GoldWave® Manual

October 2013

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I. Introduction

GoldWave is a professional digital audio editor that plays, records, edits, processes, and converts audio on your computer. This section lists some of the features of GoldWave and outlines the notation and organization of the manual.

Features

GoldWave includes a complete set of audio processing features.

- An intuitive and customizable user interface makes editing easy.
- An independent Control window provides direct access to audio devices. It contains controls for playback, rewind & fast forward, recording, volume, balance, and speed. Real-time visuals display the sound during playback and recording.
- A multiple document interface (MDI) allows several files to be opened at one time, simplifying file-to-file editing.
- Huge files are edited efficiently using a advanced virtual editing system, with configurable **hard disk** or **RAM** based storage.
- Sounds are displayed graphically as a waveform and the level of detail can be changed by zooming in or out. The waveform can be reshaped directly with the mouse when zoomed in.
- Many audio effects, such as Dynamics, Echo, Flanger, Mechanize, Reverse, Pan, and Pitch enhance, distort, or alter sounds in various ways.
- Sophisticated filters such as the Noise Reduction, Spectrum Filter, and Pop/Click, filters help restore and remaster audio.
- The Batch Processing command converts a group of sound files to a different format and type and applies any number of effects.
- The CD Reader tool digitally copies audio from a CD to a file on your system.
- An Expression Evaluator generates everything from simple tones to complex filters. Expressions for telephone dial tones, waves, and effects are included.
- A File Merger tool joins together many files into one.
- A Speech Converter tool coverts spoken audio to text and can read text using speech synthesis.
- For maximum extensibility, several plug-in interfaces are supported for file formats, visuals, and effects.

How to Use This Manual

Familiarity with the Windows interface, such as property windows, tool bars, scroll bars, etc., is recommended before reading this manual.

For those who are unfamiliar with digital audio, Appendix A briefly introduces some of the fundamentals of computer audio. Appendix D gives a tutorial for recording audio from a turntable, removing noise, and splitting the file into tracks for CD-R burning. Appendix E contains troubleshooting information and answers to common questions.

Section II: Getting Started, covers system requirements and installation. Section III: Using GoldWave explains the interface and menu structure in detail. Topics are covered in the order that they appear in GoldWave's menu. Section IV: General Information, provides support, copyright, and warranty information.

Notation

Bold or link coloured text and a vertical bar are used to denote menu commands. **File | New**, for example, means to select the **New** command from the **File** menu. This notation is used to refer to other sections within this manual as well. In the above example, you can find information by looking for **New** under the **File Menu Commands** section. If the first word is **Start**, then select the command from the main Windows task bar menu structure instead.

Options and settings are given in a `fixed width` font or in quotes.

A information icon emphasizes helpful information and techniques.

An exclamation mark emphasizes warnings and other important information.

II. Getting Started

The following sections give instructions for installing and configuring GoldWave on your computer.

System Requirements

The minimum system requirements for GoldWave are:

- Pentium based PC or compatible
- Microsoft® Windows® 2000, XP, Vista, 7, or later
- 512 Megabytes of RAM (1 GB recommended)
- 100 Megabytes of hard disk space
- Mouse
- Sound hardware with a Windows compatible driver

If you need to edit large files, you will need a large amount of hard drive space. One minute of CD-quality sound requires 20 megabytes of storage. Editing a full CD requires at least 1.5GB of storage.

For editing audio in movie files and editing **mp3** files, you must have a recent version of Windows Media Player installed.

Installation

The following section gives instructions for installing GoldWave on your system. Before running GoldWave make sure that you have an appropriate Windows sound driver installed. If you need to add one, use the "Add New Hardware" item under

Start | Settings | Control Panel. The driver and installation instructions should be included with your sound card. The current settings for your sound card are listed under the "Sound, video, and game controllers" item of the Device Manager. The Device Manager is found under "My Computer" **Properties** or the **System** icon in the Control Panel.

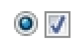
Installation From a Downloaded Program (Exe) File

If you downloaded the self-installing version of GoldWave, simply run the download. It prompts you to provide a destination folder where GoldWave will be installed. A desktop shortcut and Windows **Start** menu items are created automatically, if selected.

General Installation Instructions

Check the ReadMe.txt file for any additional information not available at the time this manual was created. New versions of GoldWave will be available from the web site: www.goldwave.com

Setting Audio Devices

To choose an audio devices to use for playback and recording, use the properties  button on GoldWave's Control window, then choose the Device tab. Select appropriate devices from the drop down lists of installed playback and recording devices. Use "Shared" quality unless you require using specific hardware sampling rates. Use the Test buttons to make sure you can hear playback and that recording is working.

Additional Settings

- Use **Options | Storage** to set a sound folder and temporary storage folder.
- Use **Options | Tool Bar** to customize tool bars.
- Use **Options | Colours** to change Sound window colours.

To associate a file type with GoldWave, such as **wav** or **mp3** files:

1. Open Windows Explorer
2. Browse to the folder where the file is located.
3. Right-click on the file.
4. Choose "Opens with" from the menu.
5. Choose "Choose default program..." from the submenu.
6. Choose the "Browse" button to find the GoldWave program, usually in "C:\Program Files\GoldWave" or "C:\Program Files (x86)\GoldWave".
7. Choose the "Open" button.
8. Check the "Always use the selected program to open this kind of file" box
9. Choose the "OK" button to close the "Open with" window.

See Windows help for more information about file type associations.

III. Using GoldWave

The following sections give information about GoldWave's user interface, features, and menu structure. The first few sections provide general overviews, while subsequent sections provide details on menu commands.

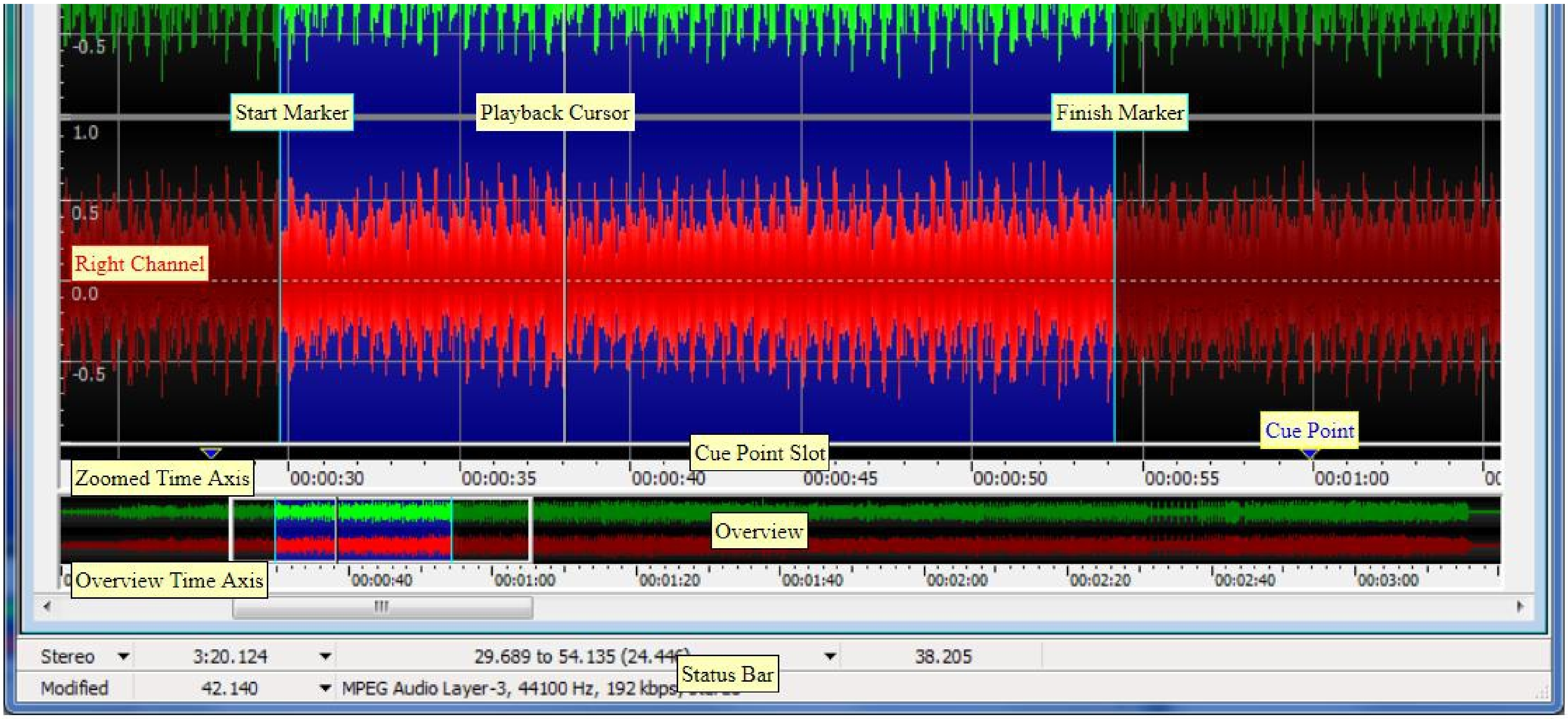
Interface Overview

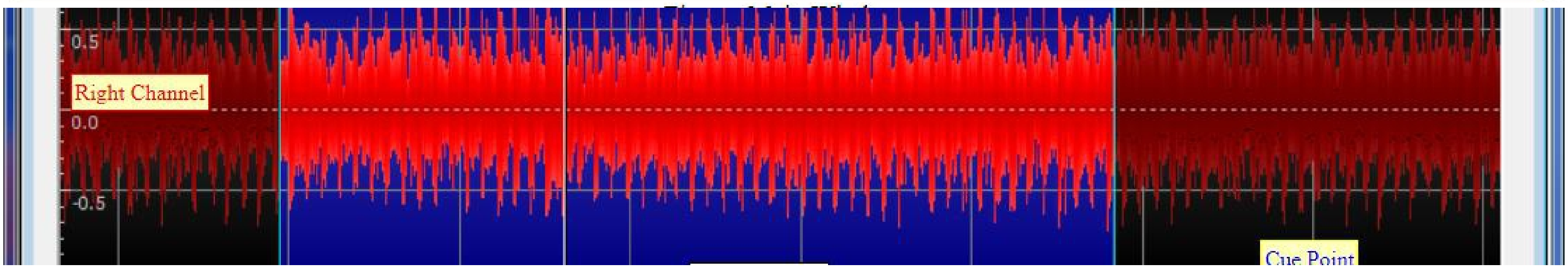
GoldWave is composed of three windows: the Main window, Sound windows, and the Control window.

Main Window

The Main window contains the main menu, two rows of tool bar buttons, and status bars (see Main Window figure below). It groups together and manages all the Sound windows.

Main Menu Main Tool Bar Effect Tool Bar Sound Window Left Channel Right Channel Start Marker Finish Marker Playback Cursor Cue Point Slot Cue Point Zoomed Time Axis Overview Overview Time Axis Status Bar
Figure: Main Window





Tool Bar

The tool bar buttons provide quick access to many of the frequently used commands. The upper bar holds File, Edit, View, and Tool commands, while the lower bar contains Effect commands. The function of each button is displayed in the lower status bar when the mouse pointer is positioned directly over it. Use the Options | Tool Bar command to configure the tool bar.

Status Bar

The status bars show attributes of the Sound window, including the sampling rate, length, selected region, channels, and general file format information. By clicking the mouse pointer over any status item that has a inverted triangle on its right side, the unit or format for that status item can be changed. If you click the mouse pointer over the length item, for example, a menu appears showing length in terms of storage size, time, and samples. If you click on the channels item, then you can select a single channel or both channels of a stereo file.

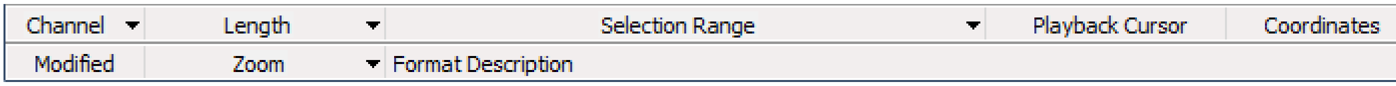


Figure: Status Bar Contents

Sound Windows

Sound windows are created when you open a file. These windows contain a waveform graph of the sound with a time axis near the bottom. For stereo sounds, two separate graphs are shown. The top green graph is the left channel and the bottom red graph is the right channel. The selected part of the sound is highlighted with a blue background between two cyan markers. A white, vertical line shows the current playback position within the sound. This line is the *playback cursor*.

A cue point slot is located just below the graph. Cue points are shown as inverted yellow and blue triangles. Overlapping cue points are shown in slightly different colours.

Near the bottom of the Sound window, a small Overview area shows the entire sound with the selected part in highlighted green and/or red with a blue background and the rest with a black background. A bevelled frame indicates what part of the sound is currently displayed and zoomed. Initially, the entire sound is selected.

Use the mouse buttons or the keyboard to change the selection. See Editing Overview for details. You can configure the window size and axes format of Sound windows using the Options | Window command. The Options | Colours command sets the colour scheme.

Control Window

The Control window interacts with your sound card. It contains buttons to play and record sounds as well as controls for volume, balance, and playback speed. Real-time visuals display audio data whenever a sound is played or recorded. See Control Overview for more details.

Progress Window

When performing time consuming processing, such as decoding a compressed file when opening it, or encoding a file when saving it, or using most effects, a progress window appears showing the amount of processing done and the estimated time remaining to complete it. Use the Cancel button to abort processing at any time. Use the priority drop-down list to reduce the load on the computer's processor to give more time to other programs. Use the notification drop-down list to set the audio notification played when processing is finished. Notifications are enabled only when processing takes more than 10 seconds.

Some poorly designed (or overclocked) computers overheat when performing complex processing, such as Noise Reduction and saving in MP3 or other compressed formats. Reducing the Priority setting helps to avoid thermal related errors.

Mouse Wheel

In GoldWave the mouse wheel supports zooming, scrolling and selection, or playback speed adjustments. Click the middle mouse button or the wheel button to display a menu to configure the behaviour of the mouse wheel. The mouse wheel works only when the Main window is active and only on the currently active Sound window.

Zoom In/Out

Zooms in and out of the waveform when the wheel is rotated up or down. The location of the mouse pointer is used as the focal point. Position the mouse over the area of interest when using the wheel. See View Menu Commands for information about viewing parts of the waveform in more detail.

Scroll and Select

When zoomed in, rotating the wheel up or down scrolls the waveform left or right. Holding the shift key moves the start marker. Holding both the shift and control keys moves the finish marker. Holding just the control key scrolls vertically, when zoomed in vertically. See Editing Overview for more information about selecting part of a file.

Playback Speed

Increases or decreases the playback speed by changing the Speed fader on the Control window.

Entering Times

GoldWave displays and accepts several different time formats. The time is separated into hours (H), minutes (M), seconds (S), and fractions of a second, like thousandths (T). Two digits are given for hours, minutes, and seconds. Zero or more digits are given for the fractional part. The basic format looks like this: `HH:MM:SS.TTTTT`. When using this format, minutes and seconds must be numbers from 0 to 59. Only five digits can be given after the decimal point. Other supported formats are given in the following table.

Table: Time Formats

Format	Description
`HH:MM:SS.TTTTT`	Hours, minutes, seconds, and fractions of a second. MM and SS must be between 0 to 59, inclusive. A decimal must be used to separate fractions of a second from seconds. Colons must be used to separate hours, minutes, and seconds. All values are optional. A time can be entered as `HH::` to specify hours only, or `MM:` to specify minutes only.

`MMMMM:SS.TTTTT`	Minutes, seconds, and fractions of a second. SS must be between 0 to 59, inclusive. Minutes can be larger than 59.
`SSSSS.TTTTT`	Seconds, and fractions of a second. Seconds can be larger than 59.
`HH:MM:SS.XX/YY`	Hours, minutes, seconds, and frames. This is the same as the first format, but instead of providing fractions of a second as a decimal, frames are used. The numerator, XX, specifies the frame number and the denominator, YY, specifies the frame rate, such as 30 for a 30fps animation, or 75 for CD frames. If HH, MM, and SS are not given, then XX may be greater than YY to specify any frame. Otherwise XX must be smaller than YY. Refer to the examples below.

Table: Time Examples

Example	**Meaning**
5	Five seconds
3:00	Three minutes
9:	Nine minutes
2::	Two hours
7.1/2	Seven and a half seconds
5123/60	Frame number five thousand one hundred twenty-three in a sixty frames per second file
34:25.15/75	The fifteenth CD aligned frame beyond thirty-four minutes and twenty-five seconds.
1::.3/4	One hour and three-quarters of a second.
12:34:56.789	Twelve hours, thirty-four minutes, fifty-six seconds, and seven hundred eighty-nine thousandths of a second.
.67	Sixty-seven hundredths of a second.

Control Overview

The Control window (see Control Window figure) is the interface to your audio hardware and drivers. On the bottom half of the window are visuals which display sound during playback and recording. On the top left area of the window is a standard set of audio controls, including play, stop, record, rewind, pause, and fast forward. A status visual is located just below these controls. In the top right area of the window are controls to set the playback device's volume, balance, and speed. A level visual is located just below these.

The Control window can be resized to change the size of the visuals or to hide them. Use the Window Menu Commands to rearrange the controls horizontally or vertically.

Play 1 Play 2 Play 3 Playback Controls Record Controls Properties Status Visual Level Visual Left Channel Visual Right Channel Visual
Figure: Control Window

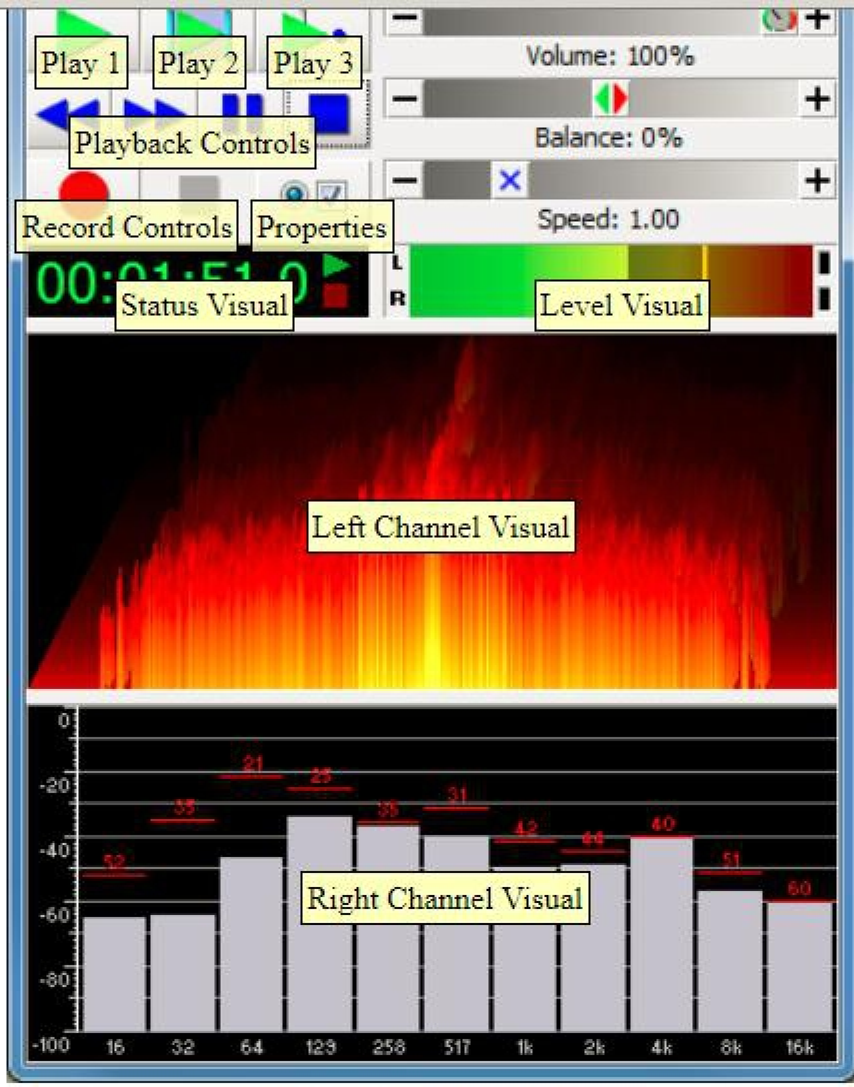


Figure: Control Window

Properties

The Properties button presents the Control Properties window containing several tabs to configure playback, recording, volumes, visuals, and devices. These options are described in the following sections. After installing GoldWave, you should take a moment to see if the correct playback and recording devices are selected under the

Device tab and familiarize yourself with the settings under the Play (see figure) and Record tabs.

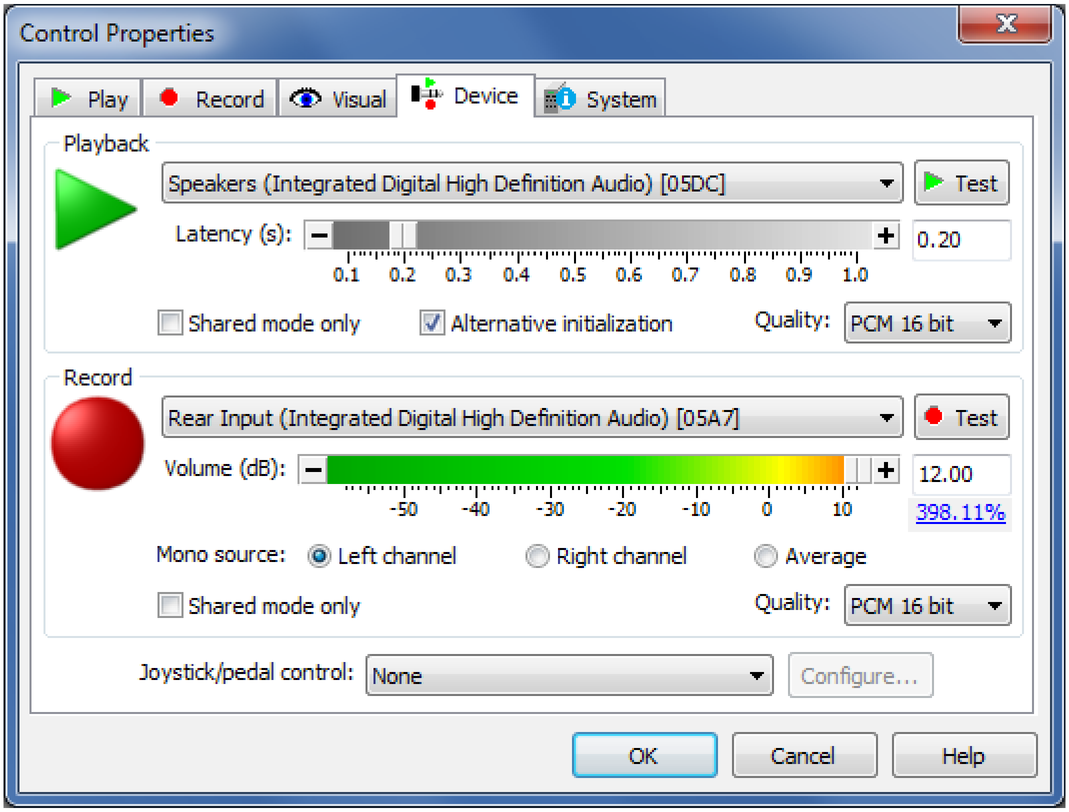
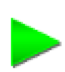
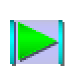
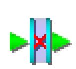


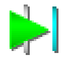
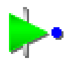
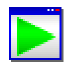
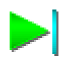
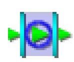
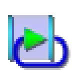
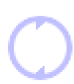
Figure: Control Properties

Play Properties

The Play Properties page contains options to configure the three play buttons and set the speed for rewind and fast forward. Play button options are given in the table.

Table: Play Button Options

Option	Image	Button function
All		Plays entire sound.
Selection		Plays region between start and finish markers (the selected part of the file).
Unselected		Plays regions outside the start and finish markers. This lets you quickly test how a cut or delete will sound without actually changing the sound. When possible, playback is confined to the region shown in the Sound window view so that the entire sound does not have to be played.

Cursor		Continues playback from the current cursor's position and stops at the finish marker. If the cursor is past the finish marker, playback begins at the start marker.
Cursor to end		Continues playback from the current cursor's position and stops at the end of the file.
View		Plays all of the sound currently shown in the Sound window view. This is useful if you zoomed in on part of the sound.
View to end		Starts playback at the left side of the sound currently shown in the Sound window view and continues playback to the end of the sound.
Finish		Plays three seconds just before the finish marker, so you can determine if that marker is in the right place without listening to the entire selection.
Intro/loop/end		This is a special playback feature that plays the sound in three sections. The beginning of the sound, outside the selection, is played first. Then the selection is played and looped. Finally the end of the sound, outside the selection, is played. Be sure to check the `Loop` box below to enable looping. This is useful for musical accompaniment, looped instrument samples, or testing loop points.
Loop point		Starts playback just before the finish marker and loops back to the start marker and continues playing briefly. Use this setting to test the current selection markers for smooth loop points.
Loop		If checked, it specifies the number of times playback should be repeated. A value of 1 loops playback once, so the selection is played twice. A zero value loops forever. The loop image appears over the image of a button that has looping selected.

Fast/Rewind Speed
The playback speed of the fast forward and rewind buttons is controlled by these values.

A value of 1.00 is normal speed. Entering a value of 3.00 for `Rewind speed`, for example, means the rewind button will play the sound backwards three times faster than normal. By entering small numbers (such as 0.1) the rewind and fast forward buttons will play very slowly. This is useful for finding pops or clicks, since the visuals will move slowly through the data.

Marker preview
Marker preview (scrubbing) plays a very short section of audio whenever an edit marker is moved with the keyboard arrow keys. The value specifies the amount of time to play just after the start marker or just before the finish marker. Set the time to 0 to disable previewing.

Record Properties

The Record Properties page includes features to monitor the recording sources, start recording automatically when a sound is detected, or delay recording for a certain length of time. The basic options are given in the table.

Table: Record Options

Option	Purpose
Use new file duration	Sets the duration for recording a new sound. See Entering Times. If this box is unchecked, then a window appears when you choose the Record New  button so that the duration may be entered. If the box is checked, the duration given here is used for all new recordings and no window appears.
Monitor input on visuals	Connects the recording source to the visuals so you can adjust volume levels before recording. Monitoring works only when the current sounds's sampling rate is compatible with the recording device or no sounds are opened. See Recording Sounds for information about selecting a different recording source and setting volumes.
Ctrl key safety	Prevents you from accidentally recording over a sound. To record, you must hold down the Ctrl key, otherwise a safety message appears.
Set finish marker at stop	Automatically moves the finish marker to the place where recording stopped. This makes it easier to trim the file after recording.

Show settings window	Displays an information window whenever recording is started. The window gives the current recording setup, including the recording device, the selected inputs, and other settings. Click on a label link to change the setup. Recording may be stopped when the setup is changed.
Filter dc offset	Automatically filters a dc offset from the recorded audio. Use this option if you see a lot of activity on the low frequency bars and VU meters even when recording silence.
Auto save	Automatically saves the file when recording ends. Bounded recording mode must be selected. If recording is manually stopped, the file is not automatically saved. Use this option with the Timer option to save a recording after a scheduled event. If you start recording in a new, untitled file, you will be prompted to provide a filename so that it can be saved under that name automatically. The default save format is used for the file type and attributes. If you start recording in an existing file, the original file will be overwritten when recording ends and recording **cannot** be undone.
Power down system	Automatically turns off the computer after saving the recording. Use this option with the Auto save and Timer options to shutdown the computer after a scheduled recording. This option is available on Windows 2000/XP or later.
Bounded to selection	Records within the selection only. Recording stops automatically at the end of the selection. If you stop recording before the end is reached, the rest of the selection is replaced with silence. Use this option to record for a fixed length of time.
Bounded and looped	The is similar to the above option, but recording restarts automatically when the end is reached and continues to record over and over until the stop button is pressed. This is

	useful if you are trying to capture a sound but do not know when it might occur. By loop recording a 1 minute sound, you will always have the last minute of audio stored for recall.
Unbounded	Recording starts at the start marker's position and continues recording until all storage is exhausted or until you press the record stop button. The file size is increased automatically to hold the new audio. This is useful if you do not know how long the recording will be.

Delayed Recording

The `Timer` option delays recording until the specified time and day of the week. Use this feature to automatically record something at a later time. The time is given in 24 hour time. A time of 06:00:00 is 6:00 AM and a time of 18:00:00 is 6:00 PM. 00:30:00 is 12:30 AM or 30 minutes past midnight. When entering the time, remember to include the seconds. Entering 18:00 means 00:18:00. You **must** press the record button to activate the timer.

Example

Record for 35 minutes on Tuesday at 7:00 PM:

1. Choose **Control Properties** from the **Options** menu.
2. Select the **Device** tab, then select the recording device you want to record.
3. If you are using Windows 2000 or XP or DirectSound mode, select the **Volume** tab, then select the input you want to record.
4. Adjust the recording volume as needed.
5. Choose the **Record** tab.
6. Check the **Timer** box.
7. Enter **19:00:00** in the Time box.
8. Select **Tuesday** from the Day drop down list.
9. Make sure the **Level activated** box is not checked.
10. Make sure the **Bounded to selection** Record mode is selected.
11. Choose OK.
12. Choose **New** from the **File** menu.
13. Select the quality settings you want.
14. Enter **35:00** in the Initial file length box.
15. Choose OK.
16. Press the record button to start the timer. The elapsed time should start counting down or show 99:99:99.9.

Remember to press the record button to activate delayed recording (timer or level activated).

Turn off any power management settings that may power down the computer.

`Level activated` recording is useful for automatically synchronizing recording to a sound source or efficiently capturing airport or police radio communications containing mostly silence that does not need to be recorded. It automatically starts recording when the sound source is above a given level and pauses recording when the sound is below the level. The `Threshold` specifies how loud a sound should be before recording begins. The value must be high enough so that noise does not trigger recording and low enough so that other sounds will. Start with a value around -20dB or record some background silence for several minutes to get a baseline and use Maximize Volume(Normalize) effect to get the peak level and set the threshold value above that. Be sure to keep the device recording volume the same. Any changes to that volume will affect the threshold (other Windows program may change the recording volome). The `Minimum duration` time specifies how long to record after the sound becomes quiet again. Using a value of 3 allows recording to continue for three seconds after the sound goes below the specified threshold. To minimize silence, use a value of 1 second or less, but not zero. A zero value causes recording to continue without stopping once triggered. The `Prebuffer` time specifies the amount of audio to store prior to activation. When activation occurs, the prebuffer audio is inserted before the currently recorded audio, allowing you to hear the sound slightly before activation. The `Time stamp cues` option provides a way of marking the date, time, and position of each activation. Cue points with the current date and time are created and can be view under the Cue Points tool. Use the edit box to specify the format for the date (this is done using the C strftime function). Some format specifiers are given below.

Specifier	Purpose
%a	Short weekday (Sun, Mon, ...)
%A	Weekday (Sunday, Monday, ...)
%b	Short month (Jan, Feb, ...)
%B	Month (January, February, ...)
%d	Day of the month (01 to 31)
%H	Hour in 24-hour clock (00 to 23)
%I	Hour in 12-hour clock (01 to 12)
%m	Numerical month (01 to 12)
%M	Minutes (00 to 59)
%p	"AM" or "PM"
%S	Seconds (00 to 59)
%y	2 digit year (00 to 99)
%Y	Year, all digits
%Z	Time zone name

Example	Cue name generated
%d %b %y, %H:%M:%S	12 Jan 05, 14:23:56

Date: %A, %B %d, %Y. Time: %I:%M:%S%p	Date: Wednesday, January 12, 2005. Time: 02:23:56PM
Today at %I:%M%p	Today at 02:23PM
%Y-%m-%d at %H=%M=%S (Easy to sort and safe for filenames)	2005-01-12 at 14=23=56

Volume Properties

The Volume Properties page is available in Windows 2000/XP or when using DirectSound mode. It lets you adjust recording volumes and select or unselect recording sources. Be sure to select the correct volume device that corresponds with the recording device selected under the Device tab.

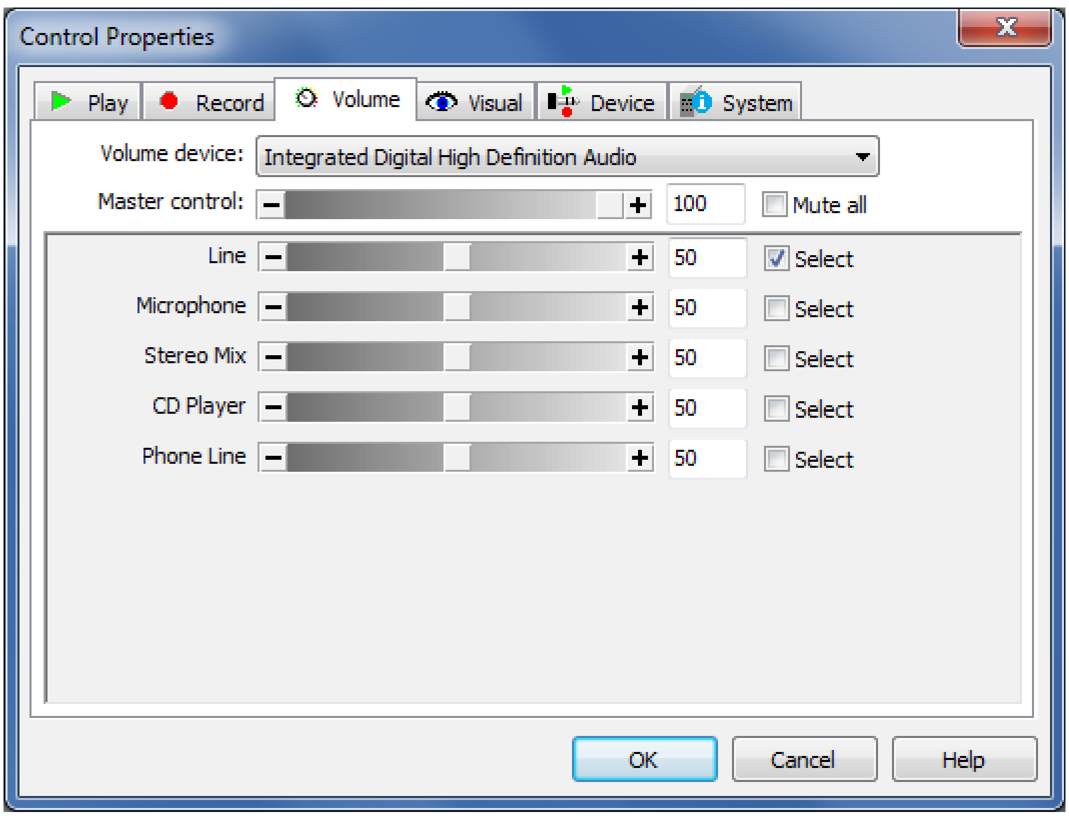


Figure: Volume Properties

A volume fader, edit box, and checkbox is shown for each source. To select a source, check the appropriate checkbox. If your sound card supports a master control, make sure that the `Mute all` option is turned off and that the master volume is not zero.

You can use the `Monitor input on visuals` option under the Record tab to activate the visuals without recording.

Note that volumes are changed regardless of whether you choose OK or Cancel to close the Properties window.

To select a different recording device, use the Device tab.

Visual Properties

The Visual Properties page controls what visuals are used in what location. There are four visuals: status, level, left, and right. The status visual is located at the upper left side. It displays elapsed time and playback and recording status. The level visual is located at the upper right side. It graphically displays the current output or input level. The left and right visuals display audio in a variety of ways, as listed in the table.

Table: Description of Visuals

Visual	**Description**
3D Bars	3 dimensional logarithmic frequency 10 band bar graph.
Analog Meter	Scaled amplitude needle meter.
Bars	Logarithmic frequency 11 band bar graph, commonly found on stereo systems.
Blank	Disables the visual and may improve performance on slower systems.
Blowing Inferno	Fire coloured, double-sided spectrum graph.
Bulge	Symmetrical, colourful frequency graph.
Envalope	Amplitude envelope.
Spectrogram	Coloured frequency spectrum, with time on the x-axis, frequency on the y-axis and colour as the magnitude. The colours, in increasing magnitude, are black, purple, blue, cyan, green, yellow, red, and white. A cyan point, for example, is higher magnitude than a blue point.
Spectrum	Frequency analysis of the sound.
Spinning Logo	Rotating GoldWave logo that tilts in response to the audio.
VU Meter	Peak and current amplitude level meter.
Waterfall	Flowing, coloured spectrogram.
Waveform	Standard amplitude waveform, much like the 1:1 zoom level in a Sound window.
X-Y Graph	The sound is plotted with the left channel against the right channel to generate Lissajous

patterns. This is often used to see the phase difference between two equal frequency signals. If the left and right channels are in phase, the pattern is a diagonal line running from the lower left to the upper right. If the channels are 90 degrees out of phase, the pattern is a circle. For general stereo sounds, it looks like a crazy scribble. The larger the scribble, the larger the difference between the channels. Monaural sounds always show a diagonal line since the left and right data are the same.

Some visuals have properties you can set, such as axes ranges, colours, display modes, etc. Right-click on a visual and select `Properties` from the popup menu to see these properties. You can resize the Control window to make the visuals larger or smaller.

Quick Select Menu
Use the quick select menu list to select your favourite visuals. The selected visuals appear in the popup menu when you right-click on a visual in the Control window.

Frame Rate
The frame rate sets the number of times per second that visuals are updated and drawn. A value of 60 or less gives good results, but you may want to use higher values to get an extra detailed spectrogram. The actual frame rate is limited by your system's speed. Use a lower frame rate for older systems.

FFT Window
When calculating any frequency related graphs, a window function must be used to smooth out analysis at the endpoints. The Kaiser 7 or Hamming windows are usually the best, but you can try the other windows just to compare the results.

Device Properties

The Device Properties page contains settings for playback and recording devices.

Playback and Record areas show the currently selected playback and recording devices. If more than one device is installed, you can select a different device from the drop down list. You can change playback and recording quality by selecting different bit depth from the `Quality` lists. Use `PCM 16 bit` quality unless your sound card supports higher bit depths. GoldWave takes exclusive control of the audio device unless you select `Shared` quality. The `Shared` option forces GoldWave to share the audio device with other programs using the sampling rate, channels, and resolution determined by the system. The system attributes are shown next to the "Quality" label. Use the `Configure` button on the System tab to change the system properties of an audio device. Shared quality is not recommended for recording because it does not allow recording at the sampling rate of the file, forcing you to create a new file for recording at the system defined rate.

Use the `Test` buttons to perform a simple test of a device. When testing a playback device, you should hear audio on the speakers or headphones. If not, check the Windows volume settings or select a different playback device. When testing a recording device, you can determine the supported sampling rate and quality and adjust the volume level.

The Playback area has additional settings for latency and initialization. `Latency` controls the amount of audio stored before sending it to the device. Using a higher value may eliminate gaps and stutters on a slow system, but it increases the delay between changing effect settings and hearing those changes during previewing. Using lower values makes effect previewing more responsive, but may cause gaps and stutters if the system is too slow to process all the audio or emulated drivers are used. This setting does not apply to recording. `Alternative initialization` solves problems with certain drivers and plug-ins. Use this option if GoldWave freezes when previewing an effect plug-in or if playback does not start properly in general. It begins playback on a separate processor without pausing or holding up the main program. This makes the interface seem more responsive when starting playback.

The Record area has a volume fader and mono options. The `Volume` fader (not available for Windows XP or DirectSound mode, use Volume tab instead) adjusts the volume level for the device. You can adjust the volume anytime during recording. The `Mono source` setting determines the input channel used when recording a mono file: `Left channel` captures audio on the left channel only; `Right channel` captures audio on the right channel only; `Average` captures audio on both channels and averages it into a single channel.

`Joystick/pedal control` allows playback and recording to be controlled using a game controller or a foot pedal. The first controller detected is used. The following table lists the modes of operation.

Table: Joystick Control

Mode	Description
None	Joystick control is disabled.
Foot pedal or buttons	A foot pedal controls playback. If the pedal has more than one button, they can be assigned to rewind, fast forward, etc. by using the `Configure` button. GoldWave supports most USB HID devices with simple button inputs, such as the VEC Infinity IN-USB-2 foot controller and most Delcom USB HID "joystick" or "programmable" foot switches.
Game controller	The main directional pad controls playback. Left is rewind, right is fast forward, down is

pause, and up unpauses. The first button (button 1 or A) starts or stops playback. The second button (button 2 or B) starts or stops recording.

Use the `Configure` button to configure foot pedal controls. Different brands of foot pedals use different switch combinations. After choosing the `Configure` button, choose an action button to assign, then hold down the pedal for at least one second.

System Properties

Use the System Properties page to change the audio interface used for playback and recording and list information about the system. This information can help locate problems with drivers, hardware, or the current setup.

GoldWave supports DirectSound and Core Audio/WASAPI. DirectSound is used for Windows XP and older versions of Windows. For Windows Vista and later, Core Audio/WASAPI is selected by default, but DirectSound may be used instead by clicking the button. DirectSound was deprecated in Vista and no longer has direct access to the audio hardware, so it is not recommended. Using WASAPI in exclusive mode is the only way to play or record audio directly through the sound hardware in recent versions of Windows.

Choose the `Configure` button to display the system's audio configuration window.

Choose the `Information` button to gather information about installed playback and recording devices.

If a device is listed as disabled, disconnected, or not present:

- Make sure all external connections are firmly in place and connected to the correct socket.
- Make sure the USB device is plugged into a working USB port and is powered on.
- Some devices, such as "Stereo Mix" or "What You Hear" are disabled in Windows by default and have to be enabled manually. Choose the `Configure` button, select the Recording tab, right-click anywhere in the list and select **Show Disabled Devices**, then right-click on the device in the list and choose **Enable**.

Playing Sounds

After opening a sound (see File | Open), use one of the play buttons, such as Play All , to play it. Each button starts playback at a different place, which can be configured under the Play tab of the Control Properties window. Right-click on one of the play buttons to quickly change the settings.

To start playback at any point in the sound, click on the time line under the waveform in the Sound window or right-click on the waveform and choose the **Play From Here** command from the popup menu. You can right-click-and-drag to select a part of the sound to play as well.

While a sound is playing, it is displayed on the visuals. The current position is displayed in the Sound window as a white, vertical line on the waveform (playback cursor). You can move the start and finish selection markers to the playback position by using the bracket keys, **[** and **]** or Edit | Marker | Drop... commands. See Editing Overview for more information about changing the selection. You can set cue points by using the Ctrl+Q key or the Edit | Cue Point | Drop Cue command.

If you do not hear anything during playback, check the following:

- Make sure the speakers are turned on.
- Make sure the speakers or headphones are connected to the correct socket (green).
- Make sure the Windows volume is not muted (click the speaker icon in the right corner of Windows Start bar).
- Make sure you see activity on the level meters and visuals in GoldWave. If not and the elapsed time is counting up, the sound contains silence.
- Try selecting a different playback device.

Pausing Playback

While a sound is playing, pause it with the pause button. Remember to use either play or stop later. Pause freezes the visuals and the current position marker so you can see the shape of the sound in the visuals or move the selection markers.

Stopping Playback

Playback can be stopped immediately with the stop button. Note that recording is stopped using a different button.

Rewinding and Fast Forwarding

Use the rewind button or fast forward button to quickly move back and forward through the sound. The current position is displayed in the Sound window as a white, vertical line on the waveform. You can adjust the speed of rewind and fast forward with the Play tab of the Control Properties window, as described previously. When one of these buttons is used to start playback, the region played is determined by the Play 3 setting.

Recording Sounds

Most computers have more than one recording input, such as microphone or line-in. To select and adjust a recording input, use the Device tab (and the Volume tab for Windows 2000/XP or DirectSound mode) of the Control Properties window. Make all connections before running GoldWave. Otherwise some devices or sources may not be listed. To adjust the volume before recording, use the Monitor input on visuals option under the Record tab of the Control Properties window.

Use the record new button to record your own sounds. Recording stops automatically when the duration has passed. If you stop recording earlier, the new file is trimmed to the length of the recording. Use the Record tab of the Control Properties window to set the default new file duration.

Use the record selection button to record into an existing sound. Audio is recorded into the selection of the Sound window replacing any audio that was previously there. Recording stops automatically when the end of the selection is reached (bounded mode) or when no more storage is available (unbounded mode). You can stop recording at any time with the recording stop button and the unrecorded part of the selection is filled with silence.

You can make room for recording in the current sound by using the Edit | Insert Silence command.

A recording pause buttons appears in place of the record button so that you can pause and unpause recording.

Many recording options are available in the Record tab of the Control Properties window. Right-click on the record button to quickly access some of these options.

Remember to press the playback button on the cassette player, record player, or CD player when recording from an external device. See the Appendix D for a tutorial.

If you want to record vocals over existing music, you'll need to use two files in GoldWave. You can record in one file while playing the other. Some sound cards have a "Stereo Mix" (or "What U Hear" or similar) recording source you can select that will allow you to record both the microphone input and the playing sound at the same time. If that source is not available, you'll need to mix the two files after recording instead.

For long recordings, turn off any power management settings that may power down the computer.

Volume and Balance Faders

Use the top volume fader to change the playback volume. Move the fader right or click the plus button to increase the volume. Move it left to decrease the volume. The current volume is shown numerically in a popup tip window to the left of the fader. A value of 100% is full volume.

Use the middle balance fader to change the left/right balance. Move the fader in the direction you want to shift the balance. Right-click on the fader to display a popup menu to quickly set the balance left, right, or center.

Note that these faders **do not** change the recording volume. See Recording Sounds for more information.

Speed Fader

The bottom speed fader changes the playback speed of the audio device. Move the fader right to increase the speed or left to decrease it. The relative speed is shown numerically to the left of the fader in a popup tip window. Right-click on the fader to display a popup menu to quickly set the speed to commonly used ratios. Note that changing the speed also changes the pitch like spinning a vinyl record faster or slower. To change the speed of the file, use the effect instead.

Editing Overview

Selecting Part of a Sound

Almost all commands in GoldWave operate on the currently selected part of a sound. The selected part, or *selection*, is the highlighted part of the sound graph between two vertical markers (see Main Window figure). The vertical markers are cyan lines located to the left side (*start* marker) and right side (*finish* marker) of the view.

GoldWave provides several ways of setting the selection. You can:

- Use the standard click-and-drag method used in most other Windows programs.
- Click the right mouse button to display a menu where you can choose Set Start Marker or Set Finish Marker.
- Click-and-drag the left mouse button over one of the cyan markers to drag it (useful for precise adjustments of the end points).
- Click-and-drag using the right mouse button, then choose Select from the menu that appears.
- Use the Edit | Marker | Set command.
- Use the Edit | Channel submenu to select one channel of a stereo file.
- Use the keyboard. See Accessibility Overview for details.

If you just click the left mouse button without dragging, the start marker is moved. If you just click the right mouse button, a context menu appears, which can be used to start playback at any position. If you click-and-drag with the right mouse button, you can play or zoom in on that area without altering the current selection.

Additional notes and techniques:

- To use the old style of selecting part of a sound (in v4 and earlier), enable the setting under the Options | Window command. Note that none of the new selection features work if you enable that setting.
- You cannot place the finish marker before the start marker. The same is true for setting the start marker after the finish marker.
- Mouse selection methods work in both the large sound graph or in the small Overview graph.
- The Edit | Marker | Snap to zero-crossing feature helps to minimize pops and clicks between edit points by finding a point where the waveform is close to zero amplitude.
- To select a certain part of the sound while playing it by using the **[** and **]** (bracket) keys or the **Drop** commands under the Edit | Marker submenu. You can use the rewind and fast forward buttons to quickly find a sound.
- In most cases, editing and effects are performed only on the selection. Some effects, such as the Resample, and Playback Rate alter the entire sound.

Redrawing the Waveform with the Mouse

You can redraw the waveform with the mouse to remove pops/clicks or other small defects. To do this, you must first zoom in so that individual samples are visible (see View | Zoom 1:1 or View | Zoom 10:1).

1. Zoom in 1:1 or closer.
2. Place the mouse arrow directly over the waveform. The arrow will change into a target crosshair.
3. Click and hold the left mouse button.
4. Move the mouse to redraw the waveform.
5. Release the mouse button to finish the changes.

Cut & Paste, Mixing, and Crossfading

Cutting and pasting audio in GoldWave works much the same way as cutting and pasting text in a word processor. Mixing and cross-fading involves combining two or more sound together so that they play at the same time.

Cut & Paste

The Edit | Cut command removes sections of audio. The Edit | Paste command inserts sections of audio from the clipboard. Before you can paste, you need to use Edit | Cut or Edit | Copy to place some audio into the clipboard.

To join several files together:

1. Open the first song.
2. Open the second song.
3. Choose **Copy** from the **Edit** menu.
4. Close the second song.
5. Choose **End** from the **Edit | Paste At** submenu.
6. Open the third song.
7. Choose **Copy** from the **Edit** menu.
8. Close the third song.
9. Choose **End** from the **Edit | Paste At** submenu.
10. Repeat steps 6 to 9 for each song you want to join.

If you want to split a large file into smaller section, use the Cue Points tool, which has a Split File button.

Mixing

The Edit | Mix command mixes one sound with another so they both play at the same time.

To add vocals to music:

1. Open the sound containing the vocals.
2. Choose **Copy** from the **Edit** menu.
3. Open the sound containing the music.
4. Choose **Mix** from the **Edit** menu.
5. Enter a volume to mix the vocals (0dB = full volume).
6. Adjust the mix time to align the vocals with the music, using the preview button as needed.
7. Choose OK to process the mix.

When mixing more than a couple of sounds, you should reduce the mixing volume and the destination volume to prevent clipping distortion. The volume of the destination sound can be reduced before mixing by using the Effect | Volume | Change Volume command.

Crossfading

A crossfade occurs when one sound fades out while another sound fades in. Radio stations often use crossfades to fade out the end of one song while fading in the next song so there is no break in the music. GoldWave's Edit | Crossfade command does the same thing by using the clipboard audio as the second song. The entire song to fade in must be copied to the clipboard before using the command.

To do a crossfade in GoldWave, follow these steps:

1. Open the first song (the one that will fade out at the end).
2. Open the second song (the one that will fade in at the beginning).
3. Choose **Copy** from the **Edit** menu.
4. Close the second song (or activate the first song window).
5. Choose **Crossfade** from the **Edit** menu.
6. Set **Duration** to 5.00 second (or whatever you prefer).
7. Select the **End of selection**, and **Linear** fade curves settings.
8. Choose OK to process the mix.

In some cases more control is needed. To do a crossfade manually in GoldWave, follow these steps:

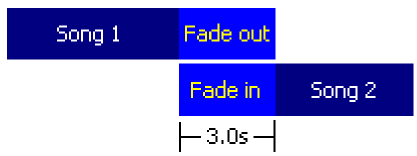


Figure: Crossfade

1. Open the first song.
2. Move the start marker to select the last 3 seconds of the song.
3. Choose **Fade out** from the **Effects | Volume** sub menu and use a −160dB final volume.
4. Open the second song.
5. Move the finish marker to select the first 3 seconds of the song.
6. Choose **Fade in** from the **Effects | Volume** sub menu and use an initial volume of −160dB.
7. Choose **Select all** from the **Edit** menu.
8. Choose **Copy** from the **Edit** menu.
9. Click on the first song to activate that Sound window.
10. Choose **Mix** from the **Edit** menu and use a volume of 0dB.
11. Choose OK to process the mix.

For extra control, use the Effect | Volume | Shape Volume create custom fades before mixing.

Storage Overview

GoldWave supports both hard drive based editing and RAM based editing. These features are described below. Hard drive storage is enabled by default. Use the Options | Storage to configure the storage mode. For uncompressed files, GoldWave will read the audio directly from the original file. It does not copy a file to temporary storage until it is edited or modified. The original file is not changed until it is saved. For most compressed files, the data has to be decompressed to temporary storage when the file is opened.

Working with compressed files may take much more storage than expected. MP3 files, for example, have to be decompressed into temporary storage before GoldWave can edit them. Such files may require over 20 times the amount of compressed storage when opened. A 10MB MP3 file could require over 200MB of storage space.

Hard Drive

In hard drive based editing, the entire sound is stored in a temporary file on your hard drive where it can be modified. This allows you to edit huge files provided the required drive space is available. Only a small amount of RAM is required for each opened sound. The drawback is that editing and effects processing take more time since audio data must be transferred to and from the drive.

RAM

In RAM based editing, the entire sound is stored in your computer's memory. This allows you to edit and process files very quickly. It saves time and reduces the load on your hard drive. The drawback is that the size of the files must be small enough to fit in the available RAM. If you edit or record large files, Windows will start swapping memory to the hard drive, which significantly degrades performance and may cause defects when recording. Also note that in the event of a system crash, it will not be possible to recover a file stored in RAM. Editing files larger than 1GB in RAM is not recommended due to limitations in Windows.

File Overview

This section explains file formats and gives general information about how files are handled by GoldWave. Several features for storing and handling files can be configured using Options | Storage and Options | File Formats.

File Format

Sound files come in a variety of forms. Usually, the form or *type* of sound can be determined from its filename extension, such as `.wav` or `.mp3`. GoldWave supports all the sound types listed in the Supported File Types table, and more depending on what file format plug-ins are installed. Each file type can have several sub-formats or *attributes*. The `.wav` type for example, can hold audio encoded or compressed in dozens of different ways, including PCM, ADPCM, companded, or MPEG1 Layer 3.

Table: Supported File Types

Extension	Comments	Cues	Info	>4GB
.aiff .aifc .afc	Apple / Macintosh sound files. Compressed files are not supported. Cue points are supported. File text information is supported in ANSI only. `NAME`, `COPY`, `ANNO`, `AUTH`, `genr`,	✓	✓ ANSI	✗

	`©url`, `©trk`, `©day`, and `©art` are preserved.			
.ac3	AC3 compressed files. GoldWave does not support these files directly, but it may open them automatically using system decoders. If not, installing the AC3Filter for DirectShow (search Google) may work.	✗	✗	✗
.ape	Monkey's Audio compressed files. Requires the APEFile plug-in.	✓	– ANSI ID3v1	✗
.asf .avi	Microsoft audio and/or video files. GoldWave can extract the audio portions of these files, but cannot save or create them. See .wma and .wmv below.	✗	– .asf only	✓
.au	Sun or NeXT files, commonly used on web pages and in Java. Supports 8 & 16 bit linear, mu-law and A-law encoded files. No support for file text information.	✗	✗	✗
.dwd	DiamondWare sound files. GoldWave supports 8 & 16 bit PCM attributes through a plug-in. No support for file text information. An external plug-in for this format is included in the plug-in SDK.	✗	✗	✗
.flac	FLAC sound files. This is a lossless compressed format. GoldWave supports 8, 16, and 24 bit attributes with low (fast), medium, and high (slow) modes of compression.	✓	✓ UTF8	✓
.iff	Amiga 8SVX files. `NAME`, `COPY`, `ANNO`, `AUTH`, and `CHAN` are all preserved. Limited support for file text information.	✗	– some	✗
.mat	Matlab files. The data must be normalized (i.e. -1.0 to 1.0) for	✗	✗	✗

	double precision data. If the "wavedata" variable is two dimensional, the data is assumed to be stereo. GoldWave saves audio data in the "wavedata" variable and the rate in the "samplingrate" variable. A 11025Hz sampling rate is assumed if none is present. No support for file text information.			
.mov	QuickTime movie files. GoldWave may use system decoders or QuickTime decoders to extract the audio portion from the file (if present). See the .mp4 and .m4a types for more information. Files cannot be saved in this format.	✗	✓	✗
.mp3	MPEG1 Layer 3 compressed sound files. To read these files, you must have an MPEG decoder installed (usually included with Windows). To save a file in this format, you must have the LAME encoder installed. See the GoldWave website for details. File text information is supported in ID3v2 tags using unicode (UTF-16). The ID3v1 tag is read, but not written.	✗	✓	✓
.m4p	Copy protected/encrypted iTune/MPEG4 sound files. These files cannot be opened in GoldWave. Upgrade them to iTunes Plus to remove the copy protection/encryption. Upgraded files become .m4a files, which can be opened in GoldWave.	✗	✗	✗
.mp4 .m4a .aac	Unencrypted iTune/MPEG4/AAC sound files. On Windows 7, GoldWave may open these files through the Media Foundation	✗	✓ ANSI	✗

	system decoders and allow saving if the AAC encoder is installed. File information is preserved, but due to a flaw in Media Foundations, cover art is lost as is the copyright text. On older versions of Windows, if Apple's QuickTime player or iTunes software is installed, GoldWave will open these files through the QTFile plug-in, which is included with GoldWave. However files cannot be saved or created. File information is read from the files.			
.opus	Opus compressed sound files. Gives better quality compression than MP3 and covers a wide range of audio applications. Files are always encoded at 48,000 Hz. Refer to the Opus website for more information. File text information is supported.	✗	✓ UTF8	✓
.ogg	Ogg Vorbis compressed sound files. Gives better quality compression than MP3. Refer to the Vorbis website for more information. File text information is supported.	✗	✓ UTF8	✓
.raw	Headerless files containing binary data in 8 bit, 12 bit, 16 bit, 24 bit, 32 bit, single or double precision IEEE, mu-law, or A-law format.	✗	✗	✓
.sds	MIDI instrument sample dump standard format. Loop points are not supported. No support for file text information.	✗	✗	✗
.smp	Sample Vision 16 bit PCM sound files. Markers/Loops are not supported. No support for file text information.	✗	✗	✗

.snd	Raw or NeXT sound files. NeXT files are automatically detected. Opening Raw files displays the File Format window for attributes. No support for file text information.	✗	✗	✗
.txt	An ASCII text file containing a series of Y values (amplitudes) in human readable form. Values range from -1.0 to +1.0 (but may be +/-10.0) for floating point data and -32768 to +32767 for integer data. No support for file text information.	✗	✗	✓
.voc	Sound Blaster files. Supports: 8 bit mono/stereo, 16 bit mono/stereo, mu-law encoded mono/stereo. ADPCM compressed files are not supported since the compression algorithm must be licensed from Creative Labs. No support for file text information.	✗	✗	✓
.vox	Dialogic ADPCM encoded raw files. The File Format window is presented where you can specify the Telephony type and 4 bit VOX ADPCM format. You can use the [Options | File Formats] command to assign a default format for .vox files. No support for file text information.	✗	✗	✗
.wav	RIFF WAVE 8 to 32 bit PCM mono or stereo, A-law encoded, mu-law encoded, and Microsoft ACM compressed files. MPEG compressed audio is support only if the MPEG decoder is installed. Only files with one `data` chunk are supported. The chunks `fact`, `LIST INFO`, `LIST adtl`, and `cue` are detected. All others are	✓ ANSI	✓ ANSI	✗

	ignored. Cue points are supported. File text information is supported in ANSI only.			
.wma	Windows Media Audio files. Supports several different bitrates from low bandwidth to high quality lossless. File text information and cue points are supported.	— name only	✓	✓
.wmv	Windows Media Video files. The first audio track is extracted from the video file. File text information and cue points are read. Files cannot be saved in this format.	✓	✓	✓
.wv	WavPack compressed files. GoldWave does not support these directly. Installing the DirectShow Filter from the WavPack website may allow GoldWave to open and convert them.	✗	✗	✗
.xac	Extended Audio Container. Currently used only by GoldWave.	✓ UTF8	✓ UTF8	✓

✓- Supported　—- Limited support　✗- Unsupported

Normally, GoldWave detects and automatically opens all the supported file types. However, there are several cases where GoldWave may not be able to open a file:

1. The file does not contain any header information and there is no file association (see Options | File Formats).
2. The file type is recognized, but the file structure is invalid or corrupt.
3. The file uses a new compression method or format that GoldWave does not recognize.
4. The file type is not supported by GoldWave or any plug-ins.

If any of these conditions occur, GoldWave displays the File Format window (shown below) so that you can specify the type and attributes manually. GoldWave lists all the file format plug-ins that support reading raw audio data. If you are working with PCM or uncompressed binary data (like CD audio), select the Raw type. If you are working with

Telephony files, select the Dialogic type. Other types may be listed depending on what plug-ins you have installed.

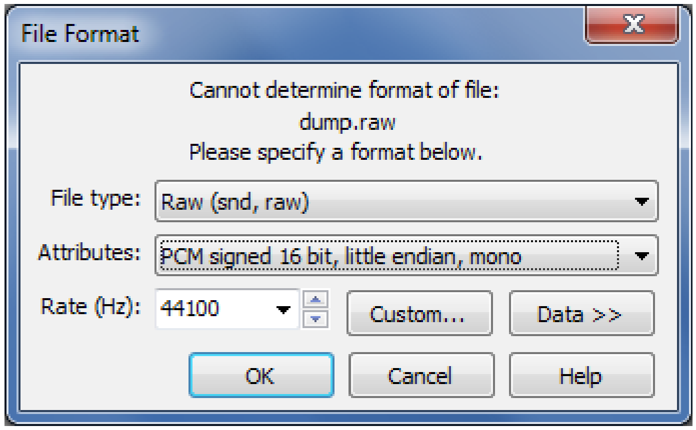


Figure: File Format

Table: Common Format Attributes

Format	**Description and Attributes**
PCM	Audio is uncompressed 8, 12, 16, or 32 bit data. A Windows system usually creates 8 bit, unsigned or 16 bit, signed, little endian data. A Macintosh system usually creates 8 bit, signed and 16 bit, signed, big endian data. The *signed* attribute tells GoldWave how the bits should be interpreted. The *endian* attribute tells GoldWave the byte ordering of the data. Big endian has the most significant byte first. Little endian has the least significant byte first.
Telephony	Audio is in a compressed format used in telephone applications. This includes mu-law, A-law, ISDN A-law (inverted A-law), and 4 bit ADPCM VOX Dialogic files.
Floating point	Audio is binary IEEE floating point *single* precision (32 bit) or *double* precision (64 bit) data. The byte ordering is usually little endian.
Text	Audio is a plain text (ASCII) file containing numbers. The *float* attribute tells GoldWave that the numbers range from -1.0 to 1.0. The *integer* attribute tells GoldWave that the numbers range from -32768 to 32767.
Encoded	Audio is compressed using an encoding algorithm. Such files cannot be opened properly through the File Format window and require a separate plug-in for decoding.

If you do not know the format, experiment with trial-and-error. Appendix A has more information about sound attributes. Start with an 8 bit or 16 bit PCM attributes, then try the mu-law or A-law formats. Generally, sounds will be noisy if the format or number of bits is incorrect, in which case you will have to close and reopen the sound using a different format. You can leave the sampling rate unchanged since it affects only the playback speed and can be changed later using Effects | Playback Rate.

File Format Plug-ins

GoldWave supports external file format plug-ins for opening and saving files. These plug-ins are created by other developers by using the GoldWave Plug-in Development Kit to handle file types that GoldWave does not support directly.

You can use Options | File Formats to enable and disable plug-ins or to change the order in which they are used.

When you open a file in GoldWave, these steps are followed:

1. If the file type is a CD audio (CDA) track, you are advised to use the CD Reader tool and no further processing occurs.
2. For all other file types, the file is passed to each file format plug-in module until one is able to handle the file. The order is configured under the File Plug-in Precedence tab of the File Format Options window.
3. If none of the plug-ins support the file format, then the Undetectable Types list under the File Format Options window is used to determine if type and attributes have been associated with the file type. If so, the file is open automatically using those attributes.
4. If there are no associations, then the File Format window (see previous section) is displayed so that the attributes can be specified manually. Chances are that compressed files cannot be open and decoded properly unless a new plug-in is installed for that file type.

Effects Overview

Effects modify, enhance, and change sounds in a variety of ways. These commands are similar to font menu commands in word processors. For example, using font commands, you can change the size of the letters. In GoldWave, using the Volume | Change Volume effect changes the "size" of a sound. Changing the colour of a font would be similar to changing the pitch of a sound.

For an introduction to some of the terms used in this section, refer to the Editing Overview section and Appendix A. A variety of volume scales may be used by effects.

Most effects in GoldWave are cumulative. This means that if you use the same effect with the same settings, then the sound is changed each time. For example, if you use the Volume | Change Volume effect with a value of -6.02dB, then the volume of the sound decreases to half its current level. If you use that effect again, the volume decrease again, giving one quarter the original volume.

Another example is the Time Warp effect. If you specify a change of 50%, then time is slowed to half and the sound is twice as long. Using the effect again at 50% makes the sound four times as long.

There are a few exceptions. The Maximize Volume effect has an absolute setting. Maximizing the volume to 0dB sets the sound's peak volume to 0dB. Using the effect again at 0dB has no affect.

Common Controls for Effects

Many effects have similar controls such as presets and shape boxes. These are explained below.

Graph Window Left-click to add or drag-and-drop point
Right-click to remove point Shape Line Shape Controls Preset Controls Preview Controls Current Point
Figure: Common Controls for Effects

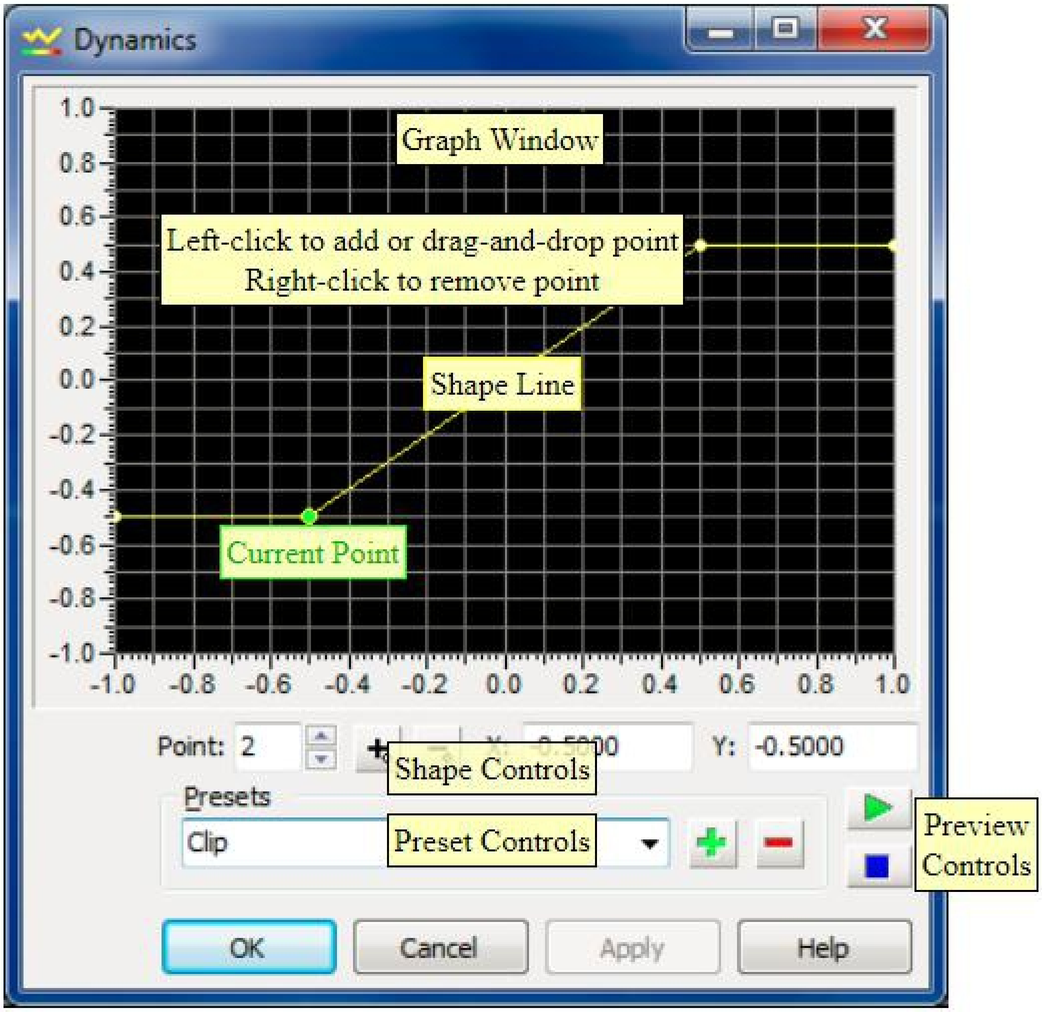


Figure: Common Controls for Effects

Presets

Presets store settings, parameters, and shapes (described below) for quick retrieval the next time the effect or command is used. Controls for presets consist of a drop down list box, an add  button, and a remove  button, as shown in the figure above.

To add a new preset:

1. Enter in all the new parameters and/or draw the new shape.
2. Type in a new name for the preset in the drop down list. This name should be the same as one currently in the list unless you intend to replace it.
3. Choose the add  button.

To delete a preset:

1. Select the preset from the drop down list.
2. Choose the remove button.
3. Choose the Yes button to confirm deletion.

To change a preset:

1. Select the preset from the drop down list.
2. Change the parameters.
3. Choose the add button.
4. Choose the Yes button to overwrite the preset when prompted.

If you change any of the presets installed by GoldWave, including the **Default** preset, they will be reset to their original settings if you reinstall or update GoldWave. Use a different preset name to retain a preset across updates.

Shape Controls

Several effects in GoldWave use Shape Controls to set graphical parameters or dynamically alter the effect across the selection. Shape Controls usually consist of a graph window and a set of controls, including a point number box, an add point button, a remove point button, an X value box, and a Y value box as shown in the figure above.

Graph Window
The graph window initially contains a single line with two endpoints, shown as large dots. By clicking the left mouse button anywhere inside this window, you can add new points to bend the line into a variety of zigzag shapes. To move a point, click on it and drag it to a new location. To remove a point, click the right mouse button over the point. Note that endpoints cannot be removed.

Controls
Points can be added, moved, and removed by using these controls. Use the `Point` box to select the current point. Change the X and Y values to move the point. Use the `add point` button to insert a new point between the current point and the next point. Use the `remove point` button to remove the current point, except if it is an endpoint.

Some dynamic effects, such as Doppler, Pan, and Shape Volume start previewing audio based on the current point's time value. If the X value of the current point is 1:00, for example, then preview playback starts at that time rather than at the beginning of the selection. This lets you preview the point's settings without playing the entire selection.

To save a shape, use the Presets controls, explained above.

Preview Controls

Previewing is a way of listening to how the current effect settings will sound without having to process the entire selection first. Preview Controls consist of a play button and a stop button. When you press the play button previewing usually starts at the beginning of the selection. In some cases, such as effects that have time based shapes, previewing will start at the current point's time rather than at the beginning of the selection.

In most cases, you can change the effect settings while the preview plays and hear the changes as they are made. For more complex effects and most shape based effects, you will need to use the **Apply** button to apply the changes or shape while previewing.

The **Apply** button is used only for previewing. It does not apply the effect to the sound file. Use the **OK** button to process the file.

FFT Settings

When performing Fast Fourier Transform (FFT) processing, the sound is divided into small blocks and processed one block at a time. The `FFT size` value controls the size of the these blocks. The number of samples to process is calculated by taking the value as a power of 2. A value of 10 gives 2 to the 10th power, or 1024 samples. By increasing the number of samples, frequencies are processed at a higher resolution, which helps to eliminate chirping and other mechanical sound distortions, but it tends to add more echo. Usually values from 11 to 12 give the best trade-off between distortions and echo.

To smooth out transitions from one block to the next, it is necessary to overlap blocks. The `Overlap` value controls how much of the FFT analysis of one block overlaps the next. A high value makes the transition between each block smoother. It also requires more processing time since overlapping samples are recalculated several times. A low value may result in rougher transitions, but processes faster. For complex audio and tempo or pitch modifying effects, higher values may give better quality.

Effect Plug-ins

Effect plug-ins are modules developed by other companies that can be used within GoldWave. These appear under the Effect | Plug-in menu. DirectX and VST plug-in wrappers are available for GoldWave to add support for many of the existing DirectX and VST Audio Plug-ins. Other plug-ins are designed to work with GoldWave directly and will appear as separate items under the plug-in menu, each with its own submenu of effects. In some cases settings for effect plug-ins can be changed using the Options | Plug-in | Effect menu.

If any errors or exceptions occur while using a plug-in, you'll need to contact the plug-in creator for assistance. GoldWave Inc. does not provide support for plug-ins created by separate (third party) developers.

Accessibility Overview

This section outlines the accessibility features in GoldWave. Almost all of GoldWave's functionality is accessible through the keyboard. See Keyboard Commands and Options | Keyboard.

Interface

If a screen reader is active when starting GoldWave, the program automatically changes its interface slightly to be more screen reader friendly.

Screen reader mode can be switched on manually by using **GoldWave Setup** in the Windows Start menu and checking the `Force screen reader mode` box.

The status visual in the Control window becomes a text box that gives the elapsed time, the current status, and the left and right peak levels (in parentheses). The levels are reset whenever playback or recording is restarted. The levels are given in percentages, but tabbing to the text box and pressing `D` changes it to decibels. Pressing `D` again changes it back to percentages. Press `R` to reset the peaks. Press `P` to pause updates to give time for the screen reader to read it. These keys only work when the status visual has the keyboard focus and the keys have not been assigned to other functions in GoldWave.

All images are removed from the menus so that standard Windows text menu items are used instead of custom drawn ones, ensuring that menus are easily readable. If menus cannot be read, be sure to uncheck the "Tool bar images in menu" box under Options | Tool Bar.

Also when a screen reader is active, GoldWave defaults to shared playback instead of exclusive playback so that the screen reader can share the audio device with GoldWave.

Almost all windows in GoldWave provide edit boxes where values and settings can be entered rather than relying on visual controls only. Use the `Tab` key to move between controls.

Use `Alt+F6` to switch between GoldWave's Main window and Control window. Use the `Tab` key to move through the buttons and faders. The arrow keys change the fader settings. Unless you need to access the volume, balance, or speed faders, it is best to dock the Control window. Use the **Tool | Control** command to dock or undock it.

Navigation

Finding certain parts of a sound file is done through keyboard navigation, which involves playing the file and moving the playback cursor until the area if interest is located. GoldWave includes many keys for playing different parts of the sound. They are listed in the following table. The amount that the cursor is moved depends on the zoom level, explained in the View section below.

The View | Auto Scroll Lock menu item **must** be checked for keyboard navigation to work.

Table: Keyboard Navigation

Keystroke	**Action**
Space	Starts playback or stop it (toggles playback). The region that is played depends on the Play 3 button settings.
J, K, L	Rewinds, plays, and fast forwards respectively. Playback always starts at the cursor's position.
Shift+J, Shift+K, Shift+L	Makes the playback speed slower, normal, and faster respectively.
F2, F3, F4, F5, F6, F7, F8	Plays 1, plays 2, plays 3, rewinds, fast forwards, pauses, and stops respectively. See Play Properties for the play button settings.
Ctrl+G	Sets the playback cursor to a specific time (go to).
H	Starts playback relative to the mouse's horizontal position in the waveform.
Shift+[	Plays three seconds of audio up to the start marker.
Shift+]	Plays three seconds of audio up to the finish marker.
Ctrl+[	Plays from the start marker to the finish marker.
Ctrl+]	Plays from the finish marker to the end.
Left	Move the cursor backward.
Right	Move the cursor forward.
Page Up	Moves the cursor one screen backward.
Page Down	Moves the cursor one screen forward.
Home	Moves the cursor to the start marker's position.
End	Moves the cursor to the finish marker's position.
Ctrl+Home	Moves the cursor to the beginning of the sound.
Ctrl+End	Moves the cursor to the end of the sound (playback stops).

Editing

In addition to using the Edit | Marker | Set and the Edit | Marker | Drop... commands, there are a number of ways of changing the selection with the keyboard. They are listed in the following table. The amount that the selection markers are moved depends on the zoom level, explained in the View section below.

Table: Keyboard Selection

Keystroke	Action
Shift+Left, Shift+Right	Moves the start marker left or right (backward or forward). See the Marker preview playback setting.
Ctrl+Shift+Left, Ctrl+Shift+Right	Moves the finish marker left or right.
Ctrl+A	Selects the entire sound.
Shift+Home	Moves the start marker to the beginning of the sound.
Shift+End	Moves the start marker to the finish marker's position.
Ctrl+Shift+Home	Moves the finish marker to the start marker's position.
Ctrl+Shift+End	Moves the finish marker to the end of the sound.
[(left bracket)	Drops the start marker at the playback cursor.
] (right bracket)	Drops the finish marker at the playback cursor.

After deleting or cutting the selection, the selection is empty, so if the playback buttons are set to play the selection, nothing will play. Use **Ctrl+A** to select the entire file.

Keep in mind that most of the playback keys stop playback at the finish marker and there may be more audio beyond it. Use **Edit | Trim** to remove any audio outside the selection, if you only want to save what is in the selection.

If recording is stopped, the finish marker is moved to the stop time. To continue recording from that point, use **Shift+End** to move the start marker to the finish marker, then use **Ctrl+Shift+End** to move the finish marker to the end. Starting recording will continue from where it was stopped.

View

The view and zoom level control how much audio is displayed on the screen (or page). By default, GoldWave displays the entire file. Use the View Menu Commands to change the amount of audio displayed. This also changes the amount the playback cursor and selection markers are moved with each keystroke. The cursor is moved one tenth of the screen size. Selection markers are moved one hundredth of the screen size. Using **View |**

10 Seconds displays 10 seconds of audio in the view. The `Page Up` or `Page Down` key moves the cursor 10 seconds backward or forward. The `Left` or `Right` key moves the cursor 1 second backward or forward. The selection keys move the start or finish marker a tenth of second (0.1). Using **View | 1 Second** displays 1 second of audio. At that level, the page, arrow, and selection keys move 1 second, one tenth of a second, and one hundredth of a second respectively.

The `Shift+Up` and `Shift+Down` keys zoom in and out by 75%. The status bars display the selection range and the current zoom level.

Use Window options to change the default initial zoom level when a file is opened.

File Menu Commands

This section explains commands under the GoldWave's File menu. The File Overview section provides general information about how GoldWave handles files.

New

Use New to create a new sound with attributes you specify. These attributes are discussed in Appendix A. Note that GoldWave allows you to create and edit sounds that may not be playable with your audio hardware. For CD quality, use stereo, with a sampling rate of 44100Hz. Several commonly used settings are provided in the Presets list. Use the `Initial file length` box to specify the time length of the file (see Entering Times). You can alter the length later with Edit | Trim or Edit | Inserting Silence.

To change the default save format shown in the status bar, see File Formats. To change the recording bit depth, see Device Properties

Open

The Open command presents a list of files in your sound folder. The sound folder can be set using the Options | Storage command. All recognized file types are listed. After you select a file, a Sound window is opened and details about the sound are displayed in the status bar. See the File Format section above if GoldWave could not open the file.

The Storage Overview section explains how the files are stored for editing. Depending on the size of the file, you may want to change the storage setting under Options | Storage.

Open URL

Use Open URL displays a window where you can enter the location of a remote file on a website to open. An active network connection is required. The file must have a finite duration. If a stream is opened, the program can never finish opening the file.

The URL is similar to a website address and must begin with the protocol, followed by the website, followed by the file's location, such as `http://www.somewhere.com/somefolder/somefile.mp3`.

⚠Only enter URLs of trusted websites. Always be vigilant when accessing any remote content.

ℹThis command may not be present on Windows XP or earlier. Windows Vista or later is required.

Close

Use Close to close the current sound. If any changes were made, you are asked to save them.

Close All

Use Close All to close all sound windows. If any changes were made to any of the sounds, you are asked to save each one.

Information

This command assigns or changes text information stored in the file, such as artist, title, copyright, and date. Information is stored in certain file types only, such as `.wav`, `.wma`, `.aiff`, `.xac`, `.ogg`, `.opus`v `.mp3` and `.m4a`. Some file types only store a subset of all the items given in the File Information window. Cover Art is supported only in `.wma`, `.ogg`, `.opus`, `.mp3`, and `.m4a` files. There is no verification of the information entered. It is up to you how to use these items and to follow any guidelines required for a particular file type.

⚠Do not replace artwork with high resolution photos from a camera! Such photos can easily exceed 3MB in size, which is excessive and completely unnecessary for artwork. Use image editing software (Windows Paint) to reduce the size and resolution of the photo. The artwork must be cropped square. A resolution of 512 by 512 pixels is recommended. The file size should be under 300KB when possible.

⚠Any additional information and metadata within a file not shown in this window may not be preserved by GoldWave.

Use the Copy and Paste buttons to copy information from one file and paste it into another. Use the Clear All button to erase all text and the artwork.

The File Format section lists the file formats supported by GoldWave and the level of text information retained. Formats supporting unicode text allow international (non-latin) character sets to be used.

When using Batch Processing or when splitting a large file with the Split File feature, the Track number can be set to ## (two pound signs). During processing or splitting, the pound signs are replaced with the sequential number of the file being processed or split.

Save

The sound is saved in a file using its original name and type. If memory or disk space is low, the file may not be saved successfully. GoldWave will inform you if this happens. If Save fails, try deleting some unneeded files or close other applications. Make sure that the file is saved successfully before closing GoldWave, otherwise the changes will be lost. Note that audio from video and movie files cannot be saved. You must save those files in an "audio only" format. You will see the Save As window if you need to save the file in a different type.

Cue points are saved only in certain file types. If you added cue points to a file that does not support them, you can use File | Save As to save it in a different type or save them in a separate Cue File.

An option to confirm saving is given under the Options | Window command.

Save As

Save As saves a sound using a different filename or file type. To save the sound using a different name, simply type in the new name in the `File name` box. To save the sound using a different type, select the type from the "Save as type" list box, then select attributes from the `Attributes` list box. If a file type supports customized attributes, then the `Attributes` label will become a link you can choose to display a configuration window.

Since each file type supports different attributes, always select the type before selecting attributes. Java and Web sounds, for example, should be saved using the "Sun (*.au)" type and the "Java/Web" attributes.

You can use Save As to encode a sound in a compressed format, such as MP3, M4A/AAC, WMA, Ogg, etc. To save as MP3, select the "MPEG Audio (*.mp3)" type and one of the many attributes. The smaller the bitrate (kbps number), the smaller the file will be. Note that quality may be reduced as well. To save as M4A (iTunes/AAC), select the "Media Foundations (*.m4a)" type. This option works only on versions of Windows that include the AAC encoder.

Use Options | File Formats to assign a default format if you always prefer to use a specific file type and attributes.

Unless the number of channels or sampling rate has changed, GoldWave does not use the compressed audio data after saving. It continues to use the original audio stored in temporary storage. You must close and reopen the file in GoldWave (or play it in a separate program) to hear how the compressed audio sounds. It is strongly recommended that you listen to the compressed file before discarding the original to ensure its quality is acceptable.

The correct type must be selected from the type box. Typing in a different extension by hand for the filename does not convert the sound to the type associated with that extension.

Save All

Saves all modified sounds. Since saving all sound can take some time, you are asked for confirmation before all sounds are saved.

Save Selection As

Save Selection As saves the selected part of the sound to a file. Use this command to save parts of a large file. The Save As window appears where you can specify the new filename, type, and attributes for the file.

Batch Processing

Batch Processing is a powerful tool for processing and converting a set of files automatically.

Use Batch Processing to:

- Compress all your `.wav` files to `.mp3` to save disk space
- Convert iTunes `.m4a` to `.mp3` for playback on any portable audio player, USB flash drive for you car or TV, etc.
- Adjust volume levels of all your songs so some songs aren't louder than others
- Update file information, metadata, tags
- Remove pops/clicks, equalize, and maximize volume levels to restore all your old vinyl albums or tape recordings
- Insert or append an audio leader or trailer in a set of files
- Create a comprehensive list of effects and edits that you use frequently so you can apply them to files currently opened in GoldWave in a few clicks

Each tab contains settings to configure edits and effects, conversions, destination folder, and file information. These are explained below.

When everything is configured, you can use the preset controls to save all the settings, then choose the Begin button to start processing all the files. A status window will appear showing the progress and listing any errors that occur.

Files are processed one at a time in the order they are listed. Each file is processed as follows:

1. The file is opened (if not currently opened) and decompressed or decoded into GoldWave's internal audio format. RAM or hard drive storage is used depending on the Options | Storage setting.
2. Effects and editing commands are executed in the order they appear on the Process tab.
3. File information is changed based on the settings on the Information tab.
4. The file is converted, compressed, or encoded into the file type and attributes specified on the Convert tab and saved in the destination folder specified on the Destination tab.
5. The original file is overwritten or deleted only if that option is selected on the Destination tab.

To process raw files that GoldWave cannot open automatically, such as `.vox` or `.pcm`, use Options | File Formats to add a default format for that type.

Source Tab
Use these controls to specify files to process. Select "Current Sound window" to process the currently active Sound window opened in GoldWave. Only items on the Process tab are applied. No other changes are made. The file is not converted and the information is not changed. Select "All Sound windows" to process all opened Sound windows in GoldWave in the same manner.

Select "Files and folders" to create a list of files to process. Add files with the Add Files button or drag-and-drop a group of files from Windows Explorer. Add an entire folder (including subfolders) or all the files in a folder of a specific type by using the Add Folder button. Remove items from the list by selecting one or more of them and choosing the Remove button. The Remove All button removes all files and folders from the list.

Process Tab
If you want to apply effects or edits to a group of files, use this tab to add a set of effects, edits, or chains to the list. If no effect processing is required, remove all effects by using the Remove All button on this tab. To remove a single effect from the list, select the effect and use the Remove button. To change the order of processing, drag-and-drop items within the list.

To add an effect, use the Add Effect button to display a tree list of all effects available and their presets. Select the preset you want to use. If you require custom effect settings, you must create new presets for the effect outside of Batch Processing. To do that, open a file, display the effect window, adjust the settings, then add a preset. Chains may be added by using the Add Chain button. Effects and chains are performed on the entire file by default. To perform them on part of the file only, add an edit to set the selection first (see the fade in/out example below).

To add an edit, use the Add Edit button. The "Set Marker/Selection" edit command specifies what part of the file (the selection) to use for all subsequent effects and edits during processing. The selection can be specified using time, percent, or cue point names or indexes. If using the Cue option to set the selection, the first cue point matching the given name is used. If a number is given, cue points are searched for a name matching the number. If no such cue point is found, then the number is used as an index in the list of cue points. So if the number `6` is given, for example, a cue point with the name "6" is searched for first. If that name isn't found, then the sixth cue point is used. The "Quick selection" drop down list contains many examples of selection settings.

To change settings for an edit or effect, remove it, add a new one with the correct settings, then drag-and-drop it to the correct position in the process list (if necessary). If you require effect settings that are not available in any of the current presets, open a file and use the Effect menu or the Effect Chain Editor tool to create a new preset or chain with the settings you require prior to using the Batch Processing command.

To add or edit a comment, either double click a line in the process list or select a line and press the quote key (`'`). If you are changing or adding a comment to an existing preset, be sure to update the preset by choosing the preset Add button.

Process Example

You have hundreds of songs and want to create a 10 second sample file for each song, with the beginning faded in, the end faded out, and the volume maximized. To set up that processing requires adding eight edits and effects, as explained below.

- Trim the file to 10 seconds:
    1. Choose the Add Edit button.
    2. Select the "Set Marker/Selection" edit command.
    3. In the "Finish marker position" group, set the `Time` to 10 and choose `After beginning` from the "Where" drop down list.
    4. Choose the Add button.
    5. Select the "Trim" edit command.
    6. Choose the Add button.
    7. Choose the Close button.
- Fade-in the beginning of the file:
    1. Choose the Add Edit button.
    2. Select the "Set Marker/Selection" edit command.
    3. In the "Finish marker position" group, set the `Time` to 1 and choose `After beginning` from the "Where" drop down list.
    4. Choose the Add button.
    5. Choose the Close button.
    6. Choose the Add Effect button.
    7. Expand the "Fade In" branch.
    8. Select the "Silence to full volume, linear" item.
    9. Choose the Add button.
    10. Choose the Close button.
- Fade-out the end of the file:
    1. Choose the Add Edit button.
    2. Select the "Set Marker/Selection" edit command.

3. In the "Start marker position" group, set the `Time` to 1 and choose the `Before end` from the "Where" drop down list.
4. Choose the Add button.
5. Choose the Close button.
6. Choose the Add Effect button.
7. Expand the "Fade Out" branch.
8. Select the "Full volume to silence, linear" item.
9. Choose the Add button.
10. Choose the Close button.

- Maximize the volume of the entire file:
    1. Choose the Add Edit button.
    2. Select the "Set Marker/Selection" edit command. Settings select the entire file by default.
    3. Choose the Add button.
    4. Choose the Close button.
    5. Choose the Add Effect button.
    6. Expand the "Maximize Volume" branch.
    7. Select the "Full dynamic range" item.
    8. Choose the Add button.
    9. Choose the Close button.

After performing all these steps, the process list will contain eight edit and effect items. Use the Convert and Destination tabs to set the file type and destination folder for the 10 second sample files (so the original files are not overwritten). Add all the songs to the Files list and begin processing.

To use clipboard related edit commands, it may be necessary to open a file and copy it prior to using Batch Processing. For example, to paste or mix an announcement at the beginning of many files, first open the file containing the announcement, use the Edit | Copy command, then use the Batch Processing command and add a "Paste" or "Mix" edit command to the Process list.

Convert Tab
If the "Convert files to this format" box is checked, then files are converted to the format specified on this tab. Otherwise no conversion is performed and a processed file will have the same format as the original file, if possible. If the same format cannot be used, then an error is reported.

Use the "Save as type" drop down list to select the destination format for the conversion, then use the `Attributes` drop down list to select the specific attributes to use for the destination type. If a save type supports customized attributes, then the `Attributes` label will become a link you can choose to display a configuration window.

If the attributes allow any sampling rate to be used, you can specify the destination rate to use by checking the `Rate` box and entering the rate in the box. Some attributes have a fixed rate, so a separate rate cannot be specified for those. If no rate conversion is needed, make sure the `Rate` box is not checked. In that case, a processed file will have the same rate as the original file.

Destination Tab

If you want all processed files to be stored in the same folder where they currently reside, select "Store all files in their original folder".

If you want all processed files to be stored in a specific folder, select "Store all files in this folder" and specify a folder in the box provided. You can click on the folder button to browse for a folder.

When adding an entire folder with subfolders by using the Add Folder button, use the "Preserve subfolder structure" box to ensure that the relative subfolders are maintained when storing processed files in a different destination folder. If this box is not checked, all files will be stored in the destination folder and no subfolders will be created. Folders listed in the file list will have double backslashes. The part before the double backslashes will be replaced by the destination folder. For example, if the destination folder is `C:\Folder1\` and the added folder (with subfolders included) is `C:\Source1\Source2\`, then the file list may look something like this:

```
C:\Source1\Source2\*.*
C:\Source1\Source2\\Source3\*.*
C:\Source1\Source2\\Source4\*.*
C:\Source1\Source2\\Source4\Source5\*.*
```

Note the double backslashes. The destination folders will be:

```
C:\Folder1\*.*
C:\Folder1\Source3\*.*
C:\Folder1\Source4\*.*
C:\Folder1\Source4\Source5\*.*
```

Taking the last item, the folder is divided into three parts with the following colour coding: the `original source root folders`, the `subfolders`, and the `pattern`.
`C:\Source1\Source2\\Source4\Source5\*.*`
The destination folder is:
`C:\Folder1\`
The original source folders are replaced by the destination folder to give:
`C:\Folder1\Source4\Source5\*.*`

⚠Using the Add Folder button more than once when preserving subfolders is not recommended unless you fully understand how the source folder, subfolders, and destination folder are manipulated.

To overwrite any files having the same name and folder as the processed file, check the "Overwrite existing files" box. GoldWave fully processes original files before overwriting them. Check the "Only if older than the original" box to process and overwrite files that have not been processed or modified recently. Files are processed only if the destination file does not exist already or if the destination file is older than the original file.

⚠Overwriting or deleting original files is **not** recommended. Processing lossy formats such as MP3 and iTunes M4A reduces the quality. Always keep a copy of the high quality original files.

Information Tab
Use this tab to control how information is processed. See File | Information. Select "Retain current text information" if you do not want the file's information to be changed. Select "Replace text information" and use the `Set Info` button to provide all the information. Any blank entries are removed from the file. Select "Replace specified text information only" to replace some of the information. Use the `Set Info` button to enter the information to replace. Any blank entries are not removed or changed in the file, except for the Title. GoldWave uses the file's name as the title text when a file currently has no title. Select "Remove text information" to remove all text information from the processed file. The title text is not set. Select "Remove text information and pictures" to remove text, cover art, and other pictures.

ℹIf the `Track number` is set to `##`, GoldWave replaces it with a sequential number based on the order in which the files are processed. The first processed file will have a track number of 01, then the next file will have 02, etc. If the `Title` is left blank and the file does not have a title, GoldWave sets it to the file's name.

⚠Depending on the settings used, all processed files may have exactly the same information (except for `##` track numbering and title mentioned above), so care must be taken when specifying file specific information such as the `Title`. Also note that not all file types can store information.

⚠When retaining information and converting to a different file type, some information from the old file type may not be valid for the new file type. GoldWave does not verify any information, so care must be taken when retaining information while converting.

Command Line

The `-process` command line parameter starts GoldWave in Batch Processing mode and processes files given on the command line. The basic syntax is as follows:

```
  "C:\Program Files\GoldWave\GoldWave.exe" -process[:preset] <filespec>
[<filespec> ...]
```
If no preset is specified, then the default (previous) settings are used. The `filespec` can specify a single file or contain wildcard characters (* and ?). Quotes are required if the `filespec` contains any spaces. Other parameters include `-proclog:filename` to set a log file, `-region:start,length` to set the initial selection for processing, and

`-clipboard:filename` to set the initial contents of the edit clipboard, and `-outfolder:pathname` to set the destination folder.

To convert all Wave files to MP3 on the command line, for example, first use the Batch Processing command to create a preset with conversion set to the "MPEG Audio" file type and save that preset as "MP3". On the command line, enter the following parameters:

```
-process:MP3 "C:\My Music\*.wav"
```

If the preset name contains spaces, use quotes around the entire parameter:

```
"-process:Trim and convert" "C:\My Music\*.wav"
```

To include all subfolders when processing, include the `-subfolders` command line parameter:

```
-process:MP3 "C:\My Music\*.wav" -subfolders
```

Make sure all folder settings under the Destination Tab are set appropriately in the preset before using this parameter and specify the absolute pathname of the folder (do not use relative pathnames). To specify a different destination folder than the one contained in the preset, include the `-outfolder` command line parameter. The "Preserve subfolder structure" setting must be set in the preset if the hierarchy is to be preserved. Quotes must be used around the entire parameter if spaces are used:

```
"-outfolder:C:\My Music\Batch Destination"
```

During processing the progress windows is displayed in minimized form, usually just above the Windows Start button. Double-click on it to open it to monitor processing. If no errors occur, the window will disappear and GoldWave will close automatically. Otherwise the progress window opens so that errors can be viewed. To use a log file instead, so that GoldWave always closes regardless of errors, use the `-proclog` command line parameter to specify the name of the log file:

```
-proclog:filename
```

Quotes are required around the entire parameter if the filename contains spaces:

```
"-proclog:C:\Log Files\Batch.log"
```

All messages and errors are written to the log file. If you do not require a log file, but still need GoldWave to always close, use `nul` as the filename:

```
-proclog:nul
```

To process many commands in a single instance, store the list of command lines in a text file and use the `@` symbol to specify that file in the `process` parameter. For example:

```
"C:\Program Files\GoldWave\GoldWave.exe" -process@batch.txt
-proclog:batch.log
```

If a log file is specified, it will be used by default for all command lines in the list file. A different log file may be specified within the list file too.

Each command line must be stored on a single line in the list file. Do not break a command line across multiple lines. A list file would look like this:

```
  -process:Trim *.wav -region:10.0,30.0
  "-process:Convert to MP3" "C:\My Music\iTunes\*.m4a" -subfolders
-proclog:itunes.log
  "-process:Match Volumes -18dB" *.wav
```

Exit

Exit closes all Sound windows then closes GoldWave. Any playback or recording is stopped. You are asked to save any changed files.

File History

A list of several recently used files is appended to the File menu. You can quickly reopen one of these files by selecting it from the menu.

Edit Menu Commands

Edit commands remove, insert, copy, and move sections of sound. For an introduction to the concepts and terms used in this section, refer to the Editing Overview section.

Undo

The Undo command reverses the most recent change made to a sound. The undo feature keeps a copy of the original data in a temporary file. The temporary folder and the number of undo levels are determined by the Options | Storage settings. To disable Undo, set the number of undo levels to zero.

Redo

The Redo command reverses the most recent undo. It restores the last undone command without any processing. The is helpful if you accidentally undo a change (such as a recording) and want to recover the change or if you want to do a quick "before and after" comparison. Redo is possible only immediately after undo. If you make any other changes to the sound, Redo cannot be used again until you Undo one or more of those changes. Many levels of Undo and Redo are possible. In other words, you can undo several changes and redo them all.

Cut

Use Cut to remove the selection from the sound and put it in the clipboard. The contents of the clipboard can then be superimposed or inserted into a Sound window using Mix or Paste. If you just want to remove the selection and do not need to paste or mix it, you should use the Delete command instead.

Note that if only one channel is selected in a stereo sound, then only that channel is removed. Since it is not possible for one channel to be longer than the other, the end of the cut channel is padded with silence (this is also true for Delete).

To cut:

1. Select the part of the sound you want to cut.
2. Choose **Cut** from the **Edit** menu.

Copy

The Copy command copies the selection into the clipboard. The selection is not removed from the sound. The contents of the clipboard can then be mixed or placed into another sound using Mix, Paste, Paste New or Replace.

To copy:

1. Select the part of the sound you want to copy.
2. Choose **Copy** from the **Edit** menu or click on the **Copy** button.

You can copy individual channels of a stereo sound by using the Edit | Channel menu to select a single channel.

Copy To

The Copy To command copies the selection to a new file. Use this command to divide a large file into smaller sections or save a piece of a file. The selection is not removed from the sound. The Save As window appears where you can specify the filename, type, and attributes for the file. This is the same as the File | Save Selection As command. To automatically split a large file into several smaller pieces, use the Cue Points tool.

To copy the selection to the file "section.wav":

1. Select the part of the sound you want to copy.
2. Choose **Copy To** from the **Edit** menu.
3. Enter the filename: **section**
4. Select "Wave (*.wav)" from the type list.
5. Choose **Save**.

You can save individual channels of a stereo sound by using the Edit | Channel menu.

Paste New

The Paste New command creates a new Sound window containing the sound copied into the clipboard. The new sound will have the attributes and length of the clipboard sound. This command is useful when you need to alter or save the clipboard audio.

To paste part of a sound into a new sound:

1. Select the part of the sound you want to copy.
2. Choose **Copy** from the **Edit** menu.
3. Choose **Paste New** from the **Edit** menu.

Paste and Paste At

After copying a sound into the clipboard, you can use these commands to insert it into another sound. The Paste command inserts the clipboard at the start marker's position. Paste At inserts the clipboard at the location you specify: at the beginning of the file, at the finish marker, or at the end of the file. The length of the sound is increased so that the clipboard will fit. The clipboard is automatically converted to match the attributes of the sound.

To insert the clipboard into the sound:

1. Move the start marker to the place where you want to paste the clipboard sound.
2. Choose **Paste** from the **Edit** menu.

To append the clipboard to the end of the sound:

1. Choose **Paste At** from the **Edit** menu.
2. Choose **End** from the **Paste At** submenu.

Copying a small selection and pasting it several times creates a stutter effect.

Mix

Use Mix to blend (combine) the clipboard with the sound. Note that before you can use Mix, you need to use the Edit | Copy command to copy audio into the clipboard. Mixing essentially allows two sounds to be played at the same time, such as vocals and music or voice-overs. You are prompted for the mix volume applied to the clipboard and the start time. A volume of 0dB is full volume. Lower values make the clipboard sound quieter. Adjust the start time to synchronize the clipboard audio with the Sound window audio you are mixing with. The Invert options allows the clipboard audio to be inverted before mixing, subtracting the audio amplitudes instead of adding them.

To mix the clipboard with the sound:

1. Move the start marker to the place where you want to start mixing the clipboard.
2. Choose **Mix** from the **Edit** menu.
3. Enter the mix volume for the clipboard.
4. Adjust the start time as required.
5. Choose OK.

Crossfade

Use Crossfade to fade and mix the ends of two sounds, such as fading out one song while fading in another. Note that before you can use Crossfade, you need to use the Edit | Copy command to copy the entire song into the clipboard. See the Crossfading overview for step-by-step instructions.

The `Duration` specifies how long the transition between the clipboard audio and selection lasts. The `Clipboard position` setting specifies which end of the selection to fade the clipboard audio. If `End of selection` is selected, then the clipboard is pasted and crossfaded at the end of the selection. If `Beginning of selection` is selected, then the clipboard is pasted and crossfaded at the beginning of the selection. The `Fade curve` settings control how the audio is faded. If the song being faded out already has a fade out at the end, use the `None` fade out curve setting. The fade is shown graphically and can be previewed. Note that any changes made while previewing will not take effect until preview is restarted.

The selection is used when performing a crossfade. Use the Edit | Select All command before using Crossfade unless you intend to crossfade within a certain part of the sound.

Replace

Use Replace to replace the selection with the clipboard. The selection is deleted and the clipboard is inserted in its place. If the clipboard is longer or shorter than the selection, the length of the file is adjusted as required. To avoid changing the length of the file or altering the timing of the sound following the selection, use Overwrite instead.

To replace part of a sound with the clipboard:

1. Select the part of the sound you want to replace.
2. Choose **Replace** from the **Edit** menu.

Overwrite

Use Overwrite to overwrite part of the sound with the clipboard beginning at the start marker's position. The amount of sound overwritten depends on the length of the clipboard. The length of the file is not changed (unless the clipboard would go beyond the length of the file) and no shifting is performed. Note that if the clipboard is longer than the current selection, then some sound outside the selection will be overwritten as well. The finish marker will be placed at the end of the overwritten sound.

Use Overwrite instead of Replace when you need to preserve tempo, timing, or alignment of the sound following the selection.

To overwrite part of a sound with the clipboard:

1. Move the start marker to the beginning of the sound you want to overwrite.
2. Choose **Overwrite** from the **Edit** menu.

Delete

The Delete command removes the selection from the sound. The selection is not copied to the clipboard. You should always use Delete instead of Cut when the selection is not needed. The Delete command is faster because it does not copy the selection to the clipboard.

Note that if only one channel is selected in a stereo sound, then only that channel is removed. Since it is not possible for one channel to be longer than the other, the end of the deleted channel is padded with silence (this is also true for Cut).

To delete:

1. Select the part of the sound you want to delete.
2. Choose **Delete** from the **Edit** menu.

Trim

The Trim command removes everything *outside* the selection. The selection is not affected. Use this command to keep part of a sound and discard everything else. This command is frequently used after recording to trim any leading or trailing silence. Note that if only one channel of a stereo sound is trimmed, the end of that channel is padded with silence. As an alternative, you can use the Copy To command to save the selection to a separate file or use the AutoTrim command to automatically trim silences.

To trim:

1. Select the part of the sound you want to keep.
2. Choose **Trim** from the **Edit** menu.

AutoTrim

The AutoTrim command removes leading and trailing silences from the ends of the selection. Unlike the Trim command, this command works only on the selection and does not remove silences outside the selection.

`Silence to keep` controls how much silence to leave on the ends. Use zero to remove all silence. Extra silence is **not** added if the existing silence is less than this amount. The

`Threshold` specifies the minimum, sustained level of audio that should not be considered silence. If the audio has a lot of background noise or hiss, use values above -30dB. If the audio is clean, values below -40dB.

Leading silence, trailing silence, or both will be removed depending on the `Trim` option you select.

Note that if only the left or right channel is selected in a stereo sound, then only that channel is trimmed. Since it is not possible for one channel to be longer than the other, the end of the trimmed channel is padded with silence, which results in trailing silence.

To trim silences from the ends of the entire file:

1. Select the entire file by using the Edit | Select All command.
2. Choose **AutoTrim** from the **Edit** menu.
3. Enter the length of silence to keep and the threshold for the silence.
4. Choose OK.

Mute

The Mute command replaces the selection with silence. Unlike Delete or Cut, the length of the sound is not changed. Use this to remove offensive language from music without interrupting the overall beat.

Insert Silence

The Insert Silence command inserts some blank space in the sound at the start marker's position. Use this command to make room for recording or to insert a delay. You are prompted to specify the duration of the silence.

To insert 1 minute of silence:

1. Move the start marker to the place where you want to insert the silence. See Editing Overview.
2. Choose **Insert Silence** from the **Edit** menu or click the **Silen** button.
3. Enter **1:00** for the duration.
4. Choose **OK**.

Select View

Use Select View to select all of the sound currently shown in the Sound window's graph. The start and finish markers are moved to the far left and far right of the view.

Select All

Use Select All to select the entire sound. The start and finish markers are moved to the beginning and end of the sound.

Channel

The Channel submenu sets which channel of a stereo sound to use or modify for editing or effects. Use this feature to copy a single channel from a stereo sound or apply an effect to only one channel. The currently selected channel is shown in the status bar. When recording or using effects such as Resample, Playback Rate, or stereo effects, the channel setting does not apply and both the left and right channels are modified.

You can right-click on the channel item in the status bar to quickly select a channel or use Options | Tool Bar to add a "Channel" toggle button to the tool bar.

Marker

The Marker submenu lists commands for changing the positions of the start and finish markers for adjusting the selection.

Set

Sets the start and finish markers to an exact time or sample position. To specify a time, choose the `Time` option and enter the time in hours, minutes, seconds, and thousandths of a second. For example, you could enter 1:04:27.873. To specify a sample position, choose the `Sample` option and enter the position.

If you want the length of the selection to be aligned to a CD sector or 1 kilobyte, select the appropriate option. When the OK button is pressed, the finish marker is adjusted to align the selection length.

Previous

Moves the start and finish markers to their previous positions. The last five positions are stored automatically whenever the markers are moved. Using this command repeatedly sets the selection back to each of those five positions. Note that *absolute positions* are stored, so any modifications, such as deleting part of the sound will not be factored into the previous positions. In such situations the previous positions may not select the same part of the sound that was previously selected (that part may be deleted).

Drop Start/Finish

When playing or recording a file, you can drop the start or finish marker at the current position. You can use the bracket keys, **[** and **]** to drop the start and finish markers respectively. Note that the start marker cannot be dropped after the finish marker.

Recall Selection Positions

Moves the start and finish markers to previously stored positions. These positions are set using the Store Selection Positions command. Hold the Ctrl key to recall the positions stored within a different Sound window.

Store Selection Positions

Saves the current positions of the start and finish markers. Use the Recall Selection Positions command to move the markers back to these positions. Hold the Ctrl key to store the positions so they can be recalled in a different Sound window.

Snap To Zero-Crossing

When editing, it is important that the waveform not change suddenly from one amplitude to the next, otherwise a click will occur. This can happen when deleting the selection. The amplitude of the waveform at the start marker may be completely different from the amplitude at finish marker. After deleting the selection, these two different amplitude will be right next to each other, causing a click.

The "Snap To Zero-Crossing" feature helps to minimize the problem by making sure that the markers are always near zero amplitude samples. When you drag and release a marker, it is automatically moved to a position where the amplitude approaches zero. This means that when you delete the selection, the amplitudes at both the start and finish markers will be more closely matched (near zero).

Since stereo sounds can have very different left and right channels, it is not always possible to find an ideal zero-crossing position. However, you can use the Edit | Channel menu to limit the snap feature to a single channel.

Note that if you zoom in close enough so that the true shape of the waveform is shown (such as View | Zoom 1:1), the snap feature is automatically turned off so that markers can be placed at any position.

Cue Point

This submenu lists commands for working with cue points.

Edit Cue Points

Displays the Cue Points tool.

Drop Cue

Drops a cue point at the current playback or recording position. If neither recording nor playback is active, then the cue is placed at the start marker. The name of the cue point is set to the time position by default, but that can be changed by the Auto Cue settings under the Cue Points tool.

Jump To Next Cue

Moves the start marker to the next cue point or to the end of the file.

Jump To Previous Cue

Moves the start marker to the previous cue point or to the beginning of the file.

Split File

Divides the file into smaller sections using cue points as split points. See Split File Button under the Cue Points tool for more information.

Show Cue Lines

Displays vertical cue lines on the Sound window graph. This setting applies to each Sound window individually, so lines can be shown in one window and not in another. When a new window is opened, the last setting selected is used.

Effect Menu Commands

This section explains commands under the GoldWave's Effect menu. Please see the Effects Overview section for general information about how GoldWave employs effects.

Censor

Censor replaces the selection with a tone (beep), static, clipboard audio, or gibberish. Use this effect to cover up profanity, cleanly overwrite dialogue with other dialogue, or make dialogue almost unintelligible.

Censor type
Specifies what sound to use to replace the selection. Most of the censor types are simple tones or noises. `Clipboard` and `Gibberish` are special types. `Clipboard` replaces the selection with audio currently in the clipboard. Use the Edit | Copy command to copy audio to the clipboard before using the Censor effect. `Gibberish` scrambles the selection by moving and reversing short sections of audio, making dialogue difficult to understand.

`Censor volume` controls the volume of the censored audio (the tone, clipboard, or gibberish). Usually this is set near full volume.

`Source volume` controls the volume of the original audio. Usually this is set to the lowest value (Off).

`Crossfade time` controls how long it takes to fade from the original audio to the censored audio. This eliminates sudden transitions (possible clicks) between original to censored regions. Usually this is set to a small value.

Examples
To censor profanity with a beep:

1. Select the word by itself making sure no surrounding dialogue is selected.
2. Display the Censor effect window (**Effect | Censor**).
3. Select the "Beep" preset.
4. Choose OK.

To replace dialogue:

1. Open the file containing the final dialogue.
2. Use Edit | Copy to copy it to the clipboard.
3. Open the file containing the dialogue to be replaced.
4. Select only the dialogue to be replaced. This must be the same length as the copied audio.
5. Display the Censor effect window (**Effect | Censor**).
6. Select the "Overwrite with clipboard" preset.
7. Choose OK.

The copied dialogue must be exactly the same length as the replaced dialogue for it to be completely overwritten. If the clipboard audio is shorter, then the trailing end of the replaced dialogue will still be present. If the clipboard audio is longer, it will be truncated to fit in the selection. This effect does not alter the length of the selection. The timing of the surrounding audio is not affected, which maintains synchronization in music or video.

See Also
Mute
Replace
Overwrite

Doppler

A Doppler effect is defined as a change in frequency of a wave caused by motion. You often hear it when police or ambulance sirens drop in pitch as they pass near by.

In GoldWave the Doppler effect dynamically alters or bends the pitch of the selection. It does this by altering the speed at which the waveform is played. Note that both pitch and tempo are changed. Shape Controls are presented where the pitch/speed can be varied over the selection from 0.25 to 2.0 times normal. You can use
Effects | Volume | Shape Volume to dynamically alter the volume as well.

The "Slow down" preset gives you a good idea of what it sounds like when the batteries start to fail in a portable tape player. Other presets can change your voice to a smurf ("Faster") or a giant ("Slower").

Dynamics

Dynamics alters the amplitude mapping of the selection. It can limit, compress, or expand a range of amplitudes. The amplitude mapping is set using Shape Controls, where x-axis and y-axis both have a range of -1 to 1. When the line stretches diagonally from the lower left corner to the upper right corner, the input amplitude (x) and output amplitude (y) are the same for every point on the line. By changing the line, the output will differ from the input.

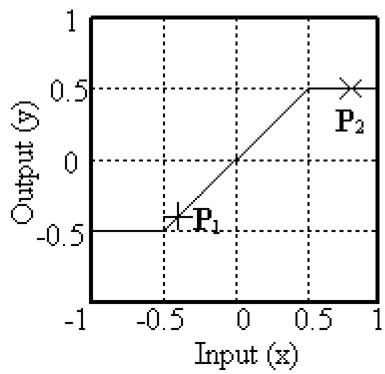


Figure: Dynamics

The figure shows an example of amplitude mapping for clipping distortion. Point $\mathbf{P_1}$ has an input value of -0.4 and an output value of -0.4. Therefore no change occurs to the amplitude. Point $\mathbf{P_2}$ on the other hand, has an input value of 0.8 and an output value of 0.5. In this example, all input amplitudes in the range of -0.5 to 0.5 remain unchanged. Any values outside this range will be limited to ±0.5, so that the final sound will have no amplitude magnitudes greater than 0.5. Essentially, any values that are too high are "clipped" to fit within the range.

In practical terms, dynamics can increase the volume of quiet sections of a sound without greatly increasing the loud sections as well. It can introduce mild or heavy distortion effects (such as the "Blare" or "Hiss noise" preset).

See Also
Compressor/Expander

Echo

Echo produces an echo or reverb effect in the selection. The parameters include number of echoes, echo delay, echo volume, feedback, stereo, and tail generation.

`Echoes` specifies the number of times the sound is repeated and mixed at a diminished volume with the given delay.

The `Delay` is the amount of time it takes for the first echo to be heard. If more than one echo is used, then each subsequent echo delay is compounded by this amount again. In other words, the second echo is delayed twice as long, the third echo is delayed three times as long, and so on.

The Volume controls how loud the first echo is. The volume of subsequent echoes is compounded in the same way the delay is.

`Feedback` makes the echo sound deeper and richer. It regenerates each echo, and the echo of those echoes, and so on, creating many more echoes than before.

The `Stereo` option makes the echo bounce between the left and right channel of a stereo sound.

The `Generate tail` option adds some silence to the end of the selection so that the trailing, fading echoes can be stored. This increases the length of the sound. Turn this setting off if you do not want to change the length of the selection or have any silence inserted. If the option is off, echoes will end abruptly rather than trailing off gradually.

Compressor/Expander

The Compressor/Expander effect dynamically alters the audio volume level. It is commonly used as a compressor, limiter, expander, or gate.

This effect does not change the size of the file. For file compression, use File | Save As and change the type and attributes.

Compressors and limiters are used to decrease or limit the dynamic range of audio. In plain English, they reduce the volume of loud sounds while leaving the rest of the sound unchanged. Expanders and gates are used to increase the dynamic range of audio. They reduce or eliminate quiet sections while leaving louder sections unchanged, which can help to reduce background noise.

Compressors always work on loud sections and expanders always work on quiet sections. Normally, both compressors and expanders only reduce the volume. However, GoldWave also allows you to boost the volume.

The `Multiplier` specifies the amount of volume change. It is the scale factor applied to the sound when the threshold is met. For compressors, it is the amount to scale the loud sections. For expanders, it is the amount to scale the quiet sections. Normally a value less than 0dB should be used to reduce the volume.

The `Threshold` specifies the audio level to activate the expander or compressor. Compressors scale the volume level of all sounds above that level. Expanders scale the volume level of all sounds below that level.

The `Attack` and `Release` times specify how quickly the expander/compressor is activated. An attack value of 0.100 means that the audio level will have to cross the threshold for at least one tenth of a second before the multiplier is used at full strength. A release value of 0.100 means the multiplier will continue to be used for one tenth of a second after the audio level no longer crosses the threshold.

Use the `Expander` or `Compressor` mode to specify what processing is required.

The `Anticipate attack` setting tells the effect to scan ahead for audio that crosses the threshold. If the `Attack` time is set to 0.100, then the effect scans ahead by 0.100 seconds. This means that the multiplier will be at full force the instant the threshold is crossed rather than building to full force 0.100 seconds later. The `Use smoother` setting smoothes out any sudden volume changes which may occur during processing with small attack/release times. The `Process channels independently` setting controls how each channel is processed. When checked, each channel is processed separately and different gains may be applied to each channel. When unchecked, one gain is applied to all channels.

Compressor Example
You have recorded some music that has a few loud moments, but you want to raise the overall volume without distorting the loud parts. Select the "Reduce loud parts" preset. After compression, use the Maximize Volume effect to stretch the volume to the full dynamic range. Alternatively you could use the "Boost quiet parts" preset.

Expander Example
You have recorded someone talking and notice background noise during the quiet parts. To reduce the noise, select the "Noise gate" preset or the "Reduce quite parts" preset.

See Also
Dynamics

Filter

Filters are used to remove a range of frequencies from a sound and can produce a variety of effects. The submenu contains several filter related effects.

Auto Offset Removal

Auto Offset Removal automatically removes a vertical shift or dc offset in the waveform. Offsets occur when audio is wired through several external devices that do not share a common ground. Computers with sound hardware integrated into the motherboard can have significant offsets as well. If the waveform is constantly above or below zero during silence, then it has an offset that must be removed. Offsets can cause pops and clicks between edit points and other problems if not removed.

Unlike Effect | Offset, this effect does not need to scan the entire selection and offsets cannot be set to specific values. Values to cancel out existing offsets are automatically calculated and used. Even varying offsets are removed.

`Duration for offset calculation` controls how long to analyze the audio for an offset. For constant, stable offsets, set this value to the maximum. For fluctuating offsets, use lower values.

Bandpass/stop

Bandpass filters block all frequencies outside a specified range, keeping only frequencies within the range. Bandstop filters block all frequencies within a specified range, keeping all other frequencies outside the range.

Frequency Range
The `From` and `To` boxes specify the initial frequency range of the filter. If the Dynamic option is selected, then a final frequency range can be given in the other `From` and `To` boxes, allowing you to fade from on frequency range to another. Otherwise the initial range remains constant for all processing.

Settings
Select Bandpass if you want to keep only the frequencies within the range. Select Bandstop if you want to block all the frequencies in the range. The remaining Static, Dynamic, and Steepness options are explained under Low/Highpass filter.

Example
To remove hum and hiss from a voice recording:

1. Set `From` to 200.
2. Set `To` to 3800.
3. Choose the `Bandpass` option.
4. Choose the `Static` option.
5. Set `Steepness` to 10 or higher.
6. Choose OK.

Equalizer

Equalizers are commonly found on stereo systems. They boost or reduces certain ranges of frequencies. Simple equalizers control only treble and bass. GoldWave's equalizer controls 7 bands, as shown in the figure.

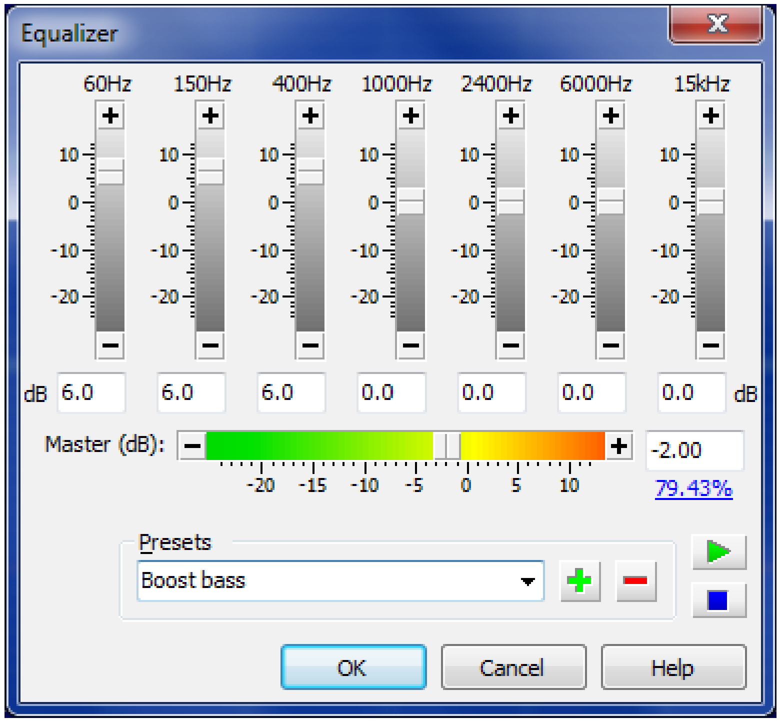


Figure: Equalizer

Center frequencies for each of the 7-bands are given at the top of each fader. Adjust the faders to boost or reduce a band by +12dB to -24dB.

To change bass, adjust the two or three left-most bands. To change treble, adjust the two or three right-most bands. Several presets are included to demonstrate bass, mid, and treble changes.

More detailed equalization is possible using the Parametric EQ, described below.

Low/Highpass

Lowpass filters block high pitched, frequencies (treble) but allow low pitched frequencies (bass) to pass. They can be used to reduce high end hiss noise or remove unwanted sounds above the given cutoff frequency. If you were to apply a lowpass filter with a cutoff frequency of 1000Hz on speech, it would make it sound mumbled and deep. Lowpass filters can also be used to eliminate aliasing noise when used before downsampling.

Highpass filters block low pitch frequencies, but allow high pitched frequencies to pass. They can remove deep rumbling hum or remove unwanted sounds below the given cutoff frequency. If you were to apply a highpass filter with a cutoff frequency of 1000Hz on speech, it would make it sound thin and hollow.

Cutoff Frequency

The `Initial cutoff` specifies the constant cutoff frequency for static filtering. If the

`Dynamic` option is selected (see below), then a final cutoff frequency can be given in the `Final cutoff` box, allowing you to fade from one cutoff to another.

Filter Options
Select `Lowpass` if you want to keep only the frequencies below the cutoff frequency. Select `Highpass` if you want to keep only the frequencies above the cutoff frequency.

If you want the cutoff frequency to remain constant throughout the selection during processing, select the `Static` option. If you want the cutoff frequency to change from the initial value to the final value, select the `Dynamic` option. Note that dynamic filtering will take more processing time.

The `Steepness` value specifies how sharply the filter cuts off frequencies outside the cutoff frequency. A higher steepness makes the filter sharper, but it also increases processing time. In technical terms, the steepness specifies the number of second order cascade filters used.

Examples
To make speech gradually become more hollow and thin:

1. Set `Initial cutoff` to 60.
2. Choose `Dynamic`.
3. Set `Final cutoff` to 1000.
4. Choose `Highpass`.
5. Choose OK.

Remove all high end hiss noise from a voice recording.

1. Set `Initial cutoff` to 4000.
2. Choose `Lowpass`.
3. Choose `Static`.
4. Set `Steepness` to 20.
5. Choose OK.

Noise Reduction

Noise Reduction uses frequency analysis techniques to remove unwanted noise, such as a background hiss, a power hum, or any continuous, consistent sound. It cannot be used to separate or remove complex or brief sounds, such as coughs, laughter and applause. It cannot remove instruments or vocals from music.

The interface, shown in the figure below, includes a frequency analysis window, with Shape and other controls. The x coordinate is the frequency in Hertz and the y coordinate is the magnitude in decibels.

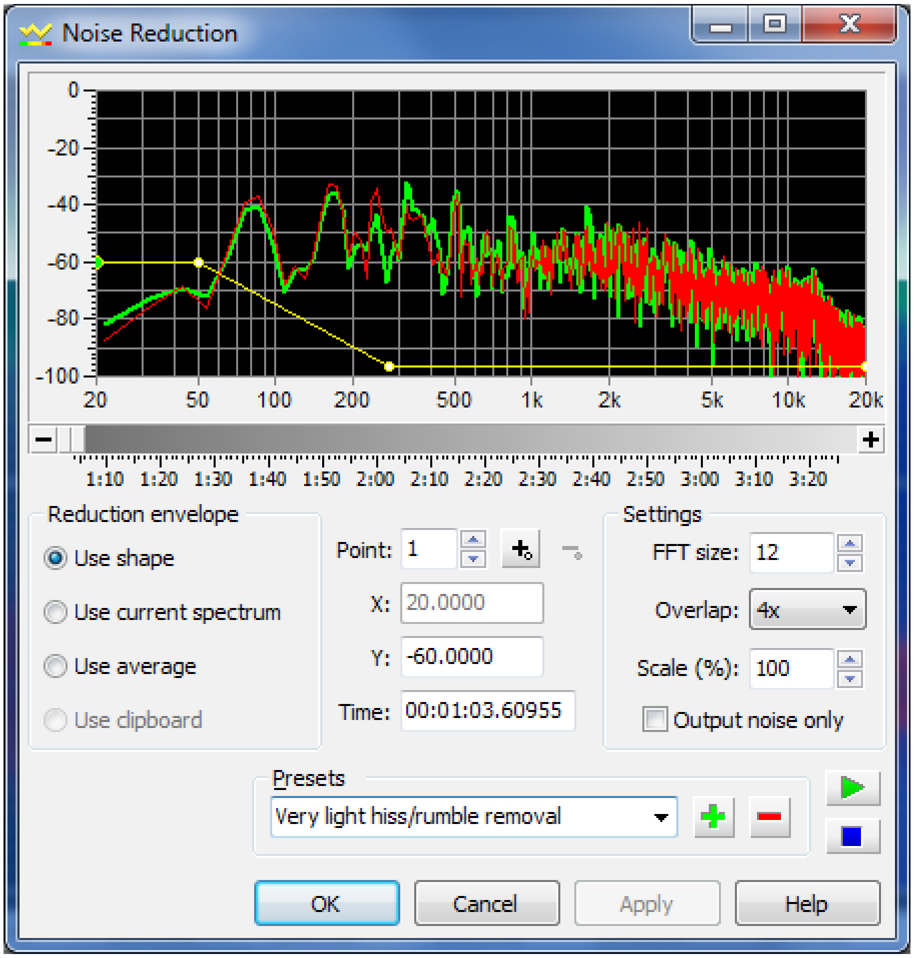


Figure: Noise Reduction

The frequency analysis provides graphical information about all frequencies within the sound at the given time. For stereo sounds, the left channel is shown in green and the right channel is shown in red. The time of the frequency analysis within the sound is given in the `Time` box. Move the scroll bar located below the analysis window to change the analysis time to show the frequency analysis of a different part of the sound. The detail of the analysis depends on the FFT size setting, explained below.

Reduction Envelope

Noise is removed using a reduction envelope. The shape of the envelope should closely match the shape of the noise you want to remove. The frequency analysis graph can help determine that shape. Change the analysis time so that it coincides with a time in the sound where only the noise is heard (use the preview button to play the file to find such a place and time). Once you have isolated the noise in the analysis graph, you can then create the envelope. The envelope can be created in four different ways, depending on the reduction envelope setting. The "Use clipboard" option generally gives the best results. Also try the presets.

Use shape

Lets you manually create an envelope shape or select a preset shape. See Shape Controls for information about creating shapes. By creating a horizontal line at about 80dB, you can remove a hiss from a sound. In some cases, you'll need to trace the outline of the graph or draw completely different shapes to reduce the unwanted frequencies levels.

Use current spectrum
Creates an envelope based on the shape of the graph shown in the frequency analysis window. This is particularly useful for removing a complex buzz or hum. It is important that the analysis Time is set to a place where the noise is heard by itself initially.

Use average
Applies an averaging envelope throughout noise reduction processing. The envelope is continuously updated, based on the average frequency analysis of the sound. Use this setting if the noise changes frequently throughout the sound.

Use clipboard
Creates an envelope based on an analysis of the waveform in the clipboard. This is the most flexible option and usually gives the best results. Before you can use this option, you must use Edit | Copy to copy a piece of noise into the clipboard. For best results, the piece should contain only the noise you want to remove from the rest of the file. The noise can even be copied from a different file. After you copy the noise, remember to change the selection to the part of the file you want to apply the noise reduction.

Settings
The `FFT size` determines the detail of the frequency analysis and the noise reduction envelope. Usually values of 11 or higher give the best results. The `Overlap` value specifies the amount of audio to overlap from one calculation to the next. A value of 8x works well. See FFT Settings in the Effects Overview for more information. The `Scale` value lets you alter the reduction envelope scale. A value of 100 uses the envelope as it is. A value of 200 doubles the envelope, which doubles the amount audio removed from the sound. A value of 50 halves the envelope, which halves the amount removed. Normally it should be set to 100. The `Output noise only` option makes the effect perform the exact opposite of noise reduction so that only the noise remains in the output. This is useful when previewing the effect to hear what is actually being removed from the audio. Do not check this box when removing noise.

Parametric EQ

A parametric equalizer (shown below) is a flexible tool for reducing or enhancing ranges of frequencies. GoldWave presents an easy to use interface where all the parameters for up to 40 bands can be configured quickly. The presets contain some commonly used templates.

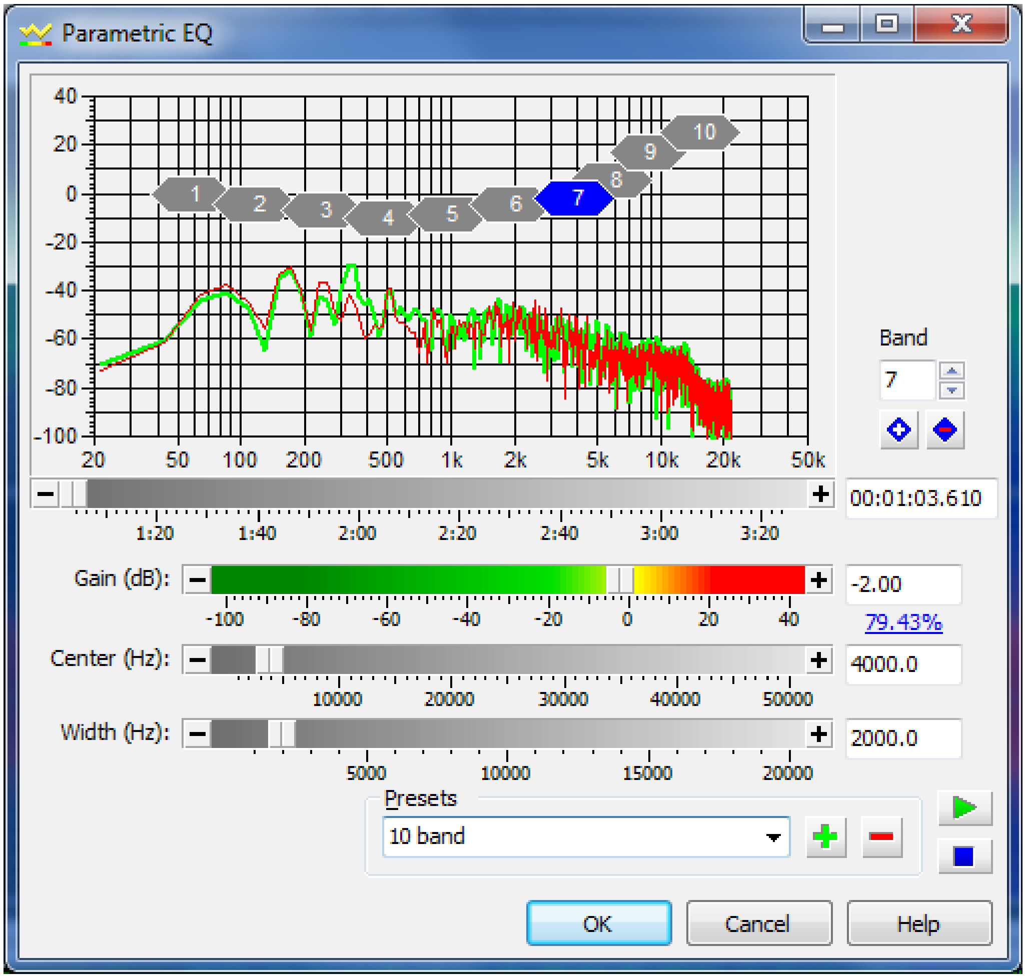


Figure: Parametric Equalizer

Graph window
The graph shows frequency on the x-axis in Hertz and the gain on the y-axis in decibels. Each enabled band is displayed in the graph as a diamond shaped box located at its center frequency and gain. The width of the box shows the bandwidth. The currently selected band is shown in blue and its exact settings are given in edit box controls.

A short time frequency analysis graph is drawn with the left channel in green and the right channel in red. The time of the analysis can be changed using the scroll bar located at the bottom of the graph. The analysis helps determine what frequencies to boost or reduce. A high pitched squeal, for example, would appear as a spike near the right side of the graph. Whereas a low pitched hum would be a spike or bump near the left side.

Controls
A band is configured by selecting its number from the Band box and adjusting the Gain, Width, and Center faders. A quicker method is to *drag-and-drop* the band to a new location on the graph. Note that because of the logarithmic frequency scale, the width of a diamond changes as you move it left or right. The bandwidth, however, remains constant.

Use the diamond plus button to add more bands. Use the diamond minus button remove existing bands. The current (blue) band given in the Band box is the one removed.

The "Notch" preset is effective for removing a simple tone from a sound, such as a 60Hz hum or telephone dial tones. The "Bass boost" and "Treble boost" presets work the same way as the bass and treble controls on a stereo system. Adjust the gain up or down to control them.

Pop/Click

A pop/click filter is a specially designed filter that searches for abrupt changes in the sound and eliminates them. Such a filter is often used to remove pops and clicks caused by dust and scratches when recording old vinyl records.

The `Tolerance` defines how abrupt a change can be before it is considered a click. It is best to start with a value near 1000% or higher. Using a lower value will detect more clicks, but may eliminate natural clicks such as drum sticks tapping together or a conductor tapping the baton. Values less than 500% should be used on short selections only.

When a click is detected, the filter attempts to reconstruct the damaged waveform based on the surrounding waveform shape making the repair almost imperceptible. However with excessive pops and clicks or at low tolerance levels, reconstructed waveforms may overlap and sound distorted. The tolerance setting should be kept as high as possible. Using a very low setting may introduce more distortion than existed in the original. This is most noticeable in voice recordings and instrument solos, particularly trumpet solos. Always start with the maximum tolerance setting for those types of sounds.

The filter requires a minimum selection of 4000 samples (about one tenth of a second at CD quality) to establish a baseline. Using the filter on a shorter selection has no effect.

Silence Reduction

This filter automatically removes silences from a sound. Use it to save storage space or to remove long pauses in speeches or police/airport radio recordings.

The `Silence threshold` specifies the volume level for the silence. Any audio below this level is considered silence and is subject to removal, provided it has a long enough duration. The `Duration` specifies how long the silence must be before it is reduced. Any silences short than this remain unchanged.

`Reduce to` is a percentage that specifies the length of the reduced silence relative to its original length. A value of 75, for example, reduces a 10 second silence to 7.5 seconds.

`Maximum length` specifies the maximum length for reduced silences. This setting overrides the `Reduce to` setting if the reduced length still exceeds the maximum. If this is set to 5 seconds in the above example, then the 10 second silence is reduced to 5 seconds.

If `Full crossfade` is checked, then the ends of the audio where silences were removed are gradually crossfaded over the entire length the remaining silence. If unchecked, then a short crossfade of one-tenth of a second is used to join together non-silent sections where silences were removed. If there is a high level of background noise that varies, then full crossfade is recommended. However, if you hear unexpected overlapping fragments after removing silences, then this option should not be used.

Smoother

Use this filter to reduce hiss and crackle. `Length` sets the length of the smoother filter. The larger the value, the more averaging is applied to the audio and the duller it will sound. Use the `Volume` setting to help offset the loss in volume for larger Length values.

Spectrum Filter

A spectrum filter is a general purpose audio filter similar to a parametric equalizer, but with much greater control. Instead of using individual bands, the entire frequency spectrum is controlled using a shape line that controls the gain. This allows many kinds of filters to be designed, such as lowpass, highpass, bandpass, bandstop, notch, peak, comb, and more. Filtering is performed in the frequency domain using Fast Fourier Transforms (FFTs).

A spectral analysis window is displayed with a shape line and several other controls. The `X` and `Y` coordinates are updated when you click-and-drag a shape point. The `X` coordinate is the frequency in Hertz and the `Y` coordinate is the magnitude in decibels. The time of the spectral analysis shown is given in the `Time` box. If you move the time scroll bar, located below the analysis window, the graph changes to show the spectral analysis of a different part of the sound. The `Master gain` controls the overall gain of the filter, which is equivalent to shifting the entire shape up or down.

Initially the shape line is horizontal at 0dB, which means that no changes in gain are made at any frequency. Alter the shape line up or down to increase or decrease the gain at a particular frequency. In technical terms, the shape line represents the frequency response function.

Graph Range

`Min` and `Min` control the range of values for the y-axis in decibels. Setting `Min` to -5 and `Max` to 5, for example, sets the graph to show a narrower range between -5dB to 5dB. That allows shape points to be set more precisely with the mouse within that range.

Settings

The `FFT size` determines the detail of the spectral analysis and the resolution of the filter. Higher settings provide a higher resolution, allowing the filter to follow the shape more accurately, with sharper cutoffs. When processing high sampling rate files, such as 88kHz or 192kHz, the `FFT size` must be set higher for the filter to follow the shape. Using too high a value may cause overshoot and oscillations in the gain. The `Overlap` value specifies the amount of audio data to overlap from one calculation to the next. The lowest value gives the fastest processing and generally works well. See FFT Settings in the Effects Overview for more information.

Flanger

A flanger effect is similar to an echo effect in that the original sound is mixed with a delayed copy of itself. Unlike an echo, where the delay is constant, a flanger varies the delay over a given range. The speed, or *frequency*, at which the delay varies is controlled as well. The Flanger effect presents a window where you can set the depth, frequency, and fixed delay parameters and control how the sound is mixed. Many presets are included to demonstrate the kinds of unusual audio effects that are possible.

Volumes

The `Source` volume specifies the volume of the original sound to mix with the final sound. A value of 0 means the original sound will not be mixed at all with the final sound. If this value is set to 100 and all other volumes are 0, no change is made to the sound. A value of -100 inverts the source, which is equivalent to subtracting the original instead of adding it to the final sound.

The `Flanger` volume specifies the volume of the delayed sound to mix with the final sound. Usually, this value should be in the range of 25 to 100, or -25 to -100 for an inverted mix.

`Feedback` specifies the level of feedback (previously generated output) to mix with the final sound. This makes the effect more full and pronounced. Set this value to 0 if you do not want any feedback. In general the feedback should be set to between -75 to 75.

The `Stereo` option allows the flanger and feedback audio to be bounced between the left and right channels of a stereo sound, giving a more pronounced stereo effect.

Frequency modulator / vibrato

`Variable delay` specifies the maximum variable delay in milliseconds. A value of 40 will allow the delay to vary from 0 to 40 milliseconds.

`Frequency` specifies how fast to vary the delay. A value of 2 will vary the delay over from 0 to its maximum twice a second. For a value of 0.2, the maximum delay is reached every five seconds.

The `Fixed delay` is added to the depth to change the minimum delay. If the variable delay is 40 and the fixed delay is 10, the delay will vary from 10 to 50 milliseconds.

The `Sine modulator` and `Triangular modulator` settings control how the delay is varied. If sine is selected, then the delay varies on a sinusoidal pattern. If triangular is selected, then the delay varies in a simple linear pattern.

Volume modulator / tremolo

`Depth` specifies how much to vary the flanger volume. If the flanger volume above is 80%, for example, a depth of 100% varies the volume from 0% to 80% and a depth of 25% varies it from to 60% to 80%.

`Frequency` specifies how fast to vary the volume. A value of 2 will vary the volume over its depth twice a second. For a value of 0.2, the full depth is reached every 5 seconds.

`Sine modulator` varies the volume based on a sine wave.

`Triangle modulator` varies the volume based on a linear triangular wave.

Interpolate

Interpolate (see figure below) uses linear interpolation to smooth out the waveform within the selection. Use this command on a tiny selection to remove a pop or click. This command should not be used on a large selection.

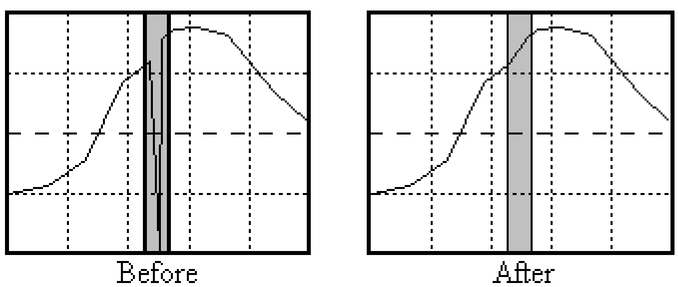


Figure: Interpolate

Invert

Invert reflects the selection about the time (horizontal) axis. The selection is essentially turned upside-down. This produces no noticeable effect in mono sounds and has a slight effect in stereo sounds. Inverting a single channel of a stereo sound produces an "in" or "out" effect.

Inverting can be used before mixing so that the two sounds are subtracted instead of added.

Mechanize

Mechanize gives a robotic or mechanical characteristic to a sound through a method known as amplitude modulation. This effect was widely used in old science fiction movies. The rate of modulation is controlled with the `Frequency` value. The sound is modulated with one of several waveforms given in the table.

Table: Mechanize Modulator Options

Waveform	**Purpose**
Sine	The sound is modulated with a sinusoid, which tends to shift the pitch and cause distortion. Using a very small frequency value (less than 2.0) causes the sound to fade in and out.
Triangle	The sound is modulated with a triangular wave. This is similar to the Sine modulator but causes more distortion at higher frequencies.
Square	The sound is modulated with a raised square wave, with amplitudes ranging from 0.0 to 1.0. This causes very heavy distortion at high frequencies and intermittent (on/off) audio at low frequencies.
Clipboard	The sound is modulated with audio stored in the clipboard. Audio must be copied to the clipboard before using this option. No attempt is made to convert the clipboard audio to a compatible sampling rate.

Offset

The Offset effect adjusts or removes a dc offset in the selection by shifting it up or down so that the wave is centered on the horizontal axis.

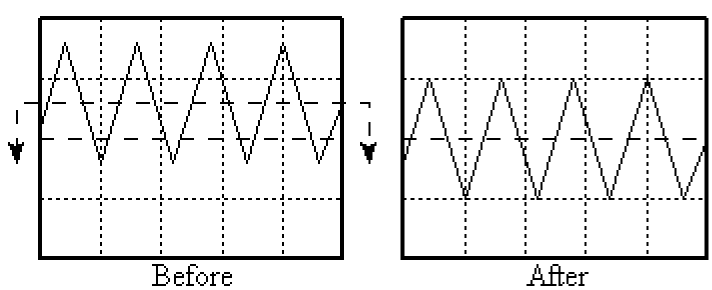


Figure: Offset

Positive values shift the graph up and a negative values shift it down. Use the Scan Offset button to automatically determine what offset to use to remove any existing offset. After scanning completes, the `Left` and `Right` values are set such that they will cancel out the offset in the given channel. If the values are zero, then no offset was detected.

Any offset should be removed to minimize pops/clicks during editing. Offsets may interfere with other effects as well.

You should check the offset from time to time after processing effects. Otherwise, the offset may increase with each effect, resulting in some distortion.

Pitch

Pitch changes the pitch (frequency) of the selection. This is useful for converting instrument samples from one note to another. The new pitch is specified using a scale factor or using semitone and fine tune values.

Scale
This option scales the pitch by a percentage value. If you set the scale to 50, that is equivalent to a downward shift by one octave. A value of 200 is the same as an upward shift of one octave and would make a voice sound like a chipmunk. A value of 75 would make a woman's voice sound like a man's.

Semitone
This option changes the pitch by semitones (notes on a piano). If your sound is a note at middle C and the semitone value is 2, the note is changed to D. A value of -1 changes the note to B. A value of 12 make the note one octave above middle C. The Fine tune value lets you make a slight pitch adjustment in hundredths of a semitone. For example, a value of 50 changes a note from C to halfway between C and C#.

Preserve length
If this option is checked, a complex algorithm is used to keep the length of the original note the same as the new note. In other words, the tempo will not be changed. In terms of a voice recording, this changes the pitch of the voice without changing the speed at which the words are spoken. This option requires a substantial amount of processing time and will affect the quality of the sound.

For information on the `FFT size` and `Overlap`, see FFT Settings in the Effects Overview.

Plug-in

This menu lists all plug-ins detected by GoldWave. If any compatible DirectX plug-ins are detected, then they are listed under a DirectX submenu.

Reverse

This command reverses the selection so that it plays backward. Now you have an easy way to capture all those "satanic verses" or reverse speech messages. You can play a sound backwards by using the rewind button on the Control window as well.

Reverb

Reverb adds body and richness to the sound by simulating the acoustic reverberation and echoes within a chamber or room.

The `Reverb time` sets the size of the reverb. A longer time simulates a larger chamber or room.

The `Volume` controls how loud the reverb is. Values less than -18dB give good results.

The `Delay scale` alters the delay of the reverb for fine tuning. Use 1.0 for a standard reverb.

Stereo

The Stereo submenu contains commands that apply to stereo files, such as swapping channels and left/right panning.

Channel Mixer

This command swaps, mixes, inverts, or combines the left and right channels in a variety of ways. The left and right channels are replaced with a mixed combination of both channels, depending on the volume levels set. Use the controls in the `Left channel` box to set the mixing volume percentages for the left channel. Use the controls in the `Right channel` box to set the mixing volume percentages for the right channel.

To swap the left and right channels:

1. In the `Left channel` box, set the `Left volume` fader to 0 and the `Right volume` fader to 100.
2. In the `Right channel` box, set the `Left volume` fader to 100 and the `Right volume` fader to 0.
3. Choose OK.

MaxMatch

This command automatically balances the left and right channels and maximizes the volume levels. Essentially, this effect uses the Match Volume effect internally on the left and right channels, then uses the Maximize Volume effect.

After processing, the left and right channels will have the same average volume level and at least one channel will have full dynamic range (1.0 or 0dB). Note that it is rarely possible for channels to have the same average and both have full dynamic range at the same time. Typically one channel will have a dynamic range of slightly less than 1.0 or 0dB.

Pan

Pan presents the Shape Controls where left and right panning can be controlled. The graph is divided into green and red regions, representing the left and right channels respectively. The line, initially located between the regions, represents the center for panning. By bending and/or moving the line, you can dynamically alter the selection's left/right balance or pan to and from each channel. The figures below show several examples of panning shapes.

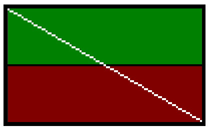


Figure: Pan from left to right

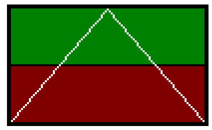


Figure: Pan from right to left and back to right

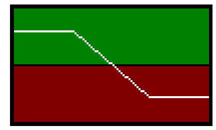


Figure: Pan from left center to right center

There are two end points on the pan shape line by default. Use the `Point` box to select the point to edit. The X and Y values control the location of the selected point. X is the time and Y is the amount of panning. A positive Y value pans to the left. A negative value pans to the right. For end points, only Y can be changed. X is fixed at the beginning and ending of the selection.

Example: Select point 1 and set Y to -1.0 to make panning start at full right. Select point 2 and set Y to 1.0 to make panning end at full left. Panning will go from right to left over the selection.

Use the `Add Point` button to insert a new point between the two end points. Note that the point numbers change, with point 3 as the end point and point 2 as the new one. Both the X and Y values can be changed for that point. To make panning go slightly left at 10 seconds into the file, for example, set X to 10.0 and Y to a positive value, such as 0.25. When adding points, be sure to select the point after which you want to insert the new point. The X value for each point is confined to adjacent points.

The `Show balance` option calculates and displays the current peak balance in yellow on the graph. For a typically stereo song, a spiked line roughly centered around zero would appear. For a 2 channel mono file, it would be a perfectly flat line at zero. For an unbalanced file, the line would be above or below zero.

The `Change volume only` setting limits the pan effect to volume changes only. Normally panning mixes the left and right channels to alter the balance. This setting prevents any mixing and only changes the relative volumes of the channels.

Reduce Vocals

This effect reduces vocals from certain stereo recording by subtracting the left and right channels and by using a bandstop filter. This works best when vocals are recorded exactly the same on the left and right channels and no stereo effects have been applied to the vocals. Note that any instruments record the same in both channels are reduced as well.

Usually subtracting the left and right channels destroys the stereo sound, giving mono output. However, by integrating a bandstop filter, the effect is able to restore some of the stereo, enhancing the output.

Increase the bandstop filter `Volume` and bring the `From` and `To` values closer together for increased stereo (increases the vocals). Decrease the bandstop filter volume and move the `From` and `To` values farther apart for reduced stereo (decreases the vocals).

Try the presets to learn what the different settings do.

Stereo Center

This effect uses a frequency based method to separate vocals from music or to separate the stereo center channel from the side channels and allows independent volume control of all channels.

The `Center channel` area controls the volume and frequency range of the center channel. To keep the center (vocals) and remove the side channels, set this volume to 0.0dB and the side channel volumes to -100dB (off). The result will be mono with both left and right channels containing the center audio. Use `From` and `To` to narrow the frequency range to analyze when computing the center channel. To retain bass and treble when adjusting vocals, use a range from about 200Hz to 12500Hz.

The `Side channels` area controls the volumes of the side channels. To remove the center channel and keep the side channels, set these volumes to 0.0dB and set the center channel volume to -100dB (off). The result will be stereo with the center audio removed from the left and right channels.

For information on the `FFT size` and `Overlap`, see FFT Settings in the Effects Overview.

Part of this effect uses a method by the developer of virtualdub.

Time Warp

Time Warp changes the playback speed or alters the tempo of the selection. This effect has many uses: it stretches or compresses a sound to fit in a certain time, it slows down instrumental music for easy transcription, or it changes the tempo of one musical passage to match rhythm and beats of another.

The `Change` option lets you specify a relative change in percent. A value of 50 makes the selection play at half speed. A value of 200 makes the selection play twice as fast. The `Length` option lets you specify a new length for the selection. Use this to make a sound fit a certain time, such as squeezing a 35 second commercial into a 30 second spot.

When previewing playback, the Apply button must be used to update settings. The Apply button is for previewing only.

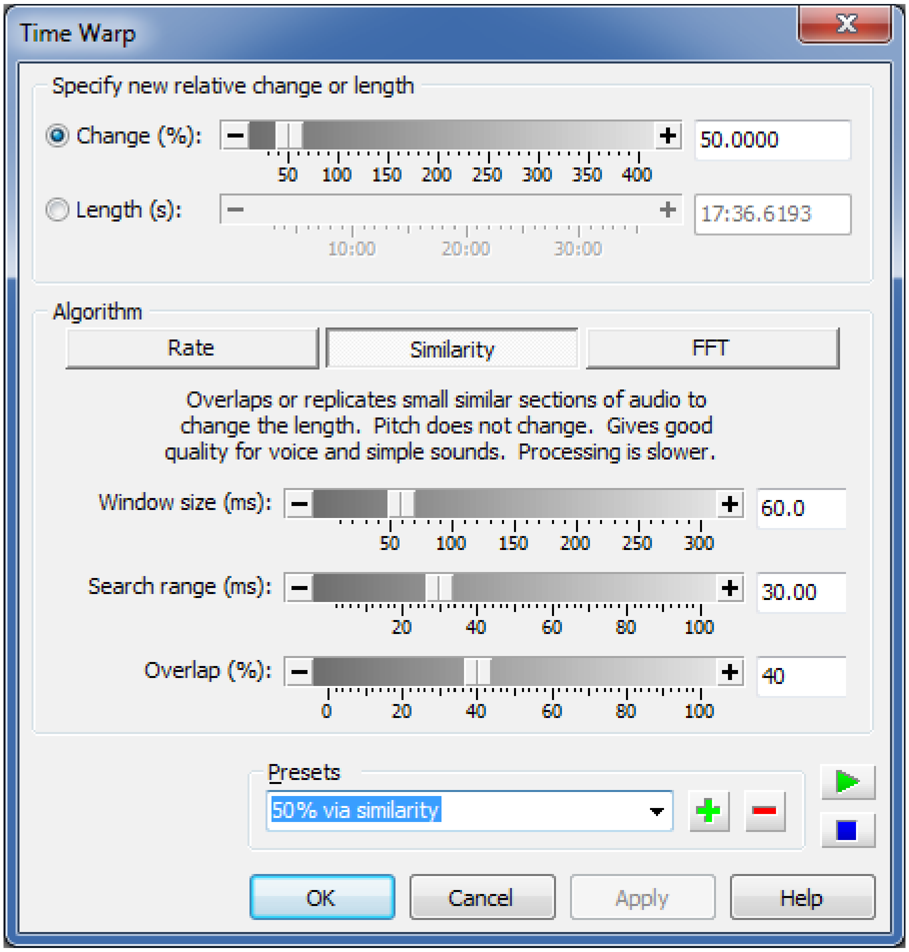


Figure: Time Warp

Three different time altering algorithms are provided, each with certain advantages and disadvantages.

Speed
The Speed algorithm simply changes the sampling rate of the entire sound so that it plays back at a different speed, similar to spinning a vinyl record faster or slower. It works the same way as the speed fader in the Control window, but in this case, the sound itself is changed. This technique is very fast and produces excellent quality, however, the pitch of the sound is changed as well. In other words, if you were to speed up a voice, the pitch becomes higher, making the voice sound like a chipmunk.

Similarity
The Similarity algorithm uses correlation to add and overlap small, similar sections of the sound. This technique preserves the pitch. It generally produces high quality voice and fair quality music when using small speed or time changes. A fair amount of time may be require for processing, depending on the Search range value. For voice, the `Window size`

should be set between 30 and 40 and the `Search range` set to between 10 and 20. For music, a larger `Window size` and `Search range` gives better results, such as 60 and 30. Larger `Search range` values increase processing time and echo. Low values increase gurgling noise. `Overlap` adjusts the amount of overlap between sections. Correlation is used to find the best matching sections to overlap, then the sections are crossfaded to reduce rough transitions from one section to the next. A value around 40% is recommended. Larger values may increase echo and significantly increase processing time. Values below 20% may cause distortion and gurgling due to shorter correlation and crossfade lengths.

FFT
The FFT algorithm uses Fourier transforms and interpolates or decimates the frequency analysis to change the length. This technique preserves the pitch, but can cause some artifacts in the sound. Best quality is obtained by using the Oscillator synthesis option, but that requires significant processing time. The `FFT size` should be set from 11 to 12 and the `Overlap` may be set to 4x for fastest processing, but 8x or 16x may give better quality. The `FFT size` and `Overlap` settings are explained in the Effects Overview section.

If you changed the speed fader in the Control window, remember to set it back to 1.00 so that the device plays at the correct speed.

Voice Over

Voice Over mixes foreground audio contained in the clipboard with background audio contained in the file. This effect automatically fades the background audio in and out based on silent and non-silent regions within the foreground audio.

Use this effect to:

- Mix a voice recording or narration over background music so that the music automatically becomes quieter when the voice is speaking.
- Mix a translation over foreign dialogue so that the foreign dialogue can still be heard in the background.
- Automatically fade between two audio sources.

This effect cannot be used until audio is copied to the clipboard. Follow these general steps:

1. Open the foreground file or voice file.
2. Make adjustments for best results.
3. Use the Edit | Copy command.
4. Open the background or music file.
5. Set the start marker where you want the voice over to begin.
6. Display the Voice Over effect window (**Effect | Voice Over**).
7. Adjust settings or select a preset and choose OK.

The Voice Over window shows a graph of the current fade and mix settings. The clipboard (foreground or voice) audio is shown in red, the original file (background or music) audio is shown in blue, and a time line is shown under the graph. By default, background audio is faded out, which is shown by the blue region sloping downward. The steepness of the slope is controlled by the `Fade out time` setting. How low the background volume goes is controlled by the `Fade volume` setting.

Next the foreground audio is mixed in, which is shown by the red blocks. The volume (height) of the foreground region is controlled by the `Voice volume` setting. The space between the two red blocks represents the amount of silence allowed in the foreground audio before the background fades in again. It is controlled by the `Silence allowed` setting.

Finally, when the foreground audio ends or contains sufficient silence, the background audio fades in again, as shown by the blue region sloping upward. The steepness of the slope is controlled by the `Fade in time` setting. The level at which the foreground audio is considered silence is controlled by the `Silence level` setting and is shown as a yellow line in the graph.

If you use the play button to start previewing, the graph displays a real-time representation of the faded and mixed audio.

`Fade out time` controls how quickly the background audio fades out before the foreground audio is mixed in.

`Fade in time` controls how quickly the background audio fades in when the foreground audio ends or contains a long enough span of silence.

`Fade volume` controls how quiet the background audio becomes when it is faded out. Set it to the lowest value (Off) if no background audio is required when foreground audio is present. Usually values under -20dB are recommended when background audio is needed.

`Voice volume` controls the volume of the foreground audio. Values around -2dB to 0dB are recommended.

`Silence level` controls the level at which the foreground audio should be treated as silence. Recordings often contain some amount of background noise. To ensure proper detection of silence and accurate automatic fading, use values of -34dB or higher. Using the lowest (Off) value is not recommended.

`Silence allowed` controls the amount of silence allowed in the foreground audio before the background audio is faded in. Usually narrations contain brief pauses where fading in is unnecessary and undesired. Adjust this setting to avoid fading in and out too frequently.

Note that this value is added to the fade in and out times to determine the total amount of silence required before the background audio starts to fade in. If `Fade out time` is 1.0, `Fade in time` is 1.0, and `Silence allowed` is 3.0, then the foreground audio would have to be silent for a total of 5.0 seconds before the background audio is faded in.

Notes
Before copying the foreground audio to the clipboard, be sure to:

- Trim off any leading and trailing silences.
- Maximize the volume (use Effect | Volume | Maximize) to ensure the volume level is optimal.
- Remove any offset with the Effect | Filter | Auto Offset Removal command. A large offset may affect silence detection.
- Make sure the sampling rate (shown in the status bar) matches the sampling rate of the background file. If it does not, use the Effect | Resample command to change it.

Volume

The Volume submenu contains several volume related commands. Volumes are usually specified in decibels (dB) or by a percentage (%) of the sound's original amplitude. For more information about volumes, see Volume Scales in Appendix A. Unlike the volume fader in the Control window, which changes the device playback volume, these commands scale the sound's data to change the volume.

Auto Gain

Auto Gain evens out the volume to a consistent level across the selection. When recording a speech, interview, or telephone conversation, for example, the volume tends to vary depending on the position of the microphone relative to the person speaking. If the person or microphone moves around, the volume fluctuates. With telephone recordings, one person often sounds louder than the other.

Auto Gain automatically varies the volume level to increase it when it is low and decrease it when it is high (but it cannot correct overloaded or clipped audio).

The primary setting is the `Target volume`. It controls the final volume of the audio. The other settings control how often the volume changes (`Update interval`), how quickly it changes from one value to another (`Attack/release`), the maximum increase allowed (`Maximum gain`), and the level at which no gain should be applied to avoid amplifying background noise (`Silence level`).

`Update interval` controls how often the volume is adjusted to match changes in the audio. Values under one second give the best results. Use a smaller value if the audio level varies quickly. Use a larger value if the audio level is mostly even already, but needs occasional adjustments.

`Attack/release` controls how quickly the volume is changed from one interval to the next. Larger values smooth out volume changes so that they are more gradual. Setting this value to zero applies the new volume instantly every update interval. Usually values less than the `Update interval` value work best.

Note that when using a non-zero setting, peaks may exceed the target volume briefly as the volume gradually changes from a higher level to a lower one.

`Target volume` controls the final peak level of the audio. The volume is increased or decreased so that the peak level always hovers around this value. A value close to or slightly below 0dB (100%) is recommend for maximum volume. Values less than 0dB can be used to limit or clamp the peak level to a certain volume. The "Peak reducer" preset shows how peaks can be reduced to 90% without affecting the average volume.

`Maximum gain` controls the maximum amount the volume can be increased. Audio containing many noisy silences can result in sudden bursts of noise if the `Silence level` setting is not set high enough. By limiting the maximum gain, explosive volume increases for quiet sections are reduced.

`Silence level` controls the level of noise to be considered as silence. Any audio below this level is ignored and not adjusted. In other words, Auto Gain is turned off while the audio level is below this threshold. Care must be taken when setting this value. Setting it too low greatly amplifies background noise. Set it as high as possible while still getting good gain results.

Notes
One drawback of the Auto Gain effect is that background noise is amplified along with the foreground audio. In recordings where the background noise is consistent, but the foreground audio varies, the end result will be consistent foreground audio with varying background noise. Use the Noise Reduction or other Filter effects in GoldWave before using Auto Gain to reduce or eliminate that problem.

For stereo files, the left and right channels are processed independently. While this ensures that the channels are balanced, it may cause some unbalanced audio for short update intervals. In such cases where stereo audio is not needed, it is best to convert the file to mono or use the Channel Mixer stereo effect in GoldWave to mix the channels into mono before using Auto Gain.

Change Volume

This command modifies the selection so that it sounds louder or quieter. You need to enter the new relative volume. Values less than 0dB make the selection quieter. Values greater than 0dB make it louder. A value of 0dB leaves the volume unchanged.

If you are trying to make the volumes of several different songs sound the same, use Effect | Volume | Match Volume instead.

Fade In

Fade In gradually increases the volume throughout the selection. You need to specify the initial volume. Normally to fade in from complete silence, you'd use the lowest possible value of −160dB.

The Logarithmic and Linear options control the shape of the fade. A Logarithmic fade will fade in more rapidly than a Linear fade.

To fade in the first 5 seconds of a sound from silence:

1. Select the first 5 seconds of the sound.
2. Choose **Fade In** from the **Effect | Volume** submenu.
3. Enter −160 for the `Initial volume`.
4. Choose OK.

Fade Out

Fade Out gradually decreases the volume throughout the selection. You need to specify the final volume. A value of −160dB fades to complete silence.

The Logarithmic and Linear options control the shape of the fade. A Logarithmic fade will fade out more rapidly than a Linear fade.

To completely fade out the last 5 seconds of a sound:

1. Select the last 5 seconds of the sound.
2. Choose **Fade Out** from the **Effect | Volume** submenu.
3. Enter −160 for the `Final volume`.
4. Choose OK.

Match Volume

The Match Volume effect makes the volumes of separate files seem similar. When creating a CD, you often find that each songs is recorded at a different volume level. This means you have to adjust the CD Player volume from one song to the next. By using the Match Volume effect, you can adjust the volume levels of each song so they all have the same average level, eliminating the need to adjust the volume for each song later.

Volume changes are based on a root-mean-square (rms) average. The rms average is calculated with silent regions (below -44dB) excluded. Files with similar average levels will seem to have similar overall volume levels.

Use this effect on each file to set the `Average` to the same value. Use the File | Batch Processing command to apply this effect to a group of files before burning them to a CD. The `Clipping control` options are explained in the table below.

If you want to match the left and right channel levels of a single file, use the Effect | Stereo | MaxMatch effect instead.

The average value to use depends on the files. You should open each file and display the Match Volume effect to see what average value it has, then apply an overall average value to all the files. To avoid clipping distortion, it is best to use the minimum average across all files. For example, if one file has an average of -20dB and all the other files have a higher average, such as -18dB, then use -20dB for all files.

Unlike the Maximize Volume effect, the Match Volume effect may result in clipping distortion if the average level is set too high. The `Final peak` area displays the resulting final peak level after processing. If the peak exceeds 0dB, the value is shown in red as a warning that clipping may occur. Use a lower average to avoid clipping.

This effect should not be used with Maximize Volume. Use one or the other, but not both (one cancels the other).

Table: Clipping Control

Option	**Purpose**
Allow clipping	Uses the given average level, regardless if clipping is required.
Reduce average level to avoid clipping	Reduces the average level just before processing begins to ensure that the audio will not be clipped. A lower level is used for the entire file, so the file may not sound as loud as other files processed with the same level where clipping was not required.
Abort processing if clipping is required	Processing is aborted immediately when clipping is required. A range error is displayed. Use this setting for batch processing to prevent writing clipped audio.

Maximize Volume (Normalize)

Maximize Volume searches the selection for the current peak volume level. It then displays the level and the position of the level within the file. You can then specify a new absolute maximum volume level. The volume of the entire selection is changed so that the maximum will match that value. A value of 0dB gives you full dynamic range. This is often referred to as "normalizing" the volume.

Some effects in GoldWave may cause the volume level to go above 0dB. After using many effects, you should use the Maximize Volume effect before saving a file to ensure that the full dynamic range is not exceeded. Otherwise clipping may result in the saved file.

Always use Maximize Volume before saving a file to avoid clipping and make full use of the dynamic range.

Shape Volume

Shape Volume presents Shape Controls where the volume envelope of the selection can be defined. The shape line is initially horizontal at 1.0, representing normal volume. By bending or moving the line, you can dynamically change the volume over the selection. Adding a point below 1.0 decreases the volume. Adding a point above 1.0, in the red section, increases the volume. Note that increasing the volume may cause clipping distortion. Several preset shapes are included.

The `Show envelope` option calculates and displays the current volume envelope of the sound. The left channel envelope is shown in green and the right channel envelope is shown in red.

Example
To use this effect to reduce the volume of background music before mixing in a voice recording or narration over the music:

1. Open the voice file and make note of its length.
2. Choose **Copy** from the **Edit**.
3. Open the music recording.
4. Choose **Shape Volume** from the **Effect | Volume** submenu.
5. Select the "Fade for voice-over" preset.
6. Adjust the trough of the shape so that its time length is slightly larger than the length of the voice file and its position is located where you want to mix the voice file.
7. Choose OK.
8. Adjust the start marker to the position where you want to mix the voice file (the beginning of the trough).
9. Choose **Mix** from the **Edit**.
10. Choose OK.

Playback Rate

The Playback Rate effect changes the playback rate of the *entire* sound. The sound plays faster (or slower) and its pitch is higher (or lower), similar to playing a vinyl record faster or slower. Essentially, this just changes the sampling rate value shown in the status bar. To change the playback speed without changing the pitch, use the Time Warp effect.

While the playback rate of the audio device is controlled with the speed fader in the Control window, this has no effect on the file's data. You must use the Playback Rate effect for the change to be savable. Some file attributes may have a fixed sampling rate. A warning is shown in that case. An MP3 file with "Layer-3, 44100Hz, 192kbps, stereo" attributes, for example, is always saved with a 44100Hz sampling rate. You must use Save As to select attributes with the rate you want.

Resample

The Resample effect changes the sampling rate of the *entire* sound. Unlike Playback Rate, this command re-calculates and interpolates all the data so that the pitch and playback time are not affected. Use this command to convert any sampling rate to the standard CD rate of 44100Hz or the standard telephony rate of 8000Hz.

Resampling is done using a high quality polyphase algorithm with a filter length of 192.

If you have a sound recorded at 44100Hz and do not need CD quality, you can save large amounts of disk space by resampling the sound to 22050Hz or 11025Hz. This reduces the size by 2 to 1 or 4 to 1.

View Menu Commands

Before reading this section, review the terms introduced in the Interface Overview and Editing Overview sections.

View commands allow you to see a more detailed graph of part of the sound. They are similar to zoom commands in the Windows Paint accessory. When you zoom in (or magnify) the sound, you see a smaller section, but with greater detail. When zoomed out, you see the entire sound, but with less detail. The Overview box near the bottom of each Sound window gives you some information about what section of the sound is currently shown in the view (see the Main Window figure).

When zoomed in on part of the sound, a scroll bar appears at the bottom of the Sound window for moving to different parts of the sound while still keeping the same level of magnification. Clicking-and-dragging the waveform with the middle mouse button (wheel button) is another way of moving around. The current level of magnification is displayed in the Main window's status bar next to the modified status. The magnification is given as two numbers separated by a colon. The first number is the number of pixels and the second number is the number of samples mapped to those pixels. For example, 1:1 means one sample is mapped to one pixel and for 51:1003, 1003 samples are mapped to 51 pixels.

Most view commands use the start marker's position as the starting location for magnification, so you should move the start marker to the position of interest first.

Use the Options | Window command to set the initial zoom level when a file is opened.

All

The entire sound is graphed in the view. In other words, it zooms all the way out so that the entire sound is visible. You can move the start and finish markers to select any part of the sound.

Specify

This magnifies the graph to any level you specify. The level can be given as a time length or as a ratio. If the Length option is selected, then the length you specify is shown in the graph. For example, use 1:00 to show one minute of audio. If the Ratio option is selected, then the given number of samples are mapped to a single pixel on the screen. Values greater than 1 display an approximation of the waveform. Values less than 1 (such as 0.1) reveal individual samples and allow direct editing of the waveform.

The Start time specifies what position in the file to begin drawing the zoomed waveform. If the given level is not possible, the closest valid level is used.

Selection

The selection is magnified, increasing the detail of the graph. You can zoom in many times by changing the selection and magnifying it again until only a single sample is shown in the view.

Click-and-drag the right mouse button over the waveform to zoom in without changing the selection.

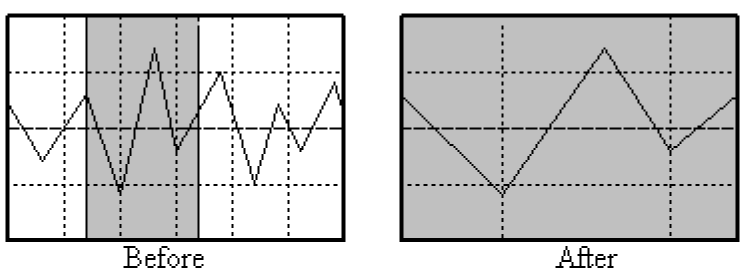


Figure: View Selection

Preset

A **Preset** button is provided in the tool bar so that you can quickly zoom to your favourite level. The sound is magnified to the level of detail specified under Options | Window. The level can be set to any value you find convenient.

Previous

This returns the view to the previous zoom level. Use this to switch back and forward between two different zoom levels.

Auto Scroll Lock

When this item is checked, the view scrolls automatically to follow playback and recording. Scrolling occurs only when zoomed in and when the current playback or recording position goes outside the view. This feature also enables keyboard navigation, which allows the playback cursor to be moved via the keyboard.

Zoom In

Magnifies the sound by a factor of 1.33x. This gives 33% more detail, but shows 33% less sound. The middle of the view is used as the zoom center. This command complements Zoom Out.

Click-and-drag the right mouse button over the waveform to zoom in on any part of the sound.

Zoom Out

Reduces magnification by a factor of 1.33x. This gives 33% less detail, but shows 33% more sound. The middle of the view is used as the zoom center. This command complements Zoom in.

Zoom 10:1

When the number to the left of the colon is greater than the number to the right, a very small section of sound is magnified at a high level of detail. At these levels, individual samples are easily visible and direct waveform editing with the mouse is possible.

Zoom 1:1

At a level of 1:1, each audio sample is represented as a single pixel on the screen. This reveals a true representation of the shape of the sound.

Zoom 1 Second, 10 Seconds, 1 Minute, 1 Hour

These show the given amount of time of audio beginning at the start marker's position. You can use View | Finish to see the audio at the end.

Vertical Zoom All

Vertically zooms all the way out so that the entire vertical amplitude range of the sound is shown.

Vertical Zoom In

Magnifies the graph vertically to show 1.33 times as much amplitude detail. Zooming is centered on the horizontal center of the view.

Vertical Zoom Out

Reduces vertical magnification to show 1.33 times less amplitude detail. This show a larger range of the amplitude. Zooming is centered on the horizontal center of the view.

Start and Finish

These commands scroll the view to either the start or finish marker's position. The view is centered over the marker's position provided its position and the level of magnification permit it to be centered. These commands are especially useful when you need to move a marker to a precise position. For example, you can zoom in 1:1 and move the start marker to an exact position and then use **View | Finish** to set the finish marker's position.

Tool Menu Commands

GoldWave includes several tools for working with audio files. These are described in the following sections.

CD Reader

The CD Reader tool digitally copies audio directly from an audio CD to a file on your hard drive, without using your sound card. This features has several advantages over normal recording:

- There is no need to create and initialize a new file.
- Audio can be compressed while extracting, saving of hard drive space.
- Recording volume controls do not have to be selected or adjusted.
- Synchronizing sound card recording with CD playback is not required.
- Many CD-ROM drives read audio several times faster than sound card recording.
- Information such as artist, title, and album can be downloaded and saved automatically.

The CD-ROM drive must be MMC compliant (Multimedia Command Standard). Use of an ASPI driver is not recommended. For Windows 2000, XP, Vista, or later, support is built-in. Due to the wide variety of interfaces and inconsistent device standards, incompatibilities may arise that will require a system reset. It is recommended that you close all other programs before proceeding.

If your system is configured correctly, the window allows you to select a CD device and to specify tracks to copy. If you have only one CD-ROM drive, only one device is listed in the drop down list. If you already have an audio CD in the drive, the track times are shown in the lists. Otherwise you will need to insert an audio CD and re-select the device from the list.

Read Tracks Tab
Use this tab to copy several tracks to separate files. The Get Titles button downloads information (album, titles, etc.) about the CD from an Internet database. An active connection to the Internet is required. You can select a single track from the list and use the Rename button (or Alt+R) to manually rename it. Use the Save Tracks button (the one with the diskette icon) to save all titles and disc information to the hard drive.

Use the Select All button to select all tracks or check the box for each track you want to save. You can preview tracks by selecting the Read Time Range tab, described below.

The Save button saves all selected tracks to separate files using the track title as the filename. You can select the file format and other options on the Save CD Tracks window that appears.

Use the Options tab to configure settings before saving.

Read Time Range Tab
Use this tab to copy any time range from the CD or to preview parts of the CD. A fast CD-ROM or DVD drive is required for previewing. You may have to increase the "Read speed" or the "Number of sectors per read" setting under the Options tab to get smooth playback. See the Options tab settings below.

The Save button saves the given time range to a single file. You can select the file format and provide a filename in the standard Save As window that appears.

Options Tab
Use the Options tab to changes settings for reading audio from the CD and the Internet database server.

Table: CD Reader Options

CD Reading Options

Option	Purpose
Number of sectors per read	Sets the number of sectors to read from the CD at one time. This value should be as large as possible, provided your CD-ROM drive supports it. Reading may fail if the value is too large for a given drive. Higher values increase overall reading speed.
Number of sectors	Sets the number of sectors to overlap. A value

to overlap	of 3 is recommended. Using a higher value more forcefully corrects jitter (positioning defects), but slows down reading because more sectors have to be re-read. Use a value of 0 if your CD-ROM drive automatically corrects jitter to speed up reading.
Read speed	Many CD-ROM drives allow the spindle speed to be controlled. For fast reading, this value should be set as high as possible. A single speed CD reads data at 150kBps. To read at 10x, for example, set the speed to 1500kBps.
Swap bytes	Changes the order of the bytes read from the CD. Some drives incorrectly return audio data in the reverse byte order. This gives loud, badly distorted audio. Check this option to correct it.
	CD Database Options

Server	Sets the server address of the database to use for downloading CD information. By default, it is freedb.freedb.org, but you can use a mirror site that is closer to your physical location for faster access. The server must contain the "/~cddb/cddb.cgi" path.
	
	Examples:
	`freedb.freedb.org/~cddb/cddb.cgi`
	`us.freedb.org/~cddb/cddb.cgi`

Proxy	Sets the HTTP proxy server for your network, if required. Leave this blank if no proxy server is used. Otherwise set the server and port in the `servername:port` format. Contact your system administrator or ISP for more information if you are unable to connect to the database. Example: `http-proxy.provider.net:8080`
Automatically download titles	Automatically downloads titles when the CD Reader tool is started or when a CD device is selected. This eliminates the need to use the Get Title buttons.
Prefix text and track number in	Prefixes titles with the given text and track number when displaying titles. Enter text in

title	the edit box with at least two # symbols for the track numbers, such as "## ", "Disc 1 ## " or "01-##-". The text must contain characters that are safe for filenames, so "Disc:1/##/ " is invalid.
Specify CD ID manually	Allows a different CD ID to be given for the database search. When the titles are about to be downloaded, a window appears where the calculated ID can be changed. A category must be specified since it is required by the database query. This option is for advanced users and should rarely be checked.
	Device Interface Options
ASPI	Uses the Advanced SCSI Programming Interface (ASPI) to access the CD-ROM drive. Usually an ASPI driver is included with Windows 95/98/ME, but not with Windows 2000/XP.
SPTI	Uses the SCSI Pass-through Interface (SPTI) to access the CD-ROM. This option is not available in Windows 95/98/ME and is for Windows 2000/XP.
	Editing Options
Open track files for editing	Automatically opens each track file in GoldWave immediately after it is successfully read from the CD. This makes it easier to edit the files later. If you do not need to open or edit the files, keep this option off to save processing time.

Troubleshooting

- If you get synchronization errors at the beginning, try continuing several times before giving up. Some drives are very inaccurate when spinning up and may require many continues before synchronization can be reliably established. If that does not help, set the "Number of sectors to overlap" value to zero on the Options tab and try again.
- If you see an error message indicating a read problem, make sure that the CD is free of dust and finger prints. If the CD contains a data track, try using the Read Time Range tab instead with a slightly increased From time and a slightly decreased To time to avoid accessing the data track.
- If preview playback is intermittent, increase the "Read speed" and the "Number of sectors per read" or reduce the "Number of sectors to overlap" under the Options tab.

Control

Use this command to show or hide the Control window. See the Control Overview section for more information.

Cue Points

Cue points mark and describe specific positions within sounds. They have numerous uses. When recording speech, for example, you can use them to hold information about the speaker or a translation of what the speaker said. For music, you can store lyrics for each verse. If you design instrument samples, cue points can hold looping points. Some multimedia applications use them to play or loop specific sections of a sound. When transferring albums to CDs, cue points can mark track division points, which can be used later to divide a large file into individual songs or tracks.

Cue points are shown as inverted triangles in the cue points slot of a Sound window, just above the time axis. If two cue points overlap, the colour of the cue point will be brighter.

Cue points are saved only in certain files types.

Cue point are adjusted automatically when a file is edited, but not when effects are applied. Any effects that alters the length of the selection, such as Time Warp or Silence Reduction, will cause the cue points within the selection to be misplaced.

There are several ways to create a new cue point:

- Use the New button on the Cue Points tool window, then enter the name and position.
- Right-click on the cue points slot in the Sound window (see the Main Window figure) and choose New Cue, then enter the name. The position is already set based on where you clicked the mouse.
- Play the file and press Ctrl+Q or use Edit | Cue Point | Drop Cue to set a cue point at the current playback position. This also works while recording.

To edit an existing cue point, you can:

- Select the cue point from the list in the Cue Points tool window and choose the Edit button.
- Right-click on the cue point in the Sound window and choose Edit Cue.
- To change a cue point's position, drag-and-drop it to the new position in the Sound window's cue point slot.

To delete a cue point, you can:

- Select the cue point from the list in the Cue Points tool window and choose the Delete button.

- Right-click on the cue point in the Sound window and choose Delete Cue.

The Delete All button removes all cue points in the file. Use this button before using the Auto Cue button if you want to remove all existing cue points before automatically generating new ones.

Additional menu commands are provided to move cue points to the start or finish marker's position or vice versa. Right-click on a cue point in the list to display the menu.

Click on a column header to sort cue points by number, position, or name.

Select a cue and use the F4 key start playback at that cue. Press F8 to stop playback.

Copy All Button
Use this button to copy all the cue point information into the clipboard. You can paste this into a text editor, such as the Notepad accessory.

Split File Button
The Split File button divides a large file onto smaller files using the cue points as split positions. If you've recorded one side of an album and need to divide it into individual songs, for example, you would set a cue point at the start of each song, then use this feature to automatically create separate files for each song. Each file can then be written to a CD-R disc as a separate audio track using CD Recorder software.

Use the Auto Cue button (below) to automatically set cue points at silences between songs.

By adding the text "[-Exclude-]" to a cue name, that cue point and section of the file is excluded from the split files.

Any information entered in File | Information before splitting is stored in each split file, if possible.

The Split File options are given in the following table.

Table: Split File Options

Option	**Purpose**
Use this folder	Sets the destination folder where all the split files are stored.
Use file's current folder	Uses the original file's folder as the destination folder for all the split files.
Overwrite existing files	If this is checked, then files with the same name that already exist in the destination folder are replaced (overwritten). If the option

is not checked, then splitting is aborted if a file with the same name is found.

Filename template

The template may contain any combination of text, number symbols (#), or tokens listed below. The drop down template list contains some commonly used presets. The number symbol (#) is replaced with the sequential split file number. Entering "Track##", for example, names the files named Track01, Track02, Track03, etc. The # symbol can be placed anywhere in the template, so names like "#### - CD1" or "#Track##" would be valid. The least significant digit is placed in the right-most # slot, so the first names would be "0001 - CD1" and "0Track01". Tokens are replaced with information from the original file, which has to be set before splitting the file.

Token	**Meaning**
#	The sequential split file number, starting at `First number`. The template may contain more than one #, each representing a single digit for the number.
<Album>	Album from the original file information.
<Artist>	Album Artist from the original file information.
<Cue Name>	Name assigned to the cue point.
<Genre>	Genre from the original file information.
<Original Filename>	Filename of the file being split.
<Title>	Title from the original file information.

First number

Sets the starting number to use in the # slots for naming split files.

Discard cue point

If checked, then split files will not contain the cue point located at the beginning of each split. If this is not checked, then the cue point's name and description are stored in each file. If you've entered a description for each cue point and want it stored in each split file, make sure this option is not checked.

Replace track number

If checked, the track number information stored in split files is replaced with the sequential split number starting at `First`

	`number`. Otherwise the track number of the original file is stored.
Set title to filename	If checked, the title information for each split file will contain the split filename. Otherwise it will contain the original file's title.
Use CD compatible wave format and alignment	Ensures that each file is stored in a CD compatible Wave format and that the length of each file is exactly aligned to a CD sector boundary, eliminating gaps between files/tracks. For accurate, glitch free alignment, you must resample the file to 44100Hz before splitting. This helps you create seamless tracks on a CD, provided you configure your CD-R software to not write silence between tracks. Note that if the end of the last track file does not contain enough audio to perfectly fill a CD sector, a tiny section of audio (usually silence) may be discarded for alignment. If the file does not end in silence you can use the Edit | Insert Silence command to add 0.0133 seconds of silence to pad the end of the file before splitting.
Use default save format and attributes	Uses the format given under the `Default Save Format` tab of the Options | File Formats window to create the files. No alignment is applied. Choose the Set button to change the format.
Use file's current format and attributes	Uses the format and attributes of the file being split, as shown in GoldWave's status bar, to create the files. No alignment is applied.

Auto Cue Button
Use the Auto Cue button to automatically create cue points. It operates in two modes: Mark Silence and Spacing.

The Split File button on the Cue Point window can be used later to split a file into individual pieces based on these cue points.

Mark Silence Mode
This mode sets cue points at silences, such as marking quiet sections between songs. A single cue point is added in the center of each detected silent region.

The `Threshold` value sets the volume level for the silence. In most cases, like vinyl recordings, the value should be -40dB or higher so that any background hiss, pops, or clicks are treated as silence. Otherwise no silence would be marked at all. If you find that no cue points appear, try increasing this value to -30dB or higher. If you find that too many points appear, delete them, then decrease this value or change the values below. Using the Pop/Click and Noise Reduction filters effect first may improve silence detection.

`Silence length` specifies how much silence is required before it is marked. Some songs contain brief silences that you usually do not want marked. This value helps to avoid marking any brief pauses within a song. Try values between 1.0 to 1.5 seconds to ignore these brief silences and mark only the longer silences between songs.

`Minimum separation between cues` controls the minimum amount of time between one cue point and the next. If you know all the songs are longer than 2 minutes, for example, then you can set this value to 2:00 to ensure no silences within a song are marked. All cue points will be at least two minutes apart.

`Cue placement within area` specifies where to place the cue point within the detected silent area. A value of 0 means at the beginning of the silence, a value of 100 means at the end of the silence. The default value of 50 places the cue point in the center of the silence area.

Spacing Mode

This mode sets cue points at fixed intervals, such as having cue points every 5 minutes. Cue points are added at the specific interval, starting at the given time.

`Start time` sets the time to begin marking the file. If you enter 1:00, then the first cue point is inserted at time 00:01:00 in the file. Normally this value would be zero.

`Interval` specifies the time interval to use between each cue point. A value of 5:00 would set cue points at five minute intervals (00:05:00, 00:10:00, 00:15:00, etc.).

Cue Naming

The `Cue naming` options specify how cue points will be named as they are added. Select `Time based` to name each cue point based on its time position within the file. A cue point added at two minutes into the file, for example, will have the name 2:00.00000.

Select `Numbered` to sequentially number each cue point as they are added. The names will be 001, 002, 003, etc. The starting number depends on how many cue points are already in the file.

Select `Lettered` to name each cue point alphabetically, with three letters, such as AAA, AAB, AAC, etc. The starting name depends on how many cue points are already in the file.

Import/Export Buttons
Use the Import button to read cue points from a CD cue file. Use the Export button to save all cue points to a CD cue file. The name of the cue file depends on the name of the current Sound file. For example, if the file you are working on is "music.wav", then the cue file is "music.cue" by default. See Options | Storage for a setting to use cue files automatically.

A CD cue file contains track information that some CD Recorder programs, such as CDRWIN and Nero, use when creating a table of contents for a CD. Creating a cue file may eliminate the need to split a large file into separate track files. You can open the cue file in Notepad to edit or view its contents.

Cue point positions in a cue file are accurate only to 1/75 of a second, which is the size of a single CD sector. They are stored more accurately within some sound file types.

Effect Chain Editor

The Effect Chain Editor allows a number of effects to be chained together so they are all processed at once. There are many advantages to using chains, such as:

- Easier automation, less work. You can apply many effects with just a few mouse clicks.
- Less storage requirements. Extra temporary storage is not required for each effect in the chain.
- Faster, more efficient processing. Audio data is read from storage, processed, then written to temporary storage only once for the entire chain rather than for each effect.
- Create new effects. Chaining effects is a way of creating new effects.
- Advanced previewing. Preview how a series of effects will sound without ever having to process the entire file.

The left window is a tree list showing all the effect plug-in modules, with the GoldWave branch expanded initially. Only effects that can be chained are listed. Effects requiring special access to the audio data (like scans) or ones that are time based cannot be chained.

You can drag-and-drop effects to the right "chain list" window or select an effect and choose the Add button. Effects are always added to the end of the chain list. Expand the branch of other listed modules to use effects in those plug-ins.

Use the Remove Last button to remove an effect from the chain list. Note that only the last effect in the list can be removed. An effect in the middle of the list cannot be removed unless all effects below it are removed first. Use the Remove All button to remove all effects in the chain list.

When an effect is added to the chain list, it appears as a button. Use the button to change settings for that effect. Settings can be changed while previewing the audio.

When you have finished creating the chain, use the Presets controls near the bottom of the Effect Chain Editor window to save the entire chain as a single preset.

Expression Evaluator

The Expression Evaluator is a general purpose tool for manipulating and generating audio data. For a detailed explanation, see Appendix C.

File Merger

Use the File Merger tool to join together separate files into a single file. Files may be added in several ways:

- Use the **Add Files** button and select one or more files to add.
- Use the **Add Opened** button to add all currently opened files.
- Use drag-and-drop or copy-and-paste from Windows Explorer (use **Ctrl+V** to paste).
- Use drag-and-drop from iTunes.

Remove a file by selecting it and choosing the `Remove` button. The `Remove All` button removes all files from the list.

Files are joined in the order they are listed. Drag-and-drop files within the list to change their order or click on a column header to sort by file name or by file date. Clicking on the same column header a second time reverses the sort order. If more files are added to the list, they will not be sorted automatically and the column header must be clicked again if sorting is required.

Set the "Preferred sampling rate" for the merged file. This rate is used only if a rate is not specified in the attributes selected after you choose the `Merge` button. Many attributes have a predefined rate. The preferred rate will be ignored for those attributes.

As each file is merged, a cue point is added at the junction point using either the file's Title information, if present, or filename. Check the `Export cue file` box to export these cue points to a separate Cue File.

Choose the `Merge` button to specify a filename, a file type, and attributes for the merged file. Merge processing begins immediately after you choose the `Save` button.

To split a file into smaller sections, see the Split File section under the Cue Points tool.

Speech Converter

The Speech Converter tool converts written text to spoken audio (text-to-speech) or spoken audio to text (speech recognition or dictation).

GoldWave uses the speech software in Windows to perform all conversion, so the quality of the voice or the accuracy of the recognition depends entirely on that software. The Speech Converter tool is not supported and will not work in versions of Windows that do not include the speech software. For more information see Microsoft's Speech website. Different voices and recognition engines are available from other vendors.

Use the **Speech** settings in the Windows Control Panel to configure text-to-speech and train speech recognition to your voice.

The Speech Converter tool consists of a text area with buttons above and below. The buttons along the top open a text file, save the text to a file, and perform basic editing functions on text. Use the Context Menu key (or Shift+F10) to display all the button commands as a menu for easier accessibility.

The buttons below the text area speak the audio, save the speech to an audio file, or take dictation from the microphone or an opened audio file.

Text To Speech

Copy the contents of a website, document, report, or even chapters from a digital book (Sherlock Holmes, for example) and Paste them into the text area to have GoldWave read them to you. You can save the audio directly to a file to copy it to your iPod or other portable player to listen to while jogging, working out, or doing other activities where reading isn't possible.

Use the `Speak` button to read the text. Playback is started from the edit cursor or at the beginning of the selection, if there is one. If the edit cursor is at the end of the text and there is no selection, then the entire contents of the text area is read.

Use the `Voice Settings` button to change the voice, volume, speed, and pitch. Windows usually includes just one voice, but others can be installed.

XML modifiers (such as <pitch middle='5'/>) are supported within the text to change the voice. Search online for "SAPI XML tags" for more information about modifying the voice using tags.

Use the `Speak To File` button to read the text directly to an audio file. Text is processed much faster than speaking through the sound hardware. Be sure to select all the text, otherwise only the selection or the text after the edit cursor is read to the file.

Audio files will be significantly larger than the original text files, so you must select a file type and attributes that minimize the audio file size while preserving good quality. Fortunately voice files do not require a high sampling rate or bitrate. The bitrate number, given in kilobits per second (kbps), controls the amount of space required per second of audio. Examples are given in the table below, with the amount of storage per minute. Note that CD quality audio requires 10MB per minute, which should be avoided if you intend to copy the audio to a portable player. The MP3 format may be the only one that will play on all portable players.

File Type	Attributes	MB/min
MPEG Audio	Layer-3, 22050 Hz, 32 kbps, mono	0.240
Windows Media Audio	WMA Voice 9, 20 kbps, 22.05 Hz, mono	0.150
Ogg	Vorbis 22050 Hz, 30kbps (0.1q), mono	0.225
Wave	PCM signed 16 bit, stereo (44100 Hz, CD Quality)	10.58

When attributes do not include a sampling rate, then the rate specified under `Voice Settings` button is used.

Speech To Text (Speech Recognition)

Use the **Dictate** button to record or convert audio to text. The source audio can be taken directly from the computer's Microphone or from the currently opened file (if present). Use the `Dictation Settings` to select the source and configure the microphone or train the recognition engine.

When processing from a file, a progress bar appears at the bottom of the window. Note that for long sections of speeches without any pauses, there will be a significant delay before any text appears and the progress bar advances.

Speech recognition is an evolving technology and is still far from perfect, so don't expect highly accurate dictation. Any background noise or music at all adversely affects accuracy. Using GoldWave's Noise Reduction effect or other filters may reduce accuracy as well. Recordings must be as clean as possible, without any effects processing.

⚠Using the Microphone source may change your computer's audio settings and GoldWave's recording input. Use Device and Volume properties to reselect the input before starting any new recordings.

Options Menu Commands

These command configure and customize GoldWave.

Colour

Use this command to change the colour scheme of Sound windows. A preview window shows the Sound's current colour scheme. Use the `Item` drop down list and the Colour button to change the colours or select a different scheme from the preset `Schemes` drop down list. You can click the mouse in the preview window to select an item you want to customize. Check the **Gradients** box to make the waveform and background colours gradients rather than plain, solid colours.

Control Properties

Displays the Control Properties window.

File Formats

Use File Format Options to:

- Associate a filename extension (such as .snd or .vox) with an audio type and attributes.
- Change the precedence or enable and disable File Format Plug-ins used by GoldWave.
- Set a default format for saving files.

These options are contained under separate tabs.

Undetectable Types Tab
Use this feature to associate a filename extension with an audio type and specific attributes. This is useful for automatically opening files that do not contain any information describing their format (raw files). For example, if you work with Dialogic telephony files, you can associate the .vox extension with a specific plug-in type. In this case, you'd use the Dialogic type, usually with "ADPCM 4 bit, 8000Hz, 32 kbps, mono" attributes. Whenever you open a .vox file, GoldWave will assume that format without asking you to specify it each time. This must be done prior to using File | Batch Processing when working with raw files.

The list shows all current associations, if any. Use the Add button to create a new association. Use the Edit button to modify an existing association. Use the Remove button to remove an old association.

Example
To associate .vox with Dialogic ADPCM decoding:

1. Choose **File Formats** from the **Options** menu.
2. Choose the **Undetectable Types** tab.
3. Choose the **Add** button.
4. Enter "vox" in the **Extension** box. Do not enter a leading period.
5. Select "Dialogic (vox)" from the **File type** drop down list.
6. Select "ADPCM 4 bit, 8000 Hz, 32 kbps, mono" from the **Attributes** drop down list.
7. Choose the OK button.

See Appendix A for additional information about file attributes.

File Plug-in Precedence Tab
Use this feature to change the order that audio files are passed to plug-ins for opening, or

to enable and disable plug-ins. By default all plug-ins found in the File plug-in folder are listed and enabled, with the built-in GoldWave plug-in listed first.

Select an item in the list and use the Lower and Higher buttons to change the order. The plug-in at the top of the list is the first one to be given the opportunity to open a file, if it recognizes the format. Otherwise, the file is passed to the next plug-in and so on until the file can be opened or no plug-ins are left. See File Format Plug-ins for more information. Check or uncheck an item to enable or disable that plug-in module.

Default Save Format Tab
Use this feature to set a default save format for new files when using the File | Save command. The same format can be used for all "save" related commands (Save As, Save Selection As, etc.) if you check the "Use this format for..." box. Use the `Save as type` drop down list to select the type first, then use the `Attributes` drop down list to select specific attributes. Whenever you create and save a file, this format will be selected by default. Note that when creating a new file, you can select attributes (sampling rate and number of channels) that differ from the default save format. Be sure to select appropriate attributes to avoid conversions later when saving.

The "Do not allow other file types..." option disables the file type and attributes boxes under "save" related commands to prevent them from being changed. It forces the use of the default format.

Keyboard

Use Keyboard Options to set keyboard assignments (shortcuts) for functions in GoldWave. The list of the current assignments is given in the window with the function in the left column and the assigned shortcut in the right column. Use the arrow keys to move between items. To change a shortcut, move to right column for the function shortcut you want to change and type out the full text of the new combination of keystrokes, such as "Ctrl+Shift+Up", "X", "Ctrl+P", "Shift+Enter", etc.

Note that the Left, Right, Page Up, and Page Down keys are used for navigation and scrolling the view and should not be reassigned. Combinations of these keys with Ctrl and Shift are also reserved. The spacebar ("Space") is used to activate the buttons in the Control window when that window has the keyboard focus.

Choose `OK` to save and use keyboard assignments. Use the `Defaults` button to erase all keyboard assignments to restore them to their original, installed settings when GoldWave is restarted.

Use the `Load` and `Save` buttons to load or save keyboard assignments from or to a text file. These buttons allow you to backup assignments or load assignments created by others.

Use the `Search for functions...` box box and `Find` button to find functions. Enter "Play", for example, to find playback related functions.

The following table lists some of the names of keys that differ from their physical label.

Table: Key Names

Key	**Text To Enter**
Control	Ctrl or ^ ^X = Ctrl+X
Backspace	BkSp
Return	Enter
Left Arrow	Left
Right Arrow	Right
Up Arrow	Up
Down Arrow	Down
Page Up	PgUp
Page Down	PgDn
Insert	Ins
Delete	Del
Numeric Keypad 5	Num 5
Numeric Keypad /	Num /
Numeric Keypad *	Num *
Numeric Keypad -	Num -
Numeric Keypad +	Num +
System Request	(not usable)
Scroll Lock	(not usable)
Pause	Pause (may not work)
Escape	Esc
Space Bar	Space
+ (plus)	Shift+= (US Keyboards)
@ (at)	Shift+2 (US Keyboards)
Media Control Buttons*	
Play/Pause	mxPlayPause
Stop	mxStop
Previous track	mxPreviousTrack
Next track	mxNextTrack
Browser Back	mxBrowserBack

Browser Forward	mxBrowserForward
Browser Refresh	mxBrowserRefresh
Browser Stop	mxBrowserStop
Play	mxPlay
Zoom	mxZoom

* Note that using the media control buttons may cause the wrong shortcut text to appear in menus. Browser buttons many be assigned to other tasks by the operating system. Also some keyboards do not map keys properly.

Scan codes for keys can be entered directly by entering `0x` followed by the hexadecimal scan code, such as `0x24` for the Home key. GoldWave shows key codes and names in the lower right corner of the window when possible. If nothing is shown or it doesn't change, then the key cannot be used.

See Also
[Accessibility](#)
[Keyboard Commands](#)

Plug-in

Presents a submenu for configuring effect, file, and visual plug-ins, if any. Plug-ins may be created by different developers. Consult the documentation included with the plug-in for more information. GoldWave Inc. does not provide technical support for plug-ins developed by other companies or individuals.

DirectX Audio Plug-in Configuration

Shows a list of compatible DirectX Audio Plug-ins currently installed on your system. Use this configuration to control what plug-ins GoldWave lists under the **Effect | Plug-in | DirectX** menu and to associate a different icon with a plug-in. Check the box next to an item in the Plug-in list to add it to the menu. Uncheck the box to remove it from the menu. Select an item, then choose an icon from the Icon list to associate that icon with the item. Note that GoldWave must be restarted before changes are acknowledged.

GoldWave requires plug-ins that process in IEEE 32 bit floating point audio (WAVE_FORMAT_IEEE_FLOAT) format. It will not list plug-ins that process only in PCM 16 bit audio format.

If you are unable to get a particular DirectX Audio Plug-in working, please contact the plug-in developer, not GoldWave Inc.

GoldWave Audio Plug-in Configuration

The GoldWave Audio Plug-in Configuration window configures the internal GoldWave file plug-in and controls general file and MP3 decoding and 8 bit encoding.

Many file types are handled directly, but some are handled through external (operating system) decoders installed on the system. These system decoders may cause problems for raw file data or non-standard or unusual audio types. Use the system options to control when the decoders should be used and what file types to avoid. DirectShow decoders were widely used in versions of Windows prior Vista. Media Foundation has replaced DirectShow as of Vista.

The MP3 encoding settings control how MP3 files are encoded. Usually these do not need to be changed.

Use the dither setting to mask quantization distortion caused by converting high quality audio to low quality 8 bit. A triangular dither masks the distortion with white noise. Keep in mind that 8 bit quality is inherently noisy, so the white noise is quite audible, but usually less distracting than quantization distortion.

⚠Dithering only affects the quality of the audio saved to the file. GoldWave maintains high quality audio internally, so it is never dithered even after saving. You must close and reopen the file from the drive to hear the 8 bit audio quality.

Storage

The Storage Options window configures folders, file storage, and undo levels. See the Storage Overview section for additional information.

Sound Folder
This specifies the folder to use when you start GoldWave. The File | Open command lists files in this folder whenever you start GoldWave. You can make GoldWave remember the last folder used or always start in the given folder. Use the folder button to browse for a folder.

The Clear Recent File List button removes the list of recently opened files under the File menu.

Temporary Storage
This controls how GoldWave stores audio, as explained in the Storage Overview section. When using hard drive storage, you need to specify the folder to use when creating temporary files. This folder should be located on a large, local hard drive with plenty of free space. Changing this folder does not affect opened files already in temporary storage. Use RAM storage only when working with small files for faster, efficient editing.

Undo Levels
This sets the number of edits/changes you can undo. A value of 5 means you can undo the five most recent changes performed on the file. A value of 0 means that you cannot

undo any changes. Larger values use much more storage to store the previous audio data. When using the RAM storage option, this should be set to a small value. Otherwise RAM may be depleted quickly and system performance will degrade.

Cue Points Storage

Use the "Automatically import and export separate cue file" setting to automatically save or load cue points from a separate cue file when saving or opening a sound file. If you open a file named "music.mp3" cue points are imported from a "music.cue" file, if it exists. If you save a file named "music.mp3", cue points are saved in a "music.cue" file (overwriting any existing file). Automatic import and export is done only for file formats that cannot store cue point information internally. Cue files are not saved automatically for other file formats (such as `.wav`). Use the Cue Points Tool to manually import or export a cue file.

When automatically exporting cue files, any existing cue file with the same name will be overwritten without notification!

Tool Bar

The Tool Bar Options let you customize GoldWave's tool bars. Initially the top tool bar contains main menu commands, while the lower tool bar contains only effect commands. Select an item and use the Add and Remove buttons or drag-and-drop items between the Available and Current lists to control the layout of the tool bars. The order which items appear in the Current list is the same as the order they will appear on the tool bar. You can drag-and-drop items within the Current list to rearrange them.

Use the Visible checkbox to show or hide the entire tool bar. Use the Gray-scale checkbox to convert the coloured images to blank-and-white. Use the Captions checkbox to display text within the Main tool bar buttons. Use the "Tool bar images in menu" checkbox to display images in GoldWave's main menu. Do not check this option if screen reader accessibility is required.

You can drag-and-drop the tool bars themselves within GoldWave's Main window to change their locations. Click-and-hold on the vertical bumps near the left edge of the tool bar to drag it.

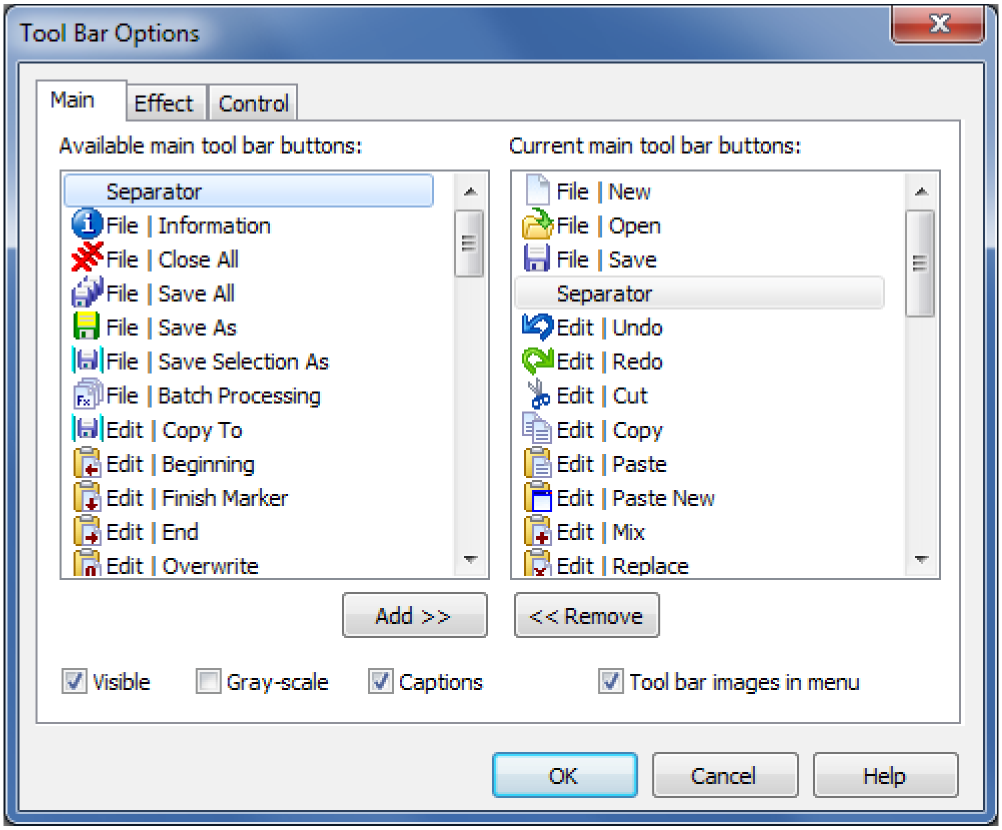


Figure: Tool Bar Options

Window

Use Window Options to configure the positions of the Main and Sound windows, set axis options, specify preset and initial zoom values, and set miscellaneous options.

Main Window Size

This controls the Main window's position and size when GoldWave is started. `Normal` gives control to Windows. `Maximize` makes the Main window occupy the entire screen. `Save position` saves the Main window's position and size when GoldWave is closed so that it will appear in the same location next time.

Sound Window Size

This controls the position and size of Sound windows. `Normal` gives control to Windows, which usually results in cascaded windows. `Maximize` makes a Sound window occupy the entire Main window. `Auto-tile` resizes all Sound windows whenever a new sound is opened or closed so that every one is visible. Arrangement is the same as the [Window | Tile Horizontally] command.

Axes Numbering

`Y` displays or sets the units of the vertical axis in Sound windows. Selecting `Off` hides the axis completely. `Normalized` shows an axis with a range of -1.0 to 1.0. `Signed 16 bit` shows an axis ranging from -32768 to 32767, which is the range of a 16 bit sample. `Unsigned 8 bit` shows an axis with a range of 0 to 255.

`X` sets the format for displaying the horizontal time axis near the bottom of Sound windows. "Hours : minutes : seconds" gives the time as three sets of numbers separated by colons, such as 12:23:56. "Minutes : seconds" gives the time as two sets of numbers separated by colons, such as 1234:56. `Seconds` gives the time as a floating point number, such as 1234.56. `Samples` gives a sample count.

Grid
Use these settings to display a custom vertical grid over the waveform, which can be used to align markers. Check the `Grid` box to enable the settings. `Spacing` is the time between each grid line in seconds. The equivalent beats-per-minute is shown. `Divisions` sets the number of divisions between each grid spacing line. These are shown as dotted lines. `Snap markers` ensures that whenever the edit markers are moved, they are snapped to the closest grid line. Note that if Edit | Marker | Snap To Zero-Crossing is checked, markers are snapped to the grid first, then the nearest zero-crossing point.

Zoom
The `Preset` zoom value is used to for the View | Preset command. See View | Specify for more information about zoom values. The `Initial` zoom specifies the zoom level to use when a file is opened. If you choose `All`, then the entire waveform is shown using whatever zoom level is required.

Miscellaneous
If the "Always confirm before saving" option is checked, you are asked to confirm saving whenever you use the File | Save command.

If the "Draw overview graph" option is checked, then the overview is graphed based on audio from the file. Otherwise it is drawn as simple lines, which is quicker since the entire file does not have to be scanned.

If the "Use left and right mouse button selection method" option is checked, then the start marker is set by using the left mouse button and the finish marker is set by using the right mouse button. No context menu appears. You will not be able to use the new right-click-and-drag features. This is the way older versions of GoldWave worked.

If the "Update default effect settings after each use" option is checked, then the last settings you use for an effect will be used the next time the effect window is shown. Essentially the "Default" preset is automatically updated when you choose the OK button on an effect window.

If "Hide persistent playback cursor" is checked, the playback cursor is not shown when playback stops. See Navigation for more information about the playback cursor.

Window Menu Commands

These commands organize Sound windows and the Control window. See Interface Overview for additional information.

Cascade, Tile, Minimize, Arrange

`Cascade` layers Sound windows on top of each other so that their title bars are visible. `Tile Horizontally` arranges Sound windows above and below each other (or side by side if necessary) so that all windows are visible. `Tile Vertically` arranges Sound windows side by side (or above and below each other if necessary) so that all windows are visible. `Minimize All` minimizes all Sound windows so only a small title bar is shown. `Arrange All` arranges minimized Sound window in rows on the bottom of the Main window.

Classic, Horizontal, Vertical Control

`Classic Control` places the Control window in the bottom right corner of the screen and arranges the controls and visuals in a square layout similar to previous versions of GoldWave. The Control window can be resized in any direction. `Horizontal Control` places the Control window along the bottom of the Main window and arranges the controls and visuals horizontally. Makes the visuals small. The Control window cannot be resized vertically in this mode. `Vertical Control` places the Control window along the right side of the Main window and arranges the controls and visuals vertically. Makes the visuals large. The Control window can be resized in any direction.

Window List

A list of all currently opened Sound windows is given at the bottom of the Window menu.

Help Menu Commands

Contents

Starts Window's Help and gives a list of contents for GoldWave help.

Manual

Starts the default web browser to display the GoldWave manual (this file).

About

Displays GoldWave version and registration information.

Command Line Parameters

Command line parameters change the way GoldWave starts when run from the Command Prompt. Also use the parameters when manually associating a file type with GoldWave. A parameter must be prefixed with either - (minus) or / (forward slash).

Syntax:
`"C:\Program Files\GoldWave\GoldWave.exe" -parameter "filename" ...`

Quotes are required around the program path and the filename. Parameters are not case sensitive. Multiple filenames and parameters can be given. Wildcards (*, ?) are supported in filenames. All files are opened or processed.

Table: Command Line Parameters

Parameter	**Purpose**
Close	Closes the program after playing the file on the command line. This parameter must be used with `Play`. If the user interacts with the program, it may not be closed. **Example** `"C:\Program Files\GoldWave\GoldWave.exe" -play -close "C:\My Music\Song.mp3"`
Config	Displays the GoldWave Setup Options window to configure core functionality in GoldWave. This is the same as using **Start | Programs | GoldWave | GoldWave Setup**. Use this only when the program is not functioning correctly due to hardware or operating system limitations or problems. Buttons to backup and restore all GoldWave settings and presets to and from a separate file are given on this window.
Clipboard:`filename`	Copies the specified file to the clipboard for batch processing. This can be used only with `Process`. If the filename contains spaces, double quotation marks must enclose the entire parameter.

Example

```
"C:\Program Files\GoldWave\GoldWave.exe" -process:Mix "My Music.wav" "-clipboard:My Vocals.wav"
```

New(`duration`,`rate`,`channels`):`filename`

Creates a new file using the default File | New settings if no other information is given. Use this with `Record` to start recording in a new file from the command line.

The duration, sampling rate, channels, and filename may be specified on the command line as shown. The default save format is used when saving the file.

When specifying a new filename on the command line, the file is overwritten without warning!

Example

```
"C:\Program Files\GoldWave\GoldWave.exe" -new
"C:\Program Files\GoldWave\GoldWave.exe" -new:Sample.wav
"C:\Program Files\GoldWave\GoldWave.exe" "-new(60):One Minute Mono.wav"
"C:\Program Files\GoldWave\GoldWave.exe" "-new(300, 48000, 2):Five Minutes DAT.wav"
```

Nosplash

Suppresses the splash window when starting the program.

Outfolder

Specifies the destination folder for all processed files for batch processing, overriding the batch preset Folder settings. If the folder name contains spaces, double quotation marks must enclose the entire parameter. If the folder hierarchy is to be preserved, then

	that option must be selected in the preset. This can be used only with `Process`.
Play	Plays the file given on the command line. Use `Close` to automatically close the program after the file has finished playing. Only the first file on the command line is played.
Portinit	Extracts all settings from the registry and saves them in an XML file in the same folder as the program itself. Use this to create a portable installation of GoldWave on a flash drive or other portable drive, so the program can be used on any computer without reinstallation. To create a portable installation: 1. Install GoldWave on a flash drive or other removable storage drive. GoldWave will install settings in the Windows registry initially. 2. Run GoldWave from that drive with the -portinit command line parameter to copy registry settings to an XML file on the drive. If GoldWave finds an XML file in the program file's folder, it loads and saves settings from that file and does not use the Windows registry. Note that some plug-ins may still use the Windows registry and may not function correctly.
Process Proclog	Starts GoldWave in batch processing mode. Most other parameters are ignored. See File | Batch Processing for details.
Record	Start recording in the file given on the command line or in a new file.

Use with `New` to create a new file for recording.

Example
Opens the program and starts recording in a new file:

```
"C:\Program
Files\GoldWave\GoldWave.exe"
-record
```

Example
If Auto save is enabled, then the following command line creates a new 3 minute file, records, saves it, then closes the program:

```
"C:\Program
Files\GoldWave\GoldWave.exe"
-new(180):newfile.wav
-record -close
```

When specifying a filename on the command line, an existing file with that name is overwritten without warning!

Region:`Start`,`Length`

Sets the initial selection for the first audio file given on the command line. `Start` is the starting point in samples. `Length` is the number of samples to select. If either `Start` or `Length` contain a decimal point, then both values are assumed to be units of time rather samples.

Examples

```
"C:\Program
Files\GoldWave\GoldWave.exe"
-region:10.0,7.5 music.wav
"C:\Program
Files\GoldWave\GoldWave.exe"
-region:441000,330750
music.wav
```

Same

Opens a file in the current running instance of GoldWave rather than starting another copy of the

	program. Use this when associating a file type with the program. The window of the current instance is brought to the foreground. To prevent that, use `-notop`.
Subfolders	Includes all subfolders when used with batch processing and a patterned filename is given (such as *.wav). This can be used only with `Process`. Before using this parameter, folder settings for the preset must be set appropriately under the Folder Tab of batch processing. **Example** `"C:\Program Files\GoldWave\GoldWave.exe" "-process:iTunes to MP3" "C:\My Music\*.m4a" -subfolders`

IV. General Information

Warranty, Trademarks, and Copyright

GoldWave ("the package") includes the following software and documentation:

```
GoldWave.exe      GoldWave application file
GWSpeed.dll       Accelerated multithreaded code
GoldWave.chm      GoldWave help
GoldWave.htm      GoldWave manual text
Images (*.PNG)    All images associated with this manual
GWPreset.reg      Presets and shapes
ReadMe.txt        Important information
WhatsNew.txt      Revision history
FLACFile.pig      FLAC file plug-in
OggFile.pig       Ogg Vorbis file plug-in
OpusFile.pig      Opus file plug-in
QTFile.pig        QuickTime file plug-in
WMAFile.pig       Windows Media Audio file plug-in
GWVST.pig         VST effect wrapper
```

The package is provided as is, without warranty of any kind. GoldWave Inc. shall not be liable for damages of any kind. Use of this software indicates you agree to this.

Support and Updates

The latest information and updates can be found on the GoldWave website:

http://www.goldwave.com

If you encounter any problems, please check the following information:

- Appendix E: Troubleshooting and Q&A
- The GoldWave website under Frequently Asked Questions

If a problem still cannot be resolved, please send a detailed description to the address below.

Questions, comments, and suggestions are welcome. You can find contact information here:

http://www.goldwave.com/contact.php

Postal Address
GoldWave Inc.
P.O. Box 51
St. John's, NF
CANADA A1C 5H5

Appendix A: An Introduction to Digital Audio

Digital Audio Basics

In the digital world, everything is reduced to an on or off state so that it can be stored in computer memory as a single *bit* of information: 1 or 0. Complex real world things, like images and audio, cannot be directly represented in such a simple manner. An image is rarely composed of black and white dots and audio is rarely just on or off.

Reducing images and audio to a digital state requires an analog-to-digital conversion. Instead of using just one bit of information, many bits are used to more accurately store the state. By using 2 bits, for example, four states are possible: 00, 01, 10, 11. For images, that could be black (00), dark gray (01), light gray (10), and white (11). For audio, that would give four different levels of loudness. Typically, many more bits are used. Most computer video cards use 16 to 32 bits to store a single dot. Sound cards typically use 16 bits for audio levels.

The number of bits to use depends on human perception and bit alignment within computers. Computer tend to bundle bits in groups of 8, called bytes, so using 8, 16, 24, or 32 bits would fit nicely in 1, 2, 3, or 4 bytes respectively. For images, 16 bits do not provide enough states to make the transition from one state to the next imperceptible, so 24 or more bits are used. For audio, 16 bits are adequate, which is what a CD contains, but audio systems using 24 bits will be common in the future.

Samples

Digital audio is composed of thousands of numbers, called samples. Each sample holds the state, or amplitude (loudness), of a sound at a given instant in time. For images, each point of light, or pixel, has a certain brightness and location and all pixels combine to make a picture (see figure below). For digital audio, all the samples combine to make a waveform of the sound.

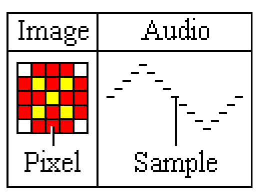


Figure: Samples

When playing audio, each sample specifies the position of the speaker at a certain time. A small number moves the speaker in and a large number moves the speaker out. This movement occurs thousands of times per second, causing vibration, which we hear as sound.

Digital Audio Attributes

There are several attributes that determine the quality and size of a digital audio file. They are the sampling rate, the bit depth, the number of channels, and the bitrate.

Sampling Rate

The *sampling rate* is the number of times, per second, that the amplitude level (or state) is captured. It is measured in Hertz (seconds^{-1}, Hz). A high sampling rate results in high quality digital sound in the same way that high resolution video shows better picture quality. Compact disks, for example, use a sampling rate of 44100Hz, whereas telephone systems use a rate of only 8000Hz. If you've ever heard music on the telephone while on hold, you'll notice a big difference in quality when compared to the original music played on a CD player.

Higher sampling rates capture a wider range of frequencies and maintain a smoother waveform. The figure below shows a real world waveform in red and the digital waveform in black at different sampling rates. You can see that increasing the sampling rate makes each step of the digital waveform narrower. The shape more closely follows the real world. In general, the height of each step is reduced as well, but that depends on the number of bits. In simple terms, the sampling rate controls the width of each step.

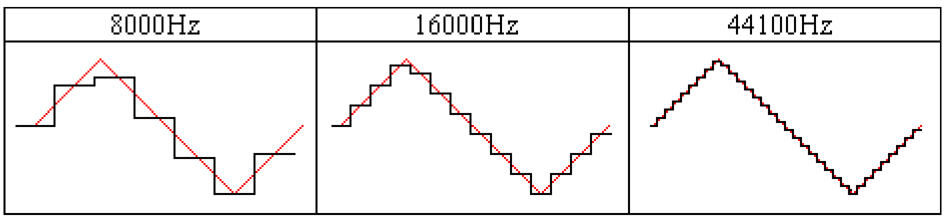


Figure: Sampling Rate

The rate to use depends upon the type of sound and the amount of storage space available. Higher rates consume a lot of space. In the above example, the CD requires over 5 times the amount of storage as the telephone system for the same digital sound. Certain types of sounds can be recorded at lower rates without loss of quality. Some standard rates are listed in the table.

Table: Standard Sampling Rates

Attributes	**Quality and Usage**	**Storage (16 bit/mono, MB/minute)**
8000Hz	Low quality. Used for telephone systems. Good for speech. Not recommended for music.	0.960
11025Hz	Fair quality. Good for speech and AM radio recordings.	1.323
22050Hz	Medium quality. Good for TV and FM radio quality music.	2.646
44100Hz	High quality. Used for audio CDs.	5.292
48000Hz	High quality. Used for digital audio tapes (DAT).	5.760

96000Hz	Very high quality. Used for DVD audio.	11.520

Bits

As explained in the Digital Audio Basics section, the number of *bits* determines how accurately the amplitude of the waveform is captured. The figure below shows a real world waveform in red and the corresponding digital waveforms with 2 bit samples and 3 bit samples.

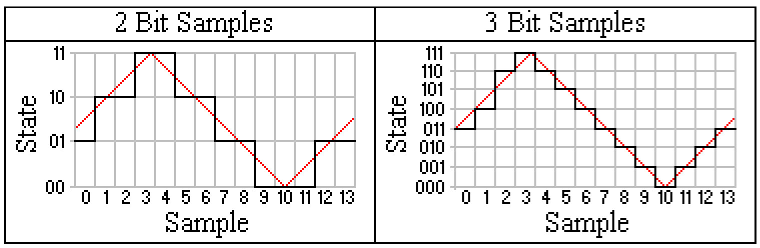


Figure: Bits

You can see that adding a single bit greatly improves the way the digital waveform conforms to the real world waveform. The 2 bit waveform looks like a rough approximation with large steps. Several amplitudes are rounded to the same state, such as samples 9 through 11. This is a source of quantization noise, explained later.

In the 3 bit waveform, no amplitudes are rounded to the same state. Each step is half the height of the 2 bit waveform, but it is still not perfect. From sample 1 to sample 2, there is a jump in the waveform, which also causes quantization noise to a much lesser extent. You'll notice that samples 0 and 1 are below the real waveform and samples 2 and 3 are above the waveform. This occurs because there are no in-between states to accurately store those amplitude levels, so the digital waveform ends up straddling the real one. Therefore more states, and bits, are needed.

8 bit and 16 bit samples are common. In an 8 bit sample, there are 256 different states or levels of amplitude. 16 bit samples have 65,536 levels. This makes a huge difference it terms of sound quality. Audio stored as 8 bit samples will often have much more quantization noise.

Signed and Unsigned

Samples can be stored as bits a couple of different ways. One way is to consider all the states as positive, with no values below zero. As shown in the figure above, the states 00, 01, 10, and 11 are the same as the positive numbers 0, 1, 2 and 3. This eliminates the need for a negative sign. Such samples are called *unsigned*. For 8 bit samples, the states would range from 0 to 255.

The other way is to use a form known as *two's complement*, which allows both positive and negative values. These samples are called *signed*. Since real world waveforms tend to fluctuate through a range of positive and negative values, signed samples are preferred. For 16 bit samples, the states would range from -32678 to 32767.

Big and Little Endian

When a sample is stored using more than 8 bits, more than one byte is needed. The term *endian* is used to describe the way bytes are ordered in computer memory. It specifies the significance of the first byte in the group. A 16 bit sample, for example, requires exactly two bytes, byte A and byte B. They can be stored as A first, then B or as B first, then A. Generally a PC will store them one way and a Mac will store them the other way due to differences in the internal processor design of those systems.

Big endian order has the most significant byte stored first, making it similar to the way we read numbers. In the number 47, the 4 is first and is most significant and the 7 is last and is least significant. This ordering is used on Mac systems.

Little endian order has the least significant byte stored first, allowing some optimizations in processing. This ordering is used on Intel and PC systems.

Channels

Digital audio can have one or more channels. Single channel audio, referred to as a monaural (or mono) audio, contains information for only one speaker and is similar to AM radio. Two channel audio, or stereo audio, contains data for two speakers or two ears, much like FM stereo. Stereo sounds can add depth, but they require twice as much storage and processing time as mono sounds. Most movie theatres have advanced audio systems with 4 or more channels, which are capable of making sounds appear to come from certain directions. GoldWave currently supports mono and stereo audio only.

Digital Audio Limitations

Since digital audio is limited by the sampling rate, the number of bits, and the number of channels, a digital waveform can never be an exact replica of the real world waveform. These limitations can lead to a number of problems, such as aliasing, clipping, and quantization noise.

Aliasing

Aliasing occurs when the sampling rate is not high enough to correctly capture the shape of the sound wave. The recorded sound will have missing tones (Figure: Aliasing, top) or new tones that never existed in the original sound (Figure: Aliasing, bottom). These problem can be eliminated by using higher sampling rates or by using anti-aliasing filters.

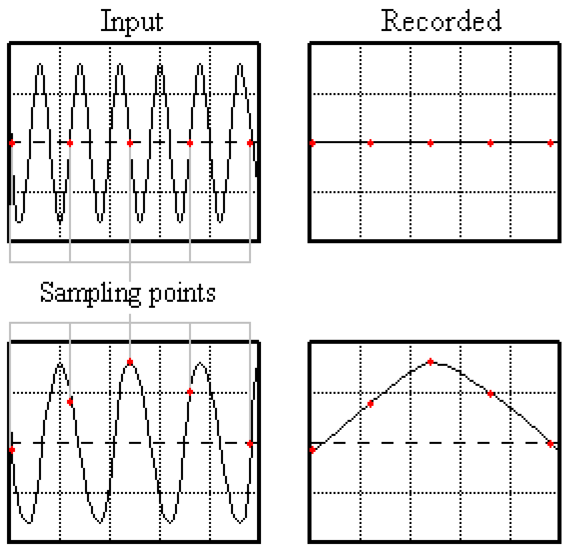


Figure: Aliasing

Higher sampling rates increase the number of sampling points. To see how this works, try adding a three points between each sampling point in the figure and redraw the graph. The recorded sound will more closely resemble the input.

Anti-aliasing filters remove all tones that cannot be sampled correctly. They prevent high pitched tones from being aliased to low pitch. Many sound cards include anti-aliasing filters in hardware.

Clipping

Clipping errors occur when the level is too high to be stored in the bits available. For example, if the maximum level for a 16 bit sample is 32767, and the actual level is 40000, then it must be clipped to 32767 to fit (see the figure). This generates distortion. To eliminate clipping, adjust the recording volume before recording. By using the Control window's Monitor input on visuals feature, you can adjust the volume to a suitable level. The volume is low enough when the VU Meter visual does not reach the top of the red region.

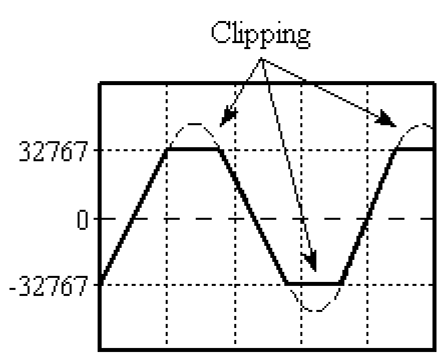


Figure: Clipping

Clipping can occur when processing effects, such as increasing the volume with the Change Volume effect. GoldWave does not clip audio internally, but it has to be clipped when sending the audio to the sound card or when saving a file. You should use the Maximize Volume effect with a 0dB setting before saving to ensure that no clipping occurs in the file.

Quantization

The real world has an infinite number of states, but the digital world has a very limited number of states. *Quantizing* is process of assigning an amplitude to this limited quantity of states. *Quantization noise* occurs when an amplitude is rounded to the nearest state for the given number of bits. This is illustrated in the Bits section. Using 2 bits causes large rounding errors and only roughly follows the real waveform. Using 3 bits has much smaller rounding errors, but is still nowhere near as smooth as the real world waveform. Using 8 bits gives a much smoother waveform, but the round errors can still be heard. Using 16 bits gives errors so small that they are almost imperceptible.

Noise

To minimize internal and external noises, make sure your sound card is installed as far away from your graphics card as possible. If the sound hardware is integrated into the computer's main board, consider purchasing and installing a separate sound card instead. Integrated hardware tends to have a much higher noise level.

Keep all microphones and audio cables away from your monitor, computer case, or other sources of electrical noise. Use shielded cables and make sure everything has a common ground connection. Plug equipment and devices into the same power strip so that they share the same ground. Equipment without a grounded plug (three pronged plug) may require special grounding. Sometimes rotating the plug helps reduce a hum or buzz.

System configuration can also affect audio quality. Due to architectural problems with PCs and excessive virtual memory swapping by Windows, you may notice an occasional gap when recording. Restarting Windows, updating the sound driver, shutting down all other programs, or installing more RAM helps to minimize that problem.

Volume Scales

The waveform levels or states can be interpreted in a number of different ways. Reading the states in their binary form is impractical, so the states are mapped to different scales that are more human-readable. These include a simple amplitude scale, a percentage scale, and a decibel scale. As the figure shows, a 1.0 amplitude level is the same as a 100% level and a 0dB level.

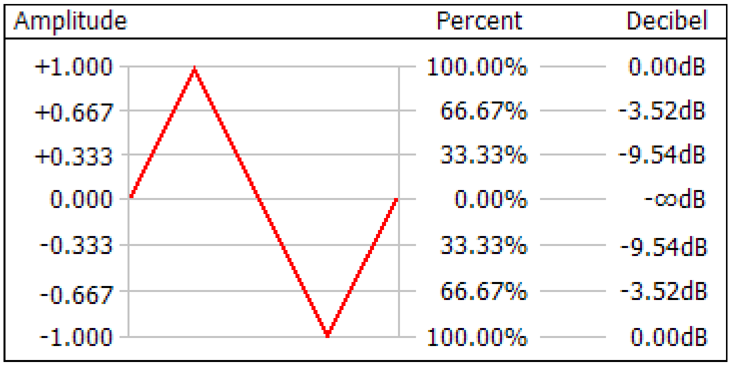


Figure: Volume Scales

Amplitude (y)

The amplitude scale simply maps the states to a linear range of -1.0 to +1.0, with zero being silence. The amplitude typically is given as a positive value when used in effects.

Percent (%)

The percent scale is 100% at the maximum and 0% at silence. It is essential the same as the absolute value of the amplitude scale converted to a percentage by multiplying by one hundred and adding a percent sign. Sometimes a negative percentage is used, such as in the Flanger effect, to invert the waveform.

Decibel (dB)

The decibel scale (specifically dBFS) is unusual in that it is 0dB at the maximum peak level and negative infinity at silence. It is a logarithmic scale, which is closer to the way human hearing perceives sound levels. You'll notice from the above figure that there are no positive levels. Levels below the maximum are negative. Only values above the maximum are positive (not shown) and such values may cause clipping. When changing the volume, positive values increase the level and negative values decrease the level.

Levels shown in most visuals are dBFS and not dB_{SPL}.

Use the equation below to convert a decibel level to a percentage level.

$$percentage = 10^{\frac{gain(dB)}{20}} \times 100\%$$

Relative vs. Absolute

When setting the volume in an effect, the value may be interpreted as a relative level or an absolute level. When changing the volume, it is usually relative. When specifying a threshold, it is usually absolute. Relative changes are cumulative. So if you apply a volume change with 0.5 amplitude (50% or -6.02dB), then the amplitude decreases to half its current level. If you apply that change again, then it decreases to one quarter of its original level. In other words, the change is relative to its current level. Given an original amplitude of A, the first change yields a result of A x 0.5 and the sample is replaced by that value. The second change takes that value and multiplies it by 0.5 again, so we get the final result of (A x 0.5) x 0.5 or A x 0.25. Most of the effect settings in GoldWave are cumulative. For relative changes, using 1.0, 100%, or 0dB does not alter the sound at all.

Absolute levels are user for thresholds, such as in Silence Reduction, or in rare cases were the absolute level is set directly, such as Maximize Volume. Absolute changes are not cumulative. If you maximize the sound with 0.5 amplitude (50% or -6.02dB), then that is what the peak level will be no matter how many times the effect is used. For absolute changes, using 1.0, 100%, or 0dB may alter the sound if it is not currently at that level.

Frequency and Pitch

Sound windows in GoldWave show sound as a waveform of amplitudes on a time axis. However, sound can be viewed in an entirely different way by examining its frequency and pitch content or *frequency spectrum*. Sounds are broken down into a combination of simple fundamental (sinusoidal) tones, each with a different frequency. This is useful for examining bass and treble levels or for isolating and studying certain sounds.

Frequency Ranges

Average human hearing spans a frequency range from about 20Hz to about 17000Hz. The figure below shows some common sounds and the frequency range they cover.

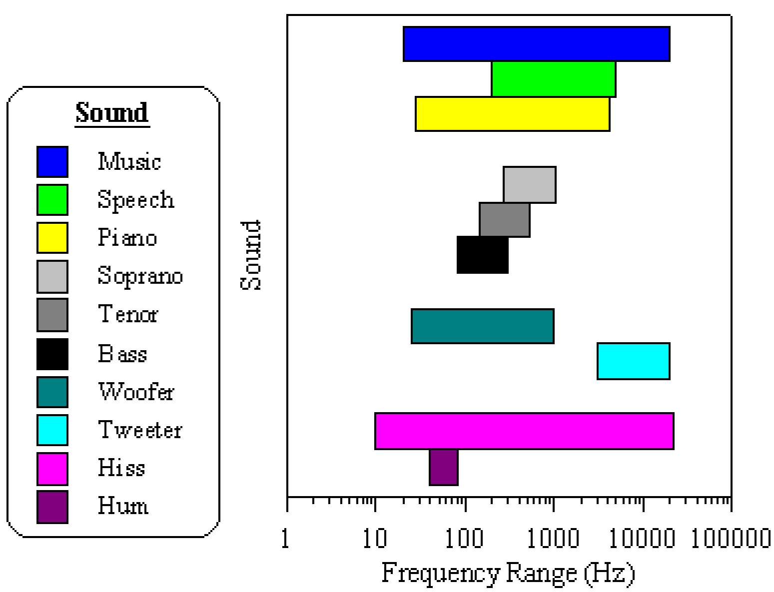


Figure: Frequency Range of Sounds

Many people wonder why it is difficult to remove vocals from music. From the figure, you'll see there is a large overlap in the frequency range of speech and music. Removing the vocals would also remove a significant part of the music. A similar problem occurs when removing hiss noise, since it often covers the entire spectrum.

Most basic stereo systems have bass and treble controls, which offer limited control over a frequency spectrum. Bass applies to low frequency sounds, such as drums, cellos, low piano notes, or a hum noise. Treble applies to high frequency sounds, such as a clash of cymbals, a tweet of a small bird, high notes on a piano, or a hiss noise.

More expensive stereo systems have Graphic Equalizers, which provide better control over a frequency spectrum. Instead of controlling just two bands (bass and treble), you can control many bands.

GoldWave provides even more control over frequency spectrums with filter effects such as Parametric EQ, Low/Highpass, Bandpass/stop, Equalizer, Noise Reduction, and Spectrum Filter.

Frequency Range and Sampling Rate

The frequency range of a digital sound is limited by its sampling rate. In other words, a sound sampled at 8000Hz cannot record frequencies above 8000Hz. In fact, the sound cannot even have frequencies above 4000Hz. According to the sampling theorem, the maximum frequency is limited to half the sampling rate. Any higher frequencies will be aliased, to lower ones, causing noise if appropriate filters are not used.

CD audio is designed to cover the full range of human hearing, which has a maximum of under 22kHz. In order to successfully record this range, the sampling theorem states that a sampling rate of at least twice the maximum must be used, so a rate of at least 44kHz is required. The actual rate is 44100Hz for standard CD players.

Frequency Spectrum Graphs

Several of GoldWave's Control visuals convert sounds into a range of frequency bands using a radix-2 fast Fourier transform (FFT) algorithm. When the results are drawn using colours, the graph is referred to as a spectrogram. When the results are drawn with lines, it is often referred to as a frequency spectrum or *frequency analysis*.

Frequency analysis graphs are displayed in the Noise Reduction, Spectrum Filter, and Parametric EQ filter effects. These help you to locate frequencies that you want to remove or enhance.

GoldWave applies a windowing function to the data before performing the FFT (see Control Visual Properties section). This reduces "discontinuity" errors that occur when dividing data into small chunks. A Kaiser window is used by default.

To make the spectrum more realistic to human hearing, magnitudes are scaled logarithmically. This means that if one frequency "sounds" twice as loud as another, it is graphed with twice the height (or the corresponding colour for the spectrogram).

File Compression

Uncompressed audio files tend to be large. CD quality audio requires ten megabytes per minute. That is not a problem with large computer hard drives available today, but it is a problem if you want to save many songs on a portable player or if you want to transfer files over the Internet. Unlike most computer data, audio data does not compress very well using typical compression methods such as those found in programs like PKZIP or WinZip. These methods preserve the data exactly so there is no loss of quality. Such compression is called *lossless* compression.

To make audio files smaller, complex algorithms have to used. Most of these algorithms sacrifice some quality so that when the data is decompressed, you do not get exactly the same quality you had originally. This type of compression is known as *lossy* compression. Ideally the quality that is lost is not perceptible, so you do not notice the difference.

The most common method of lossy audio compression is MPEG Layer-3, better known as MP3. It is capable of getting near CD quality audio in less than one tenth the size, which is about one megabyte per minute. More recent algorithms, such as AAC, Ogg Vorbis, and Windows Media Audio get even better quality in a smaller size.

Software and hardware that compress audio using complex algorithms are referred to as *codecs* (from **co**der/**dec**oder). Compression is the same as encoding and decompression is the same as decoding.

To control the level of compression in GoldWave, use File | Save As and select a different file type and/or attributes. The lower the bitrate (the *kbps* number, see below), the smaller the file will be, usually at reduced quality.

When opening, editing, and saving a file repeatedly, it is best to use a lossless file format. Every time a lossy compressed file is re-opened and saved, some quality is lost. For long term editing, use the Wave type with "PCM 16 bit" attributes or one of the lossless compressed formats such as FLAC or Windows Media Audio with "lossless" attributes.

Bitrate

Many compressed audio formats measure the compressed size as a *bitrate*. The bitrate is the number of bits per second (bps) required to store the audio. Usually the number is given in kilobits or one thousand bits. Divide that number by 8 to determine the number of kilobytes required per second.

Internet connection speed (bandwidth) is often measured in bitrates as well. A 56k modem is capable of receiving 56 kilobits per second. If you want people to stream your MP3 audio over a modem, you'll need to compress the file using a maximum bitrate of 56kbps. Due to the connection overhead and Internet protocol, a lower rate would have to be used to ensure the audio can be downloaded fast enough. For DSL and Cable Internet connections, the standard 128kbps MP3 rate can be used.

Audio files can contain a wide range of sounds, from noisy cymbal clashes to silence. Algorithms typically get much better compression on silence or simple audio sections than on complex, noisy audio. This means that the bitrate depending on whether constant bitrate or variable bitrate compression is used.

Constant Bitrate

When using constant bitrate, each section of audio compresses to exactly the same size, regardless of the content. If the audio contains silence, then the data may be padded to fill the required bitrate. If the audio contains complex music, then quality may be decreased until it fits within the bitrate.

Constant bitrate is useful for broadcast systems where the transmission rate is fixed. It also make is easy to seek to arbitrary positions within the audio stream or file.

Variable Bitrate

Variable bitrate compression uses the smallest size possible for each section of audio. If the audio contains silence, then the bitrate will be very low. If the audio contains complex music, the bitrate will be at its maximum.

Variable bitrate gives the best compression and quality. However, it makes it difficult to seek within the stream or file since there is no direct relation between time and size.

Appendix B: Keyboard Commands

In addition to all the standard menu keystrokes, such as **Alt+F O** to open a file, **Alt+E C** to copy, **Alt+E P** to paste, etc., GoldWave includes a number of additional keyboard shortcuts. These are summarized in the following table. Use Options | Keyboard to reassign shortcuts.

Keystroke	**Action**
Ctrl+A	Selects the entire sound.
Ctrl+B	Pastes the clipboard into the sound at the beginning.
Ctrl+C	Copies the selection into the clipboard.

Ctrl+D	Displays the Crossfade window. Crossfades the clipboard with the selection.
Ctrl+E	Pastes the clipboard into the sound at the end.
Ctrl+F	Pastes the clipboard into the sound at the finish marker's position.
Ctrl+G	Displays a window to set the playback cursor's location to a specific time.
H	Starts playback relative to the mouse's horizontal position in the waveform.
J, K, L	Rewinds, plays, and fast forwards respectively from the current cursor's position. Playback stops at the finish marker but can be continued. Playback stops at the start marker when rewinding.
Shift+J, Shift+K, Shift+L	Makes the playback speed slower, normal, and faster respectively.
Ctrl+J	Jumps the start marker to the next cue point.
Ctrl+Shift+J	Jumps the start marker to the previous cue point.
Alt+J	Jumps the finish marker to the next cue point.
Alt+Shift+J	Jumps the finish marker to the previous cue point.
Ctrl+K	Overwrites the selection with the clipboard.
Ctrl+M or Shift+Ctrl+Ins	Displays the Mix window. Mixes the clipboard with the sound at the start marker's position.
Ctrl+N	Creates a new sound.
Ctrl+O	Opens a sound.
Ctrl+P	Pastes the clipboard into a new Sound window.
Q	Drops a new cue point at the start marker position. Cue naming is controlled by the Auto Cue settings under the Cue Points tool.
Shift+Q	Drops a new cue point at the finish marker position. Cue naming is controlled by the Auto Cue settings under the Cue Points tool.
Ctrl+Q	Drops a new cue point at the current recording, playback, or start marker position. Cue naming is controlled by the Auto Cue settings under the Cue Points tool.
Ctrl+Shift+Q	Drops a new cue point at the current recording,

	playback, or start marker position and displays the edit window.
Ctrl+R	Replaces the selection with the clipboard contents.
Ctrl+S	Saves the file.
Ctrl+T	Trims the sound. Removes all audio outside the selection.
Ctrl+U	Displays the AutoTrim window. Removes silences within the ends of the selection.
Ctrl+V	Pastes the clipboard into the sound at the start marker's position.
Ctrl+W	Sets the select to the view (as in Select View).
Ctrl+X	Cuts the selection and copies it into the clipboard.
Ctrl+Y	Redo. Reverses last undo.
Ctrl+Z	Undoes last change.
Ctrl+Shift+B	Selects both channels of a stereo file.
Ctrl+Shift+L	Selects the left channel of a stereo file.
Ctrl+Shift+R	Select the right channel of a stereo file.
Shift+0	Zooms 10:1 horizontally.
Shift+1	Zooms 1:1 horizontally.
Shift+2	Views 1 second horizontally.
Shift+3	Views 10 seconds horizontally.
Shift+4	Views 1 minute horizontally.
Shift+5	Views 1 hour horizontally.
Shift+A	Horizontally zooms all the way out.
Shift+E	Displays the Set Marker window.
Shift+M	Stores the locations of the start and finish markers.
Shift+P	Zooms to previous horizontal zoom.
Shift+R	Moves the start and finish markers to the stored locations.
Shift+S	Horizontally zooms in on the selection.
Shift+U	Horizontally zooms to the user defined level.
Shift+V	Vertically zooms all the way out.
Shift+Y	Displays window to specify horizontal zoom.

Del	Deletes the selection, permanently.
[(left bracket)	Drops the start marker at the current playback position.
] (right bracket)	Drops the finish marker at the current playback position.
Shift+[	Plays three seconds of audio up to the start marker.
Shift+]	Plays three seconds of audio up to the finish marker.
Ctrl+[	Plays from the start marker to the finish marker.
Ctrl+]	Plays from the finish marker to the end.
Shift+\	Moves the start and finish selection markers to their previous positions.
Left	Scrolls the Sound window view left.
Right	Scrolls the Sound window view right.
Page Up	Scrolls the Sound window view left one page. The amount of time scrolled depends on the current view zoom level. If you used a play button in view mode to play the file, playback is restarted at the new scrolled position.
Page Down	Scrolls the Sound window view right one page. See the Page Up key for more details.
Home	Moves the Sound window view to the start marker's position.
End	Moves the Sound window view to the finish marker's position.
Ctrl+Home	Moves the Sound window view to the beginning of the sound.
Ctrl+End	Moves the Sound window view to the end of the sound.
Shift+Home	Moves the start marker to the beginning of the sound.
Shift+End	Moves the start marker to the finish marker's position.
Ctrl+Shift+Home	Moves the finish marker to the start marker's position.
Ctrl+Shift+End	Moves the finish marker to the end of the sound.
Shift+Left,	Moves the start marker left or right.

Shift+Right	
Ctrl+Shift+Left, Ctrl+Shift+Right	Moves the finish marker left or right.
Shift+Up	Horizontally zooms in.
Shift+Down	Horizontally zooms out.
Ctrl+Up	Vertically zooms in.
Ctrl+Down	Vertically zooms out.
Space	Plays or stops a sound (toggles playback). The region that is played depends on the play 3 button settings. Use Ctrl for play 1 and Shift for play 2.
F2, F3, F4, F5, F6, F7, F8	Plays 1, plays 2, plays 3, rewinds, fast forwards, pauses, and stops respectively.
Ctrl+F9, Ctrl+F8, Ctrl+F7	Starts, stops, and pauses recording respectively.
F11	Displays Device Controls Properties window.
F1	Starts help.
Ctrl+F4	Closes the Sound window.
Alt+F6	Switches between Main window and Device Controls window.
Ctrl+F6	Switches between Sound windows.

Appendix C: Expression Evaluator

Overview

The Expression Evaluator is a versatile tool for manipulating and generating audio data.

The `Destination` box specifies the Sound window where results of the evaluation will be stored. The drop down list contains all Sound windows in the form "*X* - *Filename*", where *X* is the wave identifier number of the Sound window. For example, a Sound window with the title "Hello.wav" could appear as "1 - Hello.wav" in the list. By default, the destination is set to the current Sound window. You can change the destination, if more than one Sound window is opened, by using the up and down keys or by selecting it with the mouse from the drop down list.

The `Source` box lists currently opened Sound windows. By selecting a source from this list, the function `waveX(n)` is placed in the expression. *X* is the wave identifier number, as explained above. Note that if the source and destination are the same, then the source changes during processing.

The large `Expression` box located at the top of the window is where an expression is entered. A list of valid operations and functions is given in a following section. In most cases, expressions will be some function of *n* or *t*, just as in regular math, where *y* is usually a function of *x* (i.e. *y = f(x)*). In the expression evaluator, we can have *destination = f(t)*, where *f(t)* is any expression you enter.

To create a simple tone, for example, you would enter the expression `sin(600*t)`. You can even alter an existing sound. To double the volume of "1 - Hello.wav", for example, you would select it as the destination and enter the expression `wave(n)*2`.

To enter an expression, you can:

- Type it in using the keyboard, or
- Select it from the Presets tree list

The evaluator uses several special variables, which you can initialize in the "Incremented variables" and "User constants" groups. These variables are discussed later.

After you have specified the destination, expression, and initial values, choose the Play button to preview it. Choose the OK button (or just press the `Enter` key) to begin evaluation. If you entered an expression incorrectly, a message will be displayed. If the expression is valid, a processing window will appear. Since the evaluation process takes time, you can stop it at any time with the Cancel button. No changes will be made to the sound if you cancel processing.

You can copy, cut and paste expression in the Expression box using the usual keystrokes (Copy = Ctrl+C, Cut = Ctrl+X, Paste = Ctrl+V). You can also copy and paste expression from the online help.

To speed up evaluation, make sure that you are using RAM storage (see Options | Storage).

Evaluation Range, Variables, and Constants

Knowledge of the structure of digital audio is essential to understand how the evaluator works. To illustrate this structure, let's assume we have the following sound:

```
Title bar:           Hello.wav
Total length:        2.0 seconds
Sampling rate:       8000Hz
Start marker:        0.5 seconds
Finish marker:       1.2 seconds
```

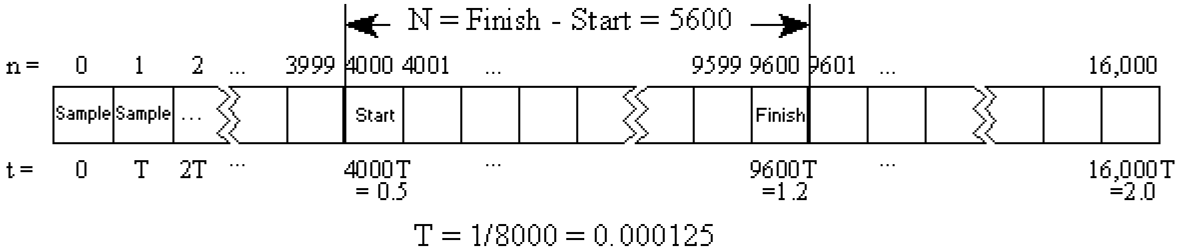


Figure: Internal Structure of Hello.wav, 8000Hz, 2 Seconds

Digital audio is stored as a series of amplitudes, which are often referred to as samples (see figure). The evaluator interprets each sample as a value between -1 and 1, inclusive. Only samples between the start and finish markers are considered valid; all other values are assumed to be zero. The number of samples selected is defined as *N*.

Each sample has a relative index number, *n*, and a time, *t*. Since the time of each sample depends on the sampling rate, it is usually written in terms of the unit of time between each sample, *T*. You many have noticed that the time, *t*, is related to the index number, *n*, by the equation *t*=*nT*. The figure shows how all these variables relate to the structure of the sound.

Using Time in an Expression

Let's assume we have entered the expression "`sin(t)`". Since expressions are evaluated over the selection range, the initial value for *t* is automatically set to start marker's position of 0.5. Choosing the OK button evaluates the expression from *t* = 0.5 to *t* = 1.2 in steps of 1/8000 of a second, as defined by *T*. This means that the expression is calculated for each sample in the selection, changing each sample as follows:

Sample_{4000} = sin(0.500000)
Sample_{4001} = sin(0.500125)
Sample_{4002} = sin(0.500250)
...
Sample_{9600} = sin(1.200000)

Using the Sample Index in an Expression

The sample index is useful for modifying an existing sound. If we want to double the amplitude of Hello.wav, for example, we need to multiply each sample by two and store it back into the sound. In this case, Hello.wav will be both the destination and the source. To set it as the destination, we simply select it from the `Destination` list. To use it as a source, we need to determine its wave identifier number. These numbers are provided in the `Source` list. Assuming it is listed as "3 - Hello.wav", we now know that its wave identifier number is 3. This number is necessary for the evaluator's `wave` function, which has the following syntax:

`waveX( n )` where: `X` = wave identifier number `n` = sample index number

Note that when the destination and the source are the same, no number is needed after the `wave` function. If "`wave(n)`" is used, then the destination sound is assumed to be the source.

In the evaluator, the index number, *n*, is *relative* to the start marker. This means that the start markers position is added to the index number (i.e. *n*+Start). For the example in the figure above, a relative index of *n*=0 has an absolute index of 4000.

The final expression is "`wave3(n)*2`". Choosing OK evaluate this expression from *n*=0 to *n*=5600 in steps of 1 (note that *N = 5600*) . This produces the following changes (remember than *n* is relative to the start marker's position):

Sample$_{4000}$ = Sample$_{4000}$ * 2
Sample$_{4001}$ = Sample$_{4001}$ * 2
Sample$_{4002}$ = Sample$_{4001}$ * 2
...
Sample$_{9600}$ = Sample$_{9600}$ * 2

Note that *N* and *n* are always integers. The evaluator rounds indices to the nearest integer, so the expression "`wave3(.7)`" would be calculated as "`wave3(1)`".

You can use the sample index number and the `wave` function to mix two or more wave together. If you have several sounds opened, you can obtain the wave identifier number for each sound from the `Source` list. If the sounds you wanted to mix were identified as 2 and 3, you would enter the expression:

wave2(n) + wave3(n)

Care must be taken when indexing signals with different sampling rates. If `wave1` is a voice recorded at 11025Hz and `wave2` is music recorded at 22050Hz, then with `wave1` as the destination you'd need to use this expression to correctly mix them:

wave1(n) + wave2(n*2)

Ideally, `wave2` would have to be low-pass filtered first. If `wave2` is the destination, the expression would be:

wave1(n/2) + wave2(n)

Also be aware that if the source and destination are the same, the source is modified during evaluation. An expression such as `wave(n)+wave(n-3000)/3` generates a recursive reverb effect because `wave(n-3000)` gives the stored, modified values instead of the original source values. This allows for feedback processing. Use separate source and destination sounds to avoid modifying the source.

The variable *N* has several uses, such as reversing a sample. If `wave2` is a new sound that has the same sampling rate and length of `wave1`, then setting the destination to `wave2` and using the expression

wave1(N-n)

makes `wave2` the reverse of `wave1`.

User Constants x, y, and f

User constants can be set to any values you choose. None of these values change during evaluation. They just provide a way of easily changing parameters within an expression without having to edit the expression directly. The constant *f* is typically used for frequency values. By using the expression

y*sin(2*pi*f*t)

you can generate any tone by specifying the frequency in the `f` box and the amplitude in the `y` box.

Conversion Between Variables

The following equations convert between time and sample index number. The *start* value is the position of the start marker (in seconds).

```
n = (t - start) / T
t = nT + start
T = 1 / (sampling rate)
```

Presets

The `Presets` tree list in the Expression Evaluator window organizes expressions in a number of groups, such as Dial Tones, Effects, Noise, and Waves. You can create new groups or add expression to existing groups.

To retrieve an expression:

1. Expand the group containing the expression in the `Presets` list.
2. Select the expression from the expanded list.

To add an expression:

1. Enter the expression in the Expression box.
2. Choose the add button in the `Presets` box.
3. Select an existing group name or type in a new group name.
4. Type in a preset name for the expression.
5. Choose the OK button.

To delete an expression:

1. Expand the group containing the expression in the `Presets` list.
2. Select the expression from the expanded list.
3. Choose the remove button in the `Presets` box.

Evaluator Operators and Functions

The following table summarizes evaluator operators and functions.

Table B.1: Evaluator Operators and Functions

Label	**Operation, function**
(,)	Parenthesis
+, *, -, /	Add, multiply, subtract (negate), and divide
%	Modulus operator (remainder)
<, >	Greater and less than operators. Compares left and right operands. Gives 1 if true or 0 if false. For example, `x > y` evaluates to 1 if `x` is greater than `y` or 0 otherwise. The expression `wave(n)*(wave(n)>0.001)` replace all samples below 0.001 with silence (zero).
^	To the power of, y^x
pi	Constant (3.14159...)
cos	Cosine
sin	Sine
tan	Tangent
acos	Arccosine
asin	Arcsine
atan	Arctangent
cosh	Hyperbolic cosine

sinh	Hyperbolic sine
tanh	Hyperbolic tangent
sqrt	Square root
abs	Absolute value
sgn	Sign (-1, 0, or 1).
log, ln	Log base 10, natural logarithm
exp	Exponential base e
step	Unit step (0 for t < 0, 1 for t >= 0)
int	Integer value
rand(n)	Generates a uniform random number between 0 and n
limit	Limits the value +/-1. If the absolute value is greater than 1, then its magnitude is set to 1.
wave*X*(n)	Sound amplitude at n. *X* specifies the Sound window as given in the `Source` list. If no *X* is specified, the destination Sound window data is used.

Signal Generation

Several signal generation expressions are listed below. Words given in *italics* represent numeric values that you must enter. To try one of the following expression, perform the following steps:

1. Choose **New** from the **File** menu.
2. Choose **OK** to create the new file.
3. Choose **Expression evaluator** from the **Tool** menu.
4. Type in the expression as given in the example. For example:

```
sin(2*pi*261.7*t)
```

5. Choose the OK button.
6. Wait for processing to complete.

Use the play button to preview the expression before processing the entire file.

Table B.2: Expressions

Type	General Expression	Examples
Sine wave	sin(2*pi**frequency**t)	Middle C: sin(2*pi*261.7*t) Telephone dial tone for

		"5": (sin(2*pi*1336*t) + sin(2*pi*773*t)) / 2
Saw wave	1 - 2*abs(1 - 2**frequency**t%2)	200Hz tone: 1 - 2*abs(1 - 2*200*t%2)
White noise	*amplitude* - rand(2**amplitude*)	Full volume white noise: 1 - rand(2)
Square wave	int(2*t**frequency*)%2*2-1	400Hz tone: int(2*t*400)%2*2-1
Sweep	sin(2*pi*t^*rate*)	Slow sweep up to 20kHz: sin(2*pi*160*(t%5)^3)
Exponential decay	(1 - *minimum*)*exp(-t) + *minimum*	50% decay a 500Hz sine wave: (0.5*exp(-t) + 0.5) * sin(2*pi*500*t)

Custom Filters

One way to create your own digital filter is to use Matlab™ (The Student Edition of Matlab, by The Math Works Inc., published by Prentice-Hall, ISBN 0-13-855974-0). It has many built-in commands that generate filter coefficients. The coefficients can then used in the Expression Evaluator command.

Example of a Low-Pass Filter

In preparation for down-sampling, you can use Matlab to generate the coefficients of a 4th order Butterworth low pass filter that will remove noise above half the Nyquist frequency (one quarter the sampling rate). Enter:

```
[b,a] = butter(4, 0.5)
```

The result should be similar to:

```
b =  0.0940  0.3759  0.5639  0.3759  0.0940
a =  1.0000  0.0000  0.4680  0.0000  0.0177
```

To implement this filter in the evaluator, assume that the sound to be filtered is in the Sound window titled Sound.wav.

1. Use File | New to create a new Sound window with the same sampling rate and channels as Sound.wav.
2. Make sure the length of the new sound is as long (or longer) than Sound.wav.
3. Use Tool | Expression Evaluator to open the expression evaluator window.
4. Set the destination to the new sound.

5. Enter the following expression (assuming Sound.wav has a wave identifier of 1):
6. `wave1(n)*0.0940 + wave1(n-1)*0.3759 +`
7. `wave1(n-2)*0.5639 + wave1(n-3)*0.3759 +`
8. `wave1(n-4)*0.0940 - wave2(n-2)*0.4860 -`
9. `wave2(n-4)*0.0177`
10. Choose OK to start filtering.

Appendix D: Tutorial

From Turntable to CD-R

This tutorial explains the steps required to record from a turntable, restore the audio, and split the recording into tracks so it can be burned to a blank CD.

Make Connections

Unless your turntable has a built-in amplifier or a headphone output, you'll need an external amplifier, receiver, or preamp to boost the turntable output. The figure below shows all the connections required. You may need two RCA cables with male connectors and a RCA to 1/8" mini-plug stereo Y-adapter cable.

If you have not connected your headphones or speakers to your computer, do so before proceeding.

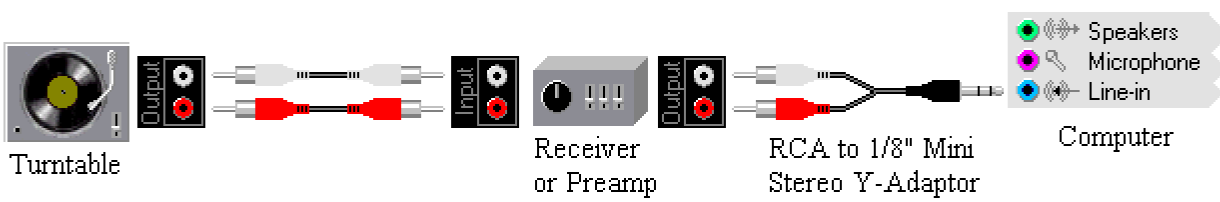


Figure: Connections

Step 1
Connect the turntable output to Phono-in on a receiver or to the input of a preamp, as shown.

Step 2
Connect the receiver or preamp audio output to the light blue Line-In socket on your computer. It is marked with an arrow pointing into the rings, as shown.

Make sure you connect to the correct socket on your computer. If you accidentally plug the receiver/preamp output into the computer's speaker socket you may damage the hardware. Connect your speakers or headphones to your computer first.

Setup Recording

Step 1
Take some time to thoroughly clean the album and carefully inspect and clean the stylus. Also review the information about avoiding Noise. Eliminating noise before it is recorded saves a lot of restoration work later and gives much better quality in the end.

Step 2
Run GoldWave, then press the F11 key or use the Options | Control Properties command, then select the Device tab. Make sure the correct playback and recording devices are selected. Usually the default settings are fine, unless you have more than one sound device. In Windows Vista and 7, you must select a recording device that matches the source to record (Line in this case).

Step 3
Select the Record tab on the Control Properties window, then check the Monitor input on visuals box. This will activate the visuals on the Control window so you can adjust volume levels without having to record.

Step 4
If you are using Windows 2000 or XP or DirectSound mode, select the Volume tab on the Control Properties window. If you are using Windows Vista, 7, or later, adjust the recording volume on the Device tab (see Step 2 above) and go to the next step.

Make sure the correct Volume device is selected. It should be the same as the Recording device on the Device tab.

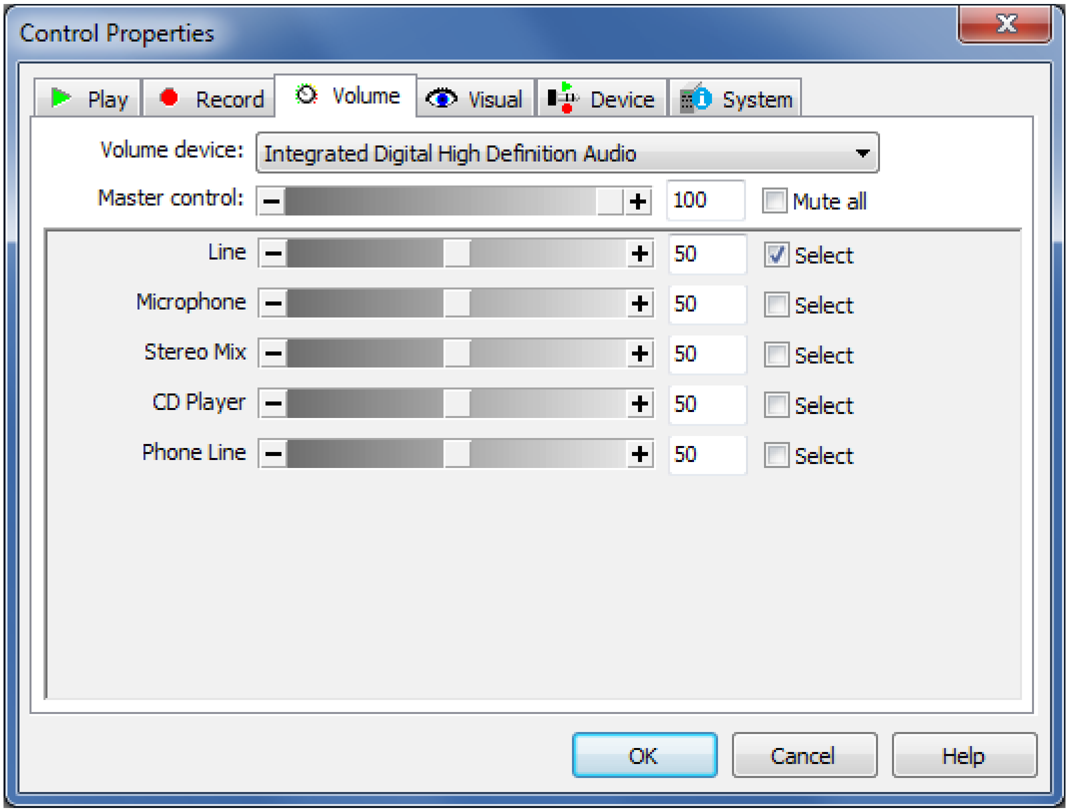


Figure: Volume Properties

Check the Select box for the Line (or Line In) item. Make sure the volume level is at least 50 initially. Make sure no other items are checked as they could be a source of noise. Choose OK to close the Control Properties window so that all new settings are used.

Step 5
Play the album on your turntable to ensure that all the connections are good and that the input level is set correctly. The Level Visual in the Control window should occasionally touch the red region (see "Good" area on the figure below). If you find that the Level Visual goes too high or never goes above the green region, then adjust the volume as explained in step 4. The small rectangular boxes on the far right of the Level Visual should not light during recording, otherwise some clipping distortion may have occurred.

Note that the volume fader on the Control window changes the playback volume only. You must adjust the recording volume under the Volume tab of the Control Properties window.

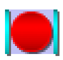
Good
Figure: Control Window

If you get no activity at all, check all connections, make sure your receiver/preamp is turned on, and make sure the correct recording device and volume source are selected as explained in steps 2 to 4. Check to see if your amplifier or preamp has an output level control that needs to be adjusted.

Step 6
Set the duration of recording.

1. Press the F11 key or use the Options | Control Properties command.
2. Select the Record tab.
3. Enter the duration for the recording, such as 25:00 to record the entire side of an album.
4. Select Bounded mode.
5. Choose OK.

If you do not know the duration of the recording and would prefer to stop recording manually, select Unbounded mode instead.

To have more control over the channels and sampling rate used for recording, use File | New to create a new file with the channels, sampling rate, and duration required, then use the  instead in the next step. Note that the recording device may not support the sampling rate or channels you've selected.

Step 7
Click the red  record button or press Ctrl+F9 to start recording in GoldWave. Start playing the thoroughly cleaned, dust free album on the turntable. Press the red  stop button to stop recording when done.

Step 8
Play the recording to see how it sounds. If you are not satisfied with the quality, make volume adjustments, re-clean the album, check connections, etc. Use Edit | Undo to undo the recording and start recording again. If the quality is satisfactory, continue to the next step.

Step 9
Trim any leading and trailing silences by selecting the entire file with the Edit | Select All command, followed by the Edit | AutoTrim command. Depending on how noisy the recording is, a threshold of −30dB or greater may be needed. Start with −40dB. If you still see flat areas on the ends of the waveform, increase it to −30dB.

If the recording is very noisy, manual trimming may be required. To do that, use View | 10 Seconds. Right-click on the waveform to set the start selection marker to a position a second or so before the music begins. Click on the timeline to start playback to check the position. Scroll to the end of the recording by using the scroll bar below the waveform. Right-click to set the finish selection marker at a position a second or so after the music ends (where the waveform goes flat). Use Edit | Trim to remove the silences on either end of the selection.

Step 10
Save the recording by using the File | Save command and providing a name for your recording.

Restoration

Step 1
Use the Effect | Filter | Pop/Click command to remove any pops and clicks from the recording. Use a tolerance setting of 2000 at first. If you find that some pops/clicks are still present, select a short area of the recording where the click occurs then use the Pop/Click filter again with a lower tolerance setting. Using a low tolerance on the entire recording is not recommended since it may distort some sounds, such as trumpet solos. Use Edit | Select All to select the entire recording when done.

Step 2 (optional)
Use the Effect | Filter | Smoother command and select the "Reduce hiss" preset to reduce crackle and hiss.

Step 3
Play the recording to find a few seconds of silence where only background noise can be heard. Select about a second of that noise, then use Edit | Copy to copy it to the clipboard. Select the entire recording by using Edit | Select All. Use Effect | Filter | Noise Reduction to display the Noise Reduction window. Choose the "Use clipboard" envelope option. Preview the settings by choosing the play button to ensure the quality sounds good. If you notice too much tingling or warbling, then lower the Scale setting and Apply the settings while previewing. Choose OK to remove the noise from the recording.

Step 4 (optional)
Use the Effect | Compressor/Expander command and select the "Noise Gate 3" preset, then choose OK. This will eliminate any remaining noise in the silences between songs.

Step 5
Use Effect | Filter | Equalizer to display the Equalizer window. Experiment with the

presets and adjust the bands to boost or reduce bass and treble. Preview the audio until you get the desired results, then choose OK to process the recording.

Step 6
Use Effect | Volume | Maximize Volume and select the "Full dynamic range" preset, then choose OK.

Step 7
Save the restored recording by using the File | Save command.

Split into Tracks

Step 1
Use Tool | Cue Points to display the Cue Points window.

Step 2
Choose the Auto Cue button. On the Auto Cue window, choose the Mark Silence button. Set the threshold to −40dB. Set the silence length to 1 second. You may want to use a longer or shorter time depending on how long the silences are between each song. Set the minimum separation to about 1 minute (1:00). Use a longer or shorter time depending on the length of the shortest song. Choose OK to automatically set cue points at the beginning of each song.

If no cue points appear in the list, choose Auto Cue again and increase the threshold (−30dB). If too many cue points appear, choose the Delete All button and use a lower threshold (−45dB) in Auto Cue. Note that if the album is a live recording with no silences between songs, you'll have to set cue points manually. Close the Cue Points window, then Play the recording and use Edit | Marker | Drop Cue to drop cue points at places where you want to split the recording into separate tracks.

Once cue points have been added, the Split File button becomes enabled.

Step 3
Choose the Split File button on the Cue Points window. Provide a destination folder where each song/track will be saved. Check the "Use CD compatible wave format and alignment" box. Choose OK to create a set of track files for each song.

Step 4
Use CD-R software to write the track files to a blank CD as audio tracks. You can import the files into iTunes or Windows Media Player to use those programs to burn a CD. Note

that you cannot just copy the files to the CD using Windows Explorer. That will create a data track. Each song must be written as a separate audio track.

End of Tutorial

Appendix E: Troubleshooting and Q&A

Troubleshooting

Problem	Cause/Solution
Cannot open large files	Make sure that hard disk storage is enabled in Options | Storage. Make sure that you have plenty of free RAM and hard drive space. CD quality sound requires 10MB per minute and 20MB per minute when editing.
Cannot play sounds	Sound driver is incorrectly installed or needs to be updated. Check that the Windows Sound Recorder accessory can play sounds. If it doesn't, the driver is not installed correctly. Make sure that your audio device is selected by using the Device Properties tab under Control Properties.
Cannot record sounds	See above. Make sure you have the correct recording source selected. See Recording Sounds under Control Overview. Make sure your audio hardware is capable of recording. Sound may be in use by the playback device; click on the stop button. Recording device may be in use by another program; close the other program.
System seems slow	Your system may not be fast enough to draw

or visuals do not update smoothly	the visuals. Try reducing the frame rate in Visual Properties tab under Control Properties or set all visuals to blank.
System freezes or crashes or an exception occurs	Make sure that you have a Pentium III or better system. If the crash occurs during recording or playback, install an updated sound, video, and motherboard drivers. If the crash occurs when you try to run GoldWave, use the **Start | Programs | GoldWave | GoldWave Setup** to disable some of the plug-ins or OpenGL. You may have encountered a problem. If you can duplicate the problem after restarting your system, contact GoldWave Inc. for more assistance.
After a crash, there is less free space on the hard drive	Delete files in the temporary storage folder specified under Options | Storage. These files usually have names like GWXXX.TMP. Note that you may be able to recover the files by opening them in GoldWave and specifying the Raw type with "32 bit IEEE float, stereo (or mono), little endian" attributes.
Visuals are out of synchronization	Many audio drivers return inaccurate positions. Make sure you have the most recent sound driver.
Pops, clicks, or stuttering during playback	Your sound card driver is rubbish. Get a new sound driver, or a new sound card, or tell your sound card manufacturer about the problem. Pops and clicks can occur at the beginning or ending of a sound if the first or last sample is not silence. Fading in/out a small selection can sometimes fix this. Enable the Edit | Marker | Snap to zero-crossing feature to avoid pops and clicks when editing.

Expression Evaluator slow	Make sure that RAM storage is selected in **Options | File**. Remember to close and reopen your sounds for this setting to apply. Your crusty old 386 system does not have a co-processor. Go to a computer junk yard and find a Pentium system.
Editing seems to be getting slower and disk activity is increasing	Files on your hard disk are becoming fragmented. Use the Windows Disk Defragmenter system tool.
Distortion in recording	You need to lower the recording volume. See Volume Properties tab under Control Properties for information about selecting and controlling recording volume. Check that all connections are correct and firm (do not connect a "line-out" to "mic-in", for example).
Gaps in recording and playback	The Windows virtual memory manager can sometimes cause gaps due to excessive swapping. Installing more RAM may help. Your system may be too slow to display the visuals. Try reducing the frame rate in the Visual Properties tab under Control Properties or set all visuals to blank.
Sound won't play for more than a few seconds.	Make sure you've selected the entire file.

Common Questions and Answers

Why do I get only silence when I try to record?

You have to select the correct recording source. See Volume Properties tab under Control Properties.

How do I record an audio CD?

Use the Tool | CD Reader command to copy the audio directly from the CD.

How can I see and adjust recording volume levels without recording?

Use the Monitor input on visuals option under the Record Properties tab.

How can I avoid stuttering and glitches in recording?

Do not use disk compression. Set visuals to Blank. Close all other programs. Restart your system before (or between) recording for long periods of time. Update your sound driver.

How do I select part of a sound?

See Editing Overview for more information.

How can I tell if the finish marker is in the right place without playing the entire selection?

Right click on the waveform just before the finish marker and choose the "Play From Here" menu item. Or, go to the Play Properties tab and select the **Finish** option for one of the play buttons. That will play a few seconds before the finish marker.

How do I edit individual amplitude samples with the mouse?

See Redrawing the Waveform with the Mouse.

Why does GoldWave show the evaluation messages?

GoldWave is not free software. If you are a licensed user and are still seeing evaluation messages, you'll need to enter your license into the program again. Directions for entering the license are included with the license.

Can I convert sound files to MIDI?

No. MIDI files do not contain digital audio. They contain notes and timing information for instruments. In other words, they contain instructions for playing the music, but not the music itself.

How are you?

Fine, thanks.

How can I split a large recording into individual songs?

Use the Cue Points tool with the Auto Cue Button and the Split File Button.