Audio Wave Editor

The **Audio Wave Editor** is a simple waveform editing tool which lets you visually manipulate any PCM encoded *.wav file you may have.

Introduction.

The Menus.

Over View

The **Audio Wave Editor** is a hard disk based waveform editing tool, this means you can edit any size file you want. It is also an MDI program, that is many files can be open at once.

The first thing to do is Open a *.wav file from menu File|Open, unregistered users are limited to a 10 second section.

The **Top** panel contains the media player controls, a file length indicator, and a free Disk space indicator (this indicates space available on the Disk where the TEMP directory is located).
A **second** panel should have appeared, containing the Start and End section indicators, a button to **play** the selected section, a button to continuously **Loop** the selected section, a position indicator, and a playing position indicator.

The graphical representation of the selected file should also have appeared in it's own window, with it's name, sampling rate, resolution and channels contained in the title bar.

Selecting a Section of the track.
With the cursor over the selected file, click the left mouse button and drag the mouse to select a section of the file. Notice how the Start and End section indicators track your position. The Start and End section indicators define the position for the selected section that the Editing, Tools, and Effects commands use.

An alternative to position the selected Section, is to select the position by manually rolling the up/down buttons on the Start and End selection indicators. You can adjust the step size, by altering Options|Fine Time Increments.

To quickly Select All the track press Q, or from menu Edit|Select All.
To Select and view All the track press Ctrl+Q, or from menu View|Full View.

Playing a Section of the track.
Clicking the Section Play button, or hitting the Space Bar, will start playing the selected section of track. If the Start and End points are in the same place then the track will be played from this point to the end of the track. With this feature you can dynamically jump about the track by clicking the mouse (left button) at the position you want to move to, and hitting the space bar.
Depressing the Loop button will cause the selected section of track to loop.

To stop the track playing you must click the stop button of the media player in the top panel.

Zooming In and Out.
To Zoom In, select the section to examine then press Ctrl+Z, or from menu View|Zoom In.
To Zoom Out, press Ctrl+A, or from menu View|Zoom Out.

Notice how the: Zoom In (Ctrl+Z), Zoom Out (Ctrl+A), Select All (Q), View All (Ctrl+Q) are all grouped together for easy access.

Keyboard Shortcuts.
Extensive use of Shortcuts has been made for all the most commonly used functions.
Editing functions are typically require two fingers (Ctrl + ?), and use the Microsoft convention for cutting and pasting, the Tools and Effects typically require one finger.

Getting started
Working with Big files

Getting started

The first thing to do is Open a *.wav file.

Click the Section Play button, or hit the Space Bar, to start playing the Track.

Editing.
Editing functions use the Microsoft convention for cutting and pasting. Cut and Paste features working similarly to those of a word processor.

The **Audio Wave Editor** has a powerful function which will Trim the file to the selected length, remove the DC Offset then Normalise it, all in one go. The function can be accessed through Edit|Trim-Offset-Normalise, or Ctrl+N. Notice how Tools|Normalise uses (N), related functions on same key.

Multiple files can be open simultaneously.
Flip between the files with (Ctrl + Tab), this will ensure Start/Stop markers aren't moved, this is particularly useful when working with big files.

Using the Tools and Effects.
The Tools and Effects are **only** applied to the Selected section.

Try this:
Click Time Compress/Expand from the Effects menu (T), reduce the selected length to about 90%.
Click the section play button again, the pitch of the selected section of track will have increased.

To flip back to the previous version, select Edit|Undo (U). Clicking U again will Redo the effect, only one level of Undo is available.

With the original waveform, select Effects|Echo (E). A panel will be presented requesting Delay time and Fading parameters, for now use the default settings, click OK. An Echo will be quickly added to the selected section of track.
Play the track to hear the effect.

All the Tools and Effects have been made as intuitive as possible. More details of each function are available elsewhere in Help file.

Watching a Playing file.
Select All of the Track (Q), depress the Loop button and press the space bar to start playing.
Selecting Special|Frequeny Analysis (Ctrl+F). This will display the instantaneous FFT of the current position, but it will probably be changing too fast to make any sense out of. Click Stop and Close to return to the Waveform.
Try using Special|Spectral Averaging instead, select 8 spectrums. Instead of displaying the instantaneous waveform the FFT is averaged. Now you can watch the tonal variations in the track.

As with all tools like this the way to learn, is to play.

Over View
Working with Big files

Working with Big files

Since an Undo facility is a must at least two copies of the file are required. This is the **major overhead** when working with big files, the writing of an entire new version of the file to disk for each modification required. This slows down normal working, and gets significantly worse with big files.

This problem can be reduced by use of the **File|Save Section and Open...** command.
Select the section to be modified, use the above command to Save the Section and Open it for Editing.
Do the modifications on this smaller file. You can flip between the two files until the modification is as desired.

Provided you don't move the Start/End markers in the original file the **Edit|Replace** command can be used to perfectly align the modified file.
Once you're happy with the modified file; Select All(Q), and Copy (Ctrl+C) it all.
Go back to the original and Replace (Ctrl+R) the section, this deletes the original section and inserts this new section.

Over View
Getting started

File Menu Commands

Record.
Initialise a new file for Recording too, the recorder is activated and ready to start.
The sound source must be specified by the software supplied with your sound card, the sound can then be recorded to disk.
The file will be record in the format as specified in the Options|Recording Setup menu, the default being 16 bit, mono at 22kHz.
When recording from CD, use the built in CD player.

Playback File.
Open a file to be played whilst recording (only available for Full Duplex Sound Cards).

Open.
Open a new file.

Save.
Replaces the existing file.

Save As...
Saves a new or existing file.

Revert.
Restore the original file.

Save Section As...
Saves the selected section to a new or existing file.

Save Section and Open...
Saves the selected section to a new or existing file and Opens it ready for Editing.

Close.
Closes the active window.

Exit.
Quits the Audio Wave Editor.

Edit Menu Commands
View Menu Commands
Tools Menu Commands
Effects Menu Commands
Special Menu Commands
Options Menu Commands

Edit Menu Commands

Undo.
Undo previous editing.

Cut.
Copy and Cut section defined by Start and End positions.

Copy.
Copy section defined by Start and End positions.

Insert.
Insert copied section at Start positions.

Mix Paste.
Mix copied section with selected file at Start positions.

Trim-Offset-Normalise.
Trim the file, Remove the DC Offset and Normalise the amplitude.
Apply to the section defined by Start and End positions.
Fast way of implementing the most common functions used after Recording a new file.

Trim.
Trim the file to the section defined by Start and End positions.

Delete.
Delete section defined by Start and End positions.

Replace.
Replace section in one file with the copied section; particularly useful with big files.

Append.
Append copied section to End of file.

Adjust Sample Rate.
Change the Sampling Rate of the whole file.

Insert Silence.
Insert up to 10 seconds of silence.

Select All.
Reset Start and End positions.

File Menu Commands
View Menu Commands
Tools Menu Commands
Effects Menu Commands
Special Menu Commands
Options Menu Commands

View Menu Commands

Full View.
View the entire waveform.

Zoom In.
Zoom In to position defined by Start and End positions.

Zoom Out.
Zoom Out by 50%.

File Menu Commands
Edit Menu Commands
Tools Menu Commands
Effects Menu Commands
Special Menu Commands
Options Menu Commands

Tools Menu Commands

Wave Adder.
Add one of the selected waveforms to the selected section.
Select the frequency of the waveform, and the amount of the original section required.
If 0% of original selected then the new waveform will have amplitude 100%.
Waveforms available: **Sine - Triangular - Square - Sawtooth.**
Noise available: **Gaussian - Uniform - Pink.**
plus **DTMF tones,** pre-emphasis is the extra gain given to the High group, typically 1.5dB.

Absolute Value.
Rectify the section so that it is all positive.

Equalisation.
Produces a more uniform amplitude for the selected section.

Invert.
Invert the section.

Mute.
Zero the section.

Normalise.
Amplify section to maximum level without clipping.

Gain Adjust.
Amplify section by a defined amount. Default 70%<100% Reduce amplitude.
100% No change.
>100% Increase amplitude, be careful of clipping.

Remove DC Offset.
Remove any DC Offset which may have been introduced.
The offset amount is calculated by the computer, and removed from the entire file.

Pop Removal.
Attempts to remove any short discontinuities within the selected section, by interpolating across the discontinuity.

Noise Gate.
Removes any signal below the specified threshold, once the Hold condition is satisfied.
Threshold Amplitude at which Noise Gate is activated.
Hold length of time over which signal must remain below the Threshold.
Release length of time before Noise Gate is switched off.

Compressor.
Compress the Signal above the Threshold by the amount determined by Ratio, once the Attack condition is satisfied.
Threshold Point at which Compression activated.
Ratio Amount of compression to apply.
Knee: **Hard** Activate compressor at Threshold.
Soft Activate compressor 6dB below Threshold, but with less compression, to produce a softer knee.
Any Signal >Threshold will be compressed by the amount Ratio.
Attack length of time after hitting the Threshold before compressor activated.
Release length of time before compressor switched off.

Filters.
Low Pass, High Pass, Band Pass and Notch Filtering are available.
If passband ripple is 0dB then a Butterworth filter is used, otherwise a Chebyshev is implemented. The filters are designed with a maximum of 12 poles.

Graphic Equaliser.
The 10 band Graphic Equaliser lets you modify the characteristics of the sound, it works the same as a standard audio equaliser. This tool is suited to creating smooth tonal variations. If you need sharper cut-off then use the **Filters** tool.

Temporal Averaging.
A fast waveform smoothing technique, the selected section being smoothed over the number of samples selected. Can be useful for smoothing out noise and glitches.

Fade In.
Fade In the section of Track.

Fade Out.
Fade Out the section of Track.

Ramp.
Fade the Track between the Left and Right amplitudes.
Align Left: Left = 100% Right: Input amplitude.
Align Right: Right = 100% Left: Input amplitude.

Split Channels.
Split the Left and Right channels of a stereo Track in to separate mono files.
A single filename is required for the new files, _l and _r are appended to the respective filenames.

Edit Menu Commands
Effects Menu Commands
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Effects Menu Commands

Multiplier.
Multiply the selected section another file.
The file used as the multiplier must be 8bits, mono, and of the same sampling rate as the current waveform.

Amplitude Modulate.
Multiply the selected section by one of the waveforms selected.
Select the frequency of the waveform, and the amount of the original section required.
If 0% of original selected then the new multiplied waveform will have amplitude 100%, a selection of >0% but <100% will mix the original with the multiplied waveform.

Frequency Modulate.
Multiply the selected section by an FM signal.
Select the centre frequency of the waveform, the amount of the original section required, and the % Deviation of the centre frequency(Fc). If % Deviation set to 100%, the FM signal will swing from;
0.5CFc---> 1.5Fc or Fc(1 - Deviation/200) ----> Fc(1 + Deviation/200).
If 0% of original selected then the new multiplied waveform will have amplitude 100%, a selection of >0% but <100% will mix the original with the multiplied waveform.

Enhance.
Increase the harmonic content of the section.
This sonically tends to give most emphasis to the lower frequencies, where the additional harmonics occur in the 300-4000Hz region; ie. where our hearing is most sensitive.

Reverse.
Reverse the section.

Time Compress/Expand.
Compress or Expand the selected section.
Specify the desired length of the selected section, the section will be expanded or compressed to fit.

Delay.
Creates a Delay effect on the selected section.
Define the amount of delay required, and the Gain of the delayed section.
Similar to an echo, but without feedback.

Echo.
Creates a simple Echo effect on the selected section.
Define the amount of delay required, and how the Delayed section will Fade out.
Setting both Fade parameters the same will fade the Echo at a fixed rate.
Settings the Fade parameters differently allows the front and back of the delay to be Fading at different rates.

Flange.
Flanging is created by modulating the pitch of the selected section.
Select the offset delay, the amount of the original section included, the amount of flanging (% deviation), and the frequency of the flanging.
The % Deviation is the % of the Delay. If Delay = 100mS, % Deviation = 20%, then the Delay will be modulated +/- 10%, at the specified frequency rate. The flanging frequency is typically low <10Hz.
If 0% of original selected then the flanged waveform will have amplitude 100%, a selection of >0% but <100% will mix the original with the flanged waveform.

Phaser.

Phasing is an effect created by shifting the phase of the signal then adding it back to the original, causing some frequencies to be attenuated and others enhanced.
Depth is the amount of the Phased signal to be added to the original.
Feedback is the amount of Feedback to apply.
Phase Shift   is the amount to shift the signal by.

Reverb.
Reverberation is a complicated, computationally intensive means of adding ambience to the sound.
Reverberation Design

Vibro.
Multiply the selected section by one of the waveforms selected.
Select the frequency of the waveform, and the amount of the original section required.
If 0% of original selected then the new multiplied waveform will have amplitude 100%, a selection of >0% but <100% will mix the original with the multiplied waveform.
The Vibro effect is created by Amplitude Modulating the original, at a low frequency <30Hzwith a large amount of the original signal included >60%.

Cross Fade.
Cross Fade the Left/Right channels of a stereo Track. The length of the fade is determined by the Start and End positions.

Bouncing Cross Fade.
Bounce the Left/Right channels of a stereo Track. The length of the area is determined by the Start and End positions, with a selected number of Bounces performed within this area.

Edit Menu Commands
Tools Menu Commands
Special Menu Commands
Options Menu Commands

Special Menu Commands

Frequency Analysis.
Perform an FFT at the current position, the size of the transform being determined by the FFT Setup option. The windowing function applied is determined in the FFT window.

Spectral Averaging.
Average the selected number of FFT spectrum sequences.

Power Spectrum Analysis.
Compute the FFT and average the signal over the required bandwidth.

FFT Setup.
Select the size of the FFT to perform.

Noise Reduction.
Reduce the Noise within the selected section.
Noise Reduction

DTMF Identification.
Try to identify any DTMF tones in the selected section.

Two **experimental** functions for the conversion of Beat and Note information in to a midi file. Originally developed to detect drum beats (well, fingers on a table) for direct midi input. The routines work OK with simple, uncluttered sounds. You will need to clean the sound up as much as possible prior to using the routines.

Beat Extraction.
Try to identify the Beats within the selected section, the results are written to a Midi file.

Note Extraction.
Try to identify the Notes within the selected section, the results are written to a Midi file.

Edit Menu Commands
Tools Menu Commands
Effects Menu Commands
Options Menu Commands

Options Menu Commands

Open CD Player.
Activates the CD player if one is available.

Recording Setup.
Select the format of any recordings to be made.

Fine Time Increments.
Modify the Fine Increments setting of the Start and End Positions.

Zero Crossing Detection.
When selecting a section with the mouse, force the computer to ensure the section starts and ends at a zero crossing point if possible. If a section small enough that each sample is discernible is selected, then the ZCD is automatically disabled.

Edit Menu Commands
Tools Menu Commands
Effects Menu Commands
Special Menu Commands

Window Menu Commands

Tile.
Tile the windows across the screen.

Cascade.
Cascade the windows.

Arrange All.
Manually arrange the windows.

File Menu Commands
Edit Menu Commands
View Menu Commands
Tools Menu Commands
Effects Menu Commands
Special Menu Commands
Options Menu Commands

Reverb

The Reverb Setup Dialog.
Reverberation is a means of adding ambience to the sound.
This is created by adding multiple (up to 8) delayed copies of itself at varying amplitudes to create the Early Reflections, feedback and cycling of these Early Reflections is used to generate the Reverberation Tail. The resulting additive and subtractive nature of the delayed waveforms produces the added ambience.

This consists of a white Setup sheet, on which delay and gain **Taps** can be placed. The length of Setup sheet can be scaled from 320mS down to 10mS, with the arrow buttons. Above the red line indicates +ve gain, below -ve gain.
The panel below the Setup sheet contains two edit boxes for accurate **Delay** and **Gain** settings. An additional two edit boxes **Left** and **Right** are to adjust the entire Gain of all Delayed taps, so each channel can be adjusted individually; these Gains can only reduce or invert the gains of the original Taps.

A **Tap,** is delay and gain indicator, up to 9 taps can be on the sheet, 8 plus the original to which only a gain adjustment can be applied.

The Original Waveform is indicated by a Tap of zero delay (red), additional Taps are blue.

To add a Tap to the sheet, place the mouse close to the left of the sheet, left click the mouse and drag out a Tap to the desired location, the delay and gain being indicated in the edit boxes. Fine adjustments can be made of the currently selected Tap via the edit boxes.

Whenever you **left click** the nearest Tap will be picked up, if the nearest is at 0mS (which all cleared Taps are) then a new Tap is selected.

The act of adding a Tap, is placing the position of an Early Reflection.
The Tap with the greatest Delay controls the amount of feedback applied to generate the reverberation tail.
Temporal Averaging (it's faster than filtering) of the reverbed signal is performed, dependant on the position of the Reverb Tap.

Note: Sound travels at approximately 1 foot per millisecond.

The Buttons.

Open the specified reverb settings.
Save the current reverb settings.
Reset the reverb settings.
OK perform the reverberation as defined by the current settings.
Cancel Exit and do nothing.

Reverberation Design

Reverberation Design

Reverberation is a means of adding ambience to the sound.
This is created by adding multiple (up to 8) delayed copies of itself at varying amplitudes to create the Early Reflections, feedback and cycling of these Early Reflections is used to generate the Reverberation Tail. The resulting additive and subtractive nature of the delayed waveforms produces the added ambience.

How Do I create Reverb effects?.
With the correct placement of Taps.
The act of adding a Tap, is effectively placing the position of an Early Reflection.
The Tap with the greatest Delay controls the amount of feedback applied to generate the reverberation tail.

The Reverberation Tail?.
This is where things can get difficult, the Early Reflections also control the creation of the Tail.

The Tap with the greatest Delay controls the amount of feedback applied, and the start position of the reverberation tail.
Placing only One Tap will create an effect similar to the simple echo.

Look at Reverb Setup: Hall04.rvb then Hall05.rvb, notice how only the last tap (with greatest Delay) has moved.

With Hall04.rvb, the first 3 taps are the Early Reflections. After the time determined by the last tap, the Early Reflections are repeated and continue to repeat with decreasing gain, specified by the last tap's height. With these settings the Reflections and Echo repeatedly overlap to produce increasing numbers of reflections, and a complex reverberation.

Hall05.rvb, the first 3 taps are in the same place, but the last tap is well separated, the reverberation time is greatly increased. The Early Reflections will be cycled to produce an increasing numbers of reflections, but as the reverberation time is much longer, the repeated Reflections will tend to die away before the next echo is expected, thus producing more discrete, yet complex Echoes.

Note: When designing Reverb effects, do so with a short transient sound.

Reverb

Noise Reduction

The Noise Reduction algorithm implemented is designed to reduce Broadband noise with minimal reduction in signal quality. Broadband noise, is noise which is present at many frequencies, typically the same frequencies as the signal. This type of noise can not be removed by simple filtering alone; eg tape hiss, noise from fans, machines, sound cards etc.

The **Noise Reduction** is a two step process.
First a 'Noise print' that is representative of the noise to be removed must be taken, a one second sample will do.This noise region is analysed and its spectral make-up established.
Select **Special|Noise Reduction|Sample Noise** to take the 'Noise print'.

Reducing the Noise
Select the Section to be processed, then select **Special|Noise Reduction|Reduce Noise.**
A Noise Reduction dialog box will appear, three parameters can be modified:

Noise Reduction: Set how much to reduce the Noise by.

Threshold Gain: Set the threshold below which the Noise Reduction will be activated.
This adds an offset to the 'Noise print', anything below which will be treated as noise.

Overlap: Controls the precision of the Noise Reduction.

Notes.
Increased Overlap = Increased precision = Increased processing time.
The Noise Reduction itself is computationally very intensive and slow even on the fastest machines.

Hint: When making a recording, record a few seconds of silence before or after whatever you want to record and use this as the 'Noise print'. The Noise reduction also works best if the DC offset is removed first, the algorithm hasn't got infinite resolution either, so sub-sonic noise may still be present. It is recommended that you High Pass filter the signal with a cut-off of ~10Hz.