* manual should contain the answers to most of your questions. In addition, you'll find extensive support material available around-the-clock at our site on the World Wide Web. If you still experience difficulties, our technical support staff is available via Email, FAX, or phone. We recommend sending your technical questions electronically (Email) for the most efficient handling of your request.

Before contacting Syntrillium for support, please be sure to check the following:

1. Please check our online documentation and manual for answers to your questions.

2. Make sure you have registered your copy of *Cool Edit Pro* with Syntrillium Software.

3. Be able to provide detailed information about your system, such as the CPU, sound card type, and hard drive space available.

4. Have your serial number ready if you call (located on back cover of the CD Jewel case).

Web Site Support

Syntrillium's web site provides an extensive amount of support materials, and the solution or answer you seek may already be posted. There may also be an update you can download to resolve the problem.

- http://www.syntrillium.com/support/

Email Support

You may submit your technical questions and problems via Email. The technical support mailbox is checked throughout the day for new messages. Syntrillium responds to emails within two business days (same day reply in most cases).

- Include your Name, phone number, serial number, as well as your return Email address.
- Describe your problem or question in detail.
- Please do NOT attach any file to your e-mail message unless requested.
- Email address: prosupport@syntrillium.com

Phone Support

When calling for support:

- Have your serial number ready. The serial number is located on the back cover of the CD Jewel case.
- Be ready to describe the problem or question in detail.

- Have your computer turned on and be ready to duplicate the problem on your system with our technical support representative.
- Support hours are Monday through Friday between 9:00 am - 4:00 p.m. (PST), excluding holidays.
- Call +1 (602) 941-4327

Fax Support

You may submit your technical questions and problems via facsimile.

- Please indicate on the cover page that the FAX is intended for technical support.
- Have your name, company, address, telephone and FAX numbers clearly on the cover page.
- Indicate your serial number on the FAX.
- Please describe your problem or question in detail.
- FAX: +1 (602) 941-8170