

GoldWave™ User's Manual

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1 Introduction

GoldWave™ is a Microsoft Windows™ 3.1 application that uses the standard audio interface to create a full digital editing and effects studio. GoldWave includes several features not found in current audio applications, such as an expression evaluator, an oscilloscope, a device controls window, and independent left/right channel editing of stereo sound files.

The expression evaluator allows you to generate everything from simple sine waves to band-pass filters. It supports over 20 common functions and operations.

The oscilloscope displays sounds as they are being played or recorded. You can see what the sound looks like.

The device controls window contains buttons for playback, recording, volume, etc. The window is separate from GoldWave's main window and allows more flexible control of audio devices.

Along with the standard editing functions of cut, copy and paste, GoldWave has several audio effects built in. These include reverse, echo, fade out, volume, and silence. GoldWave employs a multiple document interface (MDI) so that up to 5 sounds can be loaded at one time, making file-to-file editing easy. The sounds are displayed graphically and you can zoom in to see higher detail. In fact, you can zoom in far enough so that a single sample is graphed.

This manual presents and explains many features of GoldWave. The first part discusses system requirements, installation, etc. The second part covers the menu structure. Topics are covered in the order that they appear in the main menu. The final section provides general information (shareware concept, warranty, registering, etc.). Please be sure to read the shareware/warranty section! For users who are unfamiliar with digital audio, Appendix A briefly introduces some of the fundamentals of digital sound. It also provides some solutions to common recording problems. This manual assumes that users are familiar with Microsoft Windows™.

2 Getting Started

2.1 System Requirements

The minimum system requirements for GoldWave are:

- 16MHz 286 IBM PC or compatible
- Microsoft Windows™ 3.1
- 4 Megabytes of RAM
- 1 Megabyte free on your hard disk
- A mouse
- An EGA/VGA card and monitor
- An MPC audio card and software driver (or PC speaker driver)

A math processor is not required, but it is recommended if you will be using the expression evaluator often.

2.2 Notation Used in this Manual

Bold text and a vertical bar '|' are used to indicate menu commands. For example, **File | New** means to select the New command from the application's File menu.

2.3 Installation

GoldWave and all its files should be kept in a single directory of its own (the file list is given in the General Information section). The best way to do this is to unzip GOLDWAVE.ZIP into a new directory.

For example, in DOS:

- 1) C:>mkdir gw
- 2) C:>copy goldwave.zip gw
- 3) C:>cd gw
- 4) C:>pkunzip goldwave.zip

Before running GoldWave make sure that you have installed an appropriate Windows sound driver using the Control Panel "Drivers" Applet. The driver and installation instructions should be included with the hardware. If you do not have a sound card, then you can use the PC-Speaker driver (you may have to search BBS or ftp sites to find this, since it is not included with Windows). Note: Since the PC-Speaker driver does not return control to Windows until after the sound has finish playing, the oscilloscope and stop & pause buttons will not work.

To add a GoldWave icon to a Program Group use **File | New** on the Program

Manager's menu, select Program Item then click OK. Type "GoldWave v1.00" as the description. Select Browse, find and double click on "goldwave.exe", then click OK.

To run GoldWave, either double click on the GoldWave icon or use **File | Run** on the Program Manager's menu.

GoldWave allows you to choose which devices to use for input and output. Selecting **Options | Output device** presents you with a list of installed output devices and shows their capabilities. Choose the device you wish to use. A similar list is presented for **Options | Input device**.

Be sure to check the readme.txt file for any additional information.

3 Using GoldWave

3.1 Window Overview

GoldWave is composed of three types of windows: the main window, the Scope and Controls window, and sound windows. The main window contains the menu and a row of buttons. It groups together and manages all the sound windows when they are opened. At the bottom of the main window is an information bar that gives you attributes of the active sound window.

The Scope and Controls window provides an easy to use interface to your audio hardware. It contains buttons to play, record, and stop sound as well as controls for volume, balance, and playback speed (provided your audio device supports these features). An oscilloscope is displayed here whenever a sound is played or recorded. See Scope and Controls Overview in a following section.

Sound windows are created when you open a file. Up to 5 sounds can be opened at any one time. These windows contain a graph of the sound with a time axis near the bottom. For stereo sounds, two separate graphs are shown with the top (green) graph as the left channel and the bottom (red) graph as the right channel. At the bottom of the sound window is a small "Overview" box. This box shows you what part of the sound is selected (highlighted green and/or red), what part is displayed in the above graph (black background), what part is not visible in the above graph (dark grey background). Initially, the entire sound is visible and selected.

GoldWave makes extensive use of "bitmapped buttons". These buttons have pictures which help describe their purpose. Buttons containing no colour (only shades of grey) are disabled. They are disabled when the function they perform is invalid. Most buttons are disabled when you initially run GoldWave. After opening a sound file, most of these buttons will become enabled.

Note Version 1.00: the rewind and fast forward button serve no purpose and

are never enabled. The commands Borg, Helium, Time comp/exp, Print, Pitch, were not ready for this version (maybe the next version will have them, if I have time).

3.2 Editing Overview

The GoldWave editor is block based. A block is defined as the highlighted part of the sound graph between two vertical markers. All editing and effects are performed on this highlighted block. As mentioned in the previous section, when a sound file is first opened, the block is the entire sound. The vertical markers are cyan lines located at the far left side (start marker) and far right side (finish marker) of the graph. They may be difficult to see.

There are two ways to move the markers, depending on **Options | Settings** setup. GoldWave does not use Windows' standard "click-and-drag" method because it does not allow accurate positioning of the markers. The methods below allow you to independently set the start and finish markers to an individual sample (when zoomed in at a 1:1 level or better).

Method 1:

The default method (check "Quick block select" in **Options | Settings**), uses the left and right mouse buttons to set the start and finish markers respectively. Whenever the mouse arrow is inside the sound graph, the arrow changes into a vertical bar with small arrows on either side. This means that you can change a markers position.

To move the start marker:

Click the left mouse button on the graph at the point where you want to place the start marker.

To move the finish marker:

Click the right mouse button on the graph at the point where you want to place the finish marker.

Note that you cannot place the finisher marker before the start marker. The same is true for setting the start marker after the finish marker.

The drawback of this method is that it is easy to accidentally change the start/finish marker when clicking on a sound window.

Method 2:

If the "Quick block select" check box in **Options | Settings** is not checked, then selection method 2 is used.

To move a marker, position the mouse arrow over it. You should see the arrow change into a vertical bar with small arrows on either side. You can then click and drag the marker to a new location. For example, to move the start marker from its original position:

- 1) Move the mouse arrow to the far left side of the graph. Make sure it is inside the graph area and the arrow shape has changed.
- 2) Press and hold the left mouse button.
- 3) Move the mouse to the right.
- 4) Release the mouse button when the start marker is at the desired position.

Note Version 1.00: You must have a mouse to move the start/finish markers since there are no keystrokes which will move them.

To find a certain part of the sound, you can play the sound and mark it as it plays. GoldWave indicates the part being played with a white vertical line.

In most cases, editing and effects are performed only on the active sound window, which is the one that is on top.

Window's clipboard (a temporary storage area that all applications can use) is used for most of the editing commands. Note that cutting or copying a large block into the clipboard can consume a lot of memory.

See Editing under the Options section for information about controlling independent channel editing.

3.3 Scope and Controls Overview

The Scope and Controls window is an easy to use interface to your audio hardware and/or drivers. In the upper half of this window is the oscilloscope box in which sound is graphed "real-time" during playback and recording. In the middle of the window is a standard set of tape recorder controls. From left to right the buttons are: rewind, pause, fast forward, play, stop, and record (in version 1.00 the rewind and fast forward buttons serve no purpose). In the lower half of the window are controls to set the devices output volume, balance, and playback speed (not many devices support speed control).

3.3.1 Playing sounds

After opening a sound, you can use the play button (the one with a green triangle that points to the right). While a sound is playing, it is shown graphically

in the oscilloscope box. The current position is shown in the graph of the sound window with a vertical white line. Sounds can be played only if the driver supports the sound's format and attributes. By using **Edit | Copy** and **Edit | Paste**, it is possible to convert a sound to a playable format.

3.3.2 Pausing playback

While a sound is playing, you can pause it with the Pause button (two vertical blue bars). Remember to use either play or stop later. Pause will freeze the oscilloscope so you can see the shape of the sound.

3.3.3 Stopping sounds

Recording and playback can be stopped immediately with the stop button (red square).

3.3.4 Recording sounds

The record button (red circle with letter 'R' in it) allows you to record your own sounds. Recording occurs inside the block. The block's data will be overwritten. To make room for recording without overwriting the block, you can use **Edit | Insert space**. Recording will automatically stop when the block is full. You can stop recording at any time with the stop button.

Recording is successful only if the driver supports the active sound's attributes/format. You can try recording into a new sound (**File | New**) and use **Edit | Copy** and **Edit | Paste** to overcome this limitation.

3.4 File Menu Commands

3.4.1 New

Creates a new sound file 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. Check **Options | Output device** for attributes compatible with your hardware.

3.4.2 Open (Open Button) and Supported File Formats

Presents you with a list of files. By default, all files with the extension ".wav" are listed. You can change the extension to search for different sound formats. After you select a file, it is opened and given its own sound window. Its attributes are displayed in the information bar.

Sound files are similar to document files in that different word processors

save files in different formats. For example, Window's Write uses a special format with a ".wri" filename extension, but Notepad uses a plain format with a ".txt" extension.

GoldWave supports all the sound formats listed in the table.

Extension	Comments
.au	Sun or NeXT format. Supports 8&16-bit linear and uLAW encoded files. Encoded files are converted to 8 bits or 16 bits, depending on Options Settings selections (8 bits by default). The conversion process can take a minute or so for long sound files. The header block is preserved.
.iff	Amiga 8SVX format. The blocks NAME, COPY, ANNO, AUTH, and CHAN are all preserved.
.snd	Raw or NeXT format. NeXT format is automatically detected. Raw format presents a dialog to specify the format.
.wav	RIFF WAVE 8-bit and 16-bit PCM mono or stereo. Only files with one data chunk are supported. Only the INFO chunk is preserved.
.voc	Sound Blaster files. Only 8-bit unpacked mono files are supported.

GoldWave will present an attribute dialog if the file is corrupted, unrecognized, or contains data without header information (raw files). This dialog allows you to select one of three general sound formats and give specific attributes. If you don't know the file's format or attributes, experiment with trial-and-error. Generally, sounds will be noisy if the format or number of bits is incorrect, in which case you will have to close and re-open the sound using a different format.

If the sampling rate is wrong, the playback speed will be too fast or too slow. You can use **Effects** | **Playback rate** to quickly re-adjust this. Appendix A has more information about sound attributes.

3.4.3 Save

Saves the active sound file to the disk. The sound is saved with the same name and format that it was opened. 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 and close other applications. Make sure that the file is saved successfully before closing GoldWave, otherwise data may be lost.

3.4.4 Save as

Allows you to save a sound file under a different name. You cannot save a file in a different format. You should provide an appropriate extension, since none will be appended.

3.4.5 Exit

Closes all sound windows and quits GoldWave. All sound playback / recording is stopped. You will be asked to save any changes.

3.5 Edit Menu Commands

This section assumes that you are familiar with the terms introduced in the "Editing Overview" section.

See Also: Editing in the Options section.

3.5.1 Cut (Cut Button)

Removes the block from the active sound and puts it in the clipboard. The contents of the clipboard can then be pasted into a sound window using **Paste**.

To cut:

- 1) Move the start and finish markers to the part of the sound you want to cut.
- 2) Choose **Cut** from the Edit menu or click on the Cut button.

3.5.2 Copy (Copy Button)

Copies the block into the clipboard. The block is not removed. The contents of the clipboard can then be pasted into a sound window using **Paste**.

To copy:

- 1) Move the start and finish markers to the part of the sound you want to copy.
- 2) Choose **Copy** from the Edit menu or click on the Copy button.

3.5.3 Paste (Paste Button)

Places the sound in the clipboard into the active sound. The sound is inserted at the start marker's position. The length of the active sound is increased so that the clipboard sound will fit. The clipboard sound is automatically converted to match the attributes of the active sound.

To paste:

- 1) Move the start marker to the place where you want to paste the clipboard sound.
- 2) Choose **Paste** from the Edit menu or click on the Paste button.

Tips: By copying a small block and pasting it several times, a stutter effect can be achieved.

You can use Paste to convert stereo to mono by copying from a stereo file and pasting to a mono file.

If a sound format is not supported by your audio device, use **File | New** to make a new compatible sound. **Copy** from the original, then **Paste** to the new.

Technical Note: Clipboard data is automatically up-sampled or down-sampled to keep pitch and playback speed constant. You can use **Effects | Playback rate** to change the playback speed before copying/cutting.

3.5.4 Mix (Mix Button)

This is a special editing feature for sounds. The sound in the clipboard is blended (superimposed) with the active sound. You are asked for the mix volume. This volume is applied to the clipboard sound as it is being mixed (see Volume in the Effects section).

To mix:

- 1) Move the start marker to the place where you want to mix the clipboard sound.
- 2) Choose **Mix** from the Edit menu or click on the Mix button.
- 3) Enter the volume to mix the clipboard (100 = normal volume).
- 4) Choose OK.

Tip: With this feature you could add your own vocals to music.

Technical Note: Clipboard data is automatically up-sampled or down-sampled to keep pitch and playback speed constant. You can use **Effects | Playback rate** to change the playback speed before copying/cutting.

3.5.5 Delete (Delete Button)

Delete removes the block from the active sound. The block is not copied to the clipboard. Note that if only the left or right channel is active in a stereo sound, then only that channel is removed. Since it is not possible for one channel to be larger than the other, the end of the deleted channel is padded with silence (this is also true for **Cut**).

To delete:

- 1) Move the start and finish markers to the part of the sound you want to delete.
- 2) Choose **Delete** from the Edit menu or click on the Delete button.

3.5.6 Insert space (Space Button)

Inserts some blank space in the active sound at the start marker's position. You are asked how long (in seconds) the space should be. This command can be used to increase recording time.

3.5.7 Select view (Sel Vw Button)

Moves the start and finish markers to the far left and far right of the graph. It selects the part of the sound shown in the graph.

Tip: After editing or adding an effect, a quick way to play the entire sound is to click on the Out button, then the Sel Vw button, then the Play button.

3.6 View Menu Commands

This section assumes that you are familiar with the terms introduced in the "Editing Overview" section.

The following commands allow you to get more (or less) detail about the actual shape of the sound. They are similar to Paintbrush's zoom commands. The Overview box near the bottom of each sound window gives you useful information about what section of the sound is currently in the graph.

When you have zoomed into (magnified) a part of the sound, a scroll bar will appear at the bottom of the sound window so you can move to different parts of the sound while still keeping the same level of magnification.

If you opened a long sound, the graph will not be accurate. Generally, the longer the length of time that is graphed, the less accurate the graph will be.

3.6.1 Zoom block (Block Button)

Increases detail by graphing only the part of the sound in the block. It essentially magnifies the block. By selecting a new block, you can zoom in even further.

To zoom in on the block:

- 1) Move the start and finish markers to the part of the sound you want to magnify.
- 2) Choose **Zoom block** from the View menu or click on the Block button.

3.6.2 Zoom 1:1 (1:1 Button)

Gives you an accurate, full-detail graph of the sound at the start marker's position. Technically speaking, one sample is mapped to one pixel on the screen. This level of magnification is good for finding defects (pops and clicks) in sounds.

3.6.3 Unzoom (Out Button)

Shows the entire sound in the graph. When sound files are opened, the graph is unzoomed.

3.7 Effects Menu Commands

This section assumes that you are familiar with the terms introduced in the "Editing Overview" section.

With these commands, you can change a sound or create new sounds. The commands are comparable to Font Menu Commands in word processors. For example, using font commands, you can change the size of text. In GoldWave, using the "Volume" effect changes the "size" (amplitude/volume) of a sound wave. Note: Some of the effects may require a few minutes to complete.

See Also: Editing in the Options section

Technical Note: Even though the word 'volume' is used throughout this section, the amplitude is changed linearly (logs are too slow).

3.7.1 Echo

Produces an echo effect with the block. The echo delay and volume (see Volume) can be entered after choosing this command. You may want to try the default values first.

The longer the delay, the longer it takes for the echo to "bounce" back. Try values less than 0.1 for a large room, 0.3 for a baseball stadium, above 0.3 for a

canyon echo.

The lower the volume, the quieter the echo will be. Values less than 50 give good results.

To make the echo sound correct, GoldWave extends the effect slightly outside the block boundary. This may increase the length of the sound. If you do not want to modify sound outside the block or you do not want the length of the sound to change, use the **Options | Editing** dialog to disable modification outside the block.

To add an echo:

- 1) Move the start and finish markers to the part of the sound you want to add an echo.
- 2) Choose **Echo** from the Effects menu.
- 3) Enter the delay time, then choose OK.
- 4) Enter the volume, then choose OK.

3.7.2 Fade out

Gradually decreases the volume throughout the block. The percentage of fade can be entered after choosing this command. The fade percentage is the amount that the volume should decrease. A value of 100 fades to complete silence. A value of 50 fades to half the original volume.

To fade out part of a sound:

- 1) Move the start and finish markers to the part of the sound you want to fade.
- 2) Choose **Fade out** from the Effects menu.
- 3) Enter the fade percentage, then choose OK.

Tip: To fade in a sound, **Reverse** the block first, **Fade out**, and **Reverse** it again.

3.7.3 Invert

Reflects the sound about the x (time) axis. This produces no noticeable effect in mono sounds and has a slight effect in stereo sounds. Anyone have a use for this effect?

3.7.4 Maximize volume

Searches the block for the maximum percent that the volume can be increased without producing distortion or clipping errors. This volume is then applied to the sound. If the percentage is above 2000% or equal to 100%, no changes will be made. See Volume for more information.

3.7.5 Playback rate

Changes the playback rate of the entire sound. The sound will play faster (or slower) and its pitch will be higher (or lower). Essentially, this just changes the first number in the information bar. Values of 11025, 22050, and 44100 are the recommended values.

Tip: If you want to convert the actual sampled data from one sampling rate to another, you could use **Edit | Copy** and **Edit | Paste**. If the file you copy from is sampled at 22050Hz and the file you paste to is sampled at 11025, then the paste command will down-sample the data as it is being pasted. This cuts the size of the sound file in half. Up-sampling is also possible this way.

3.7.6 Reverse

Reverses the block so that it plays backward. Now you have an easy way to listen to all those "satanic verses".

3.7.7 Silence

Erases the block. The sound in the block is muted (set to zero volume).

3.7.8 Volume

Modifies the block so that it sounds louder or softer. The percentage of volume can be entered after selecting this command. Values less than 100 make the sound softer and values greater than 100 make the sound louder. A value of 100 has no effect. Enter 50 to half the volume and 200 to double the volume.

By using independent left/right editing (**Options | Editing**), the internal balance of a sound can be adjusted.

The Scope and Controls window can also affect volume. It only changes the device's output volume, whereas the Volume effect actually changes the volume within a sound.

3.7.9 Expression evaluator

Allows you to create new effects or generate signals. See appendix B for more details.

3.8 Options Menu Commands

3.8.1 Input device

Presents a list of input devices that were installed with Control Panel's Drivers applet. If no input devices are installed, a warning message is displayed and recording is disabled.

The list shows you the name of the device/driver. Above the list is a table that tells you what sound attributes/formats the driver supports. All boxes with check marks in them are supported by the currently selected driver. You can select any driver to see its capabilities.

Tip: Since you can choose the input device, you can have several devices/drivers installed at once. Note: The drivers must not interfere with each other.

3.8.2 Output device

Presents a list of output devices that were installed with Control Panel's Drivers applet. If no output devices are installed, a warning message is displayed and playback is disabled.

The list shows you the name of the device/driver. Above the list is a table that tells you what sound attributes/formats the driver supports. All boxes with check marks in them are supported by the currently selected driver. You can select any driver to see its capabilities.

Tip: Since you can choose the output device, you can have several devices/drivers installed at once. For example, it is possible to have the PC-Speaker driver and a Sound Blaster driver installed at the same time. Note: The drivers must not interfere with each other.

3.8.3 Aux volume

Sets the output volume on an auxiliary device. This may be useful when recording from a CD-ROM. A list of installed auxiliary devices is displayed or a warning message is given if no auxiliary devices are present.

To change auxiliary output volume:

- 1) Select a device. If the device supports volume control, the "left" control will be enabled. If the device supports stereo volume control, both the "left" and "right" controls will be enabled.
- 2) Adjust the volume with the "left" and/or "right" controls.

Note: This feature is untested (I don't have the hardware to test it).

3.8.4 Settings

Changes several features in GoldWave.

Sound Path is used by **File | Open**. You can specify the directory where you keep your sound files. If you type "d:\sound\wave", for example, the **File | Open** dialog will automatically list the files in that directory whenever you start GoldWave.

If **Save window positions** is checked, the main window and Scope and Controls window positions will be saved when you exit GoldWave. This means you will not have to adjust window positions every time you start GoldWave.

If **Load uLaw as 8-bits** is checked, any encoded file (".au" files) will be converted to 8-bit PCM format. This will cause some loss of quality, but will allow playback on a larger variety of audio devices. If the box is not checked, the data is converted to 16-bit PCM format. If your audio device supports 16-bit PCM files, uncheck this box.

The **Scope active** check box enables or disables the oscilloscope. Unchecked, the oscilloscope is disabled during playback and recording (the oscilloscope box will still be present).

The **Quick block select** check box allows you to choose which method to use when selecting a block. See Editing Overview for more information.

3.8.5 Editing

Sets the behaviour of editing and effects. These options apply only to the currently active sound. This means you can have different "behaviours" for different sound windows.

Channel Editing lets you control which channel (or channels) will be edited for stereo sounds. For example, if you choose "Left channel", all effects and editing will only change or use the left channel. You can then use **Edit | Copy** to extract the left channel into the clipboard or use **Edit | Mix** to mix the clipboard with the left channel and so on.

The **Modify outside block** check box allows you to protect sounds outside the block from being changed (**Effects | Echo** uses this).

3.8.6 Register

Allows you to register your copy of GoldWave in your own name. To get a password, please see Registering in Section 4.

3.9 Window Menu Commands

The Window Menu contains a list of all sound windows currently opened. These commands organize sound windows.

3.9.1 Tile

Arranges sound windows side-by-side so that they are all fully visible.

3.9.2 Cascade

Layers sound windows on top of each other so that their title bars are visible.

3.9.3 Arrange icons

Arranges minimized sound window icons in rows.

3.10 Help Menu Commands

3.10.1 Contents

Starts Window's on-line help utility and gives a list of contents for GoldWave help.

3.10.2 About

Displays version/registration information. Shows how much memory is available under the GoldWave icon.

4 General Information

4.1 ShareWare Concept and Warranty

GoldWave is designated as ShareWare. This means that you may use GoldWave free for a 30 day trial period. At the end of this period, if you decide to keep GoldWave, you are required to register (see Registering below). GoldWave has not been crippled in any way. Therefore, your honesty and support are essential for future development of this application. You are encouraged to give copies to all your friends and upload it to BBS and ftp sites (see below for restrictions).

GoldWave brings affordable multimedia to the multitudes.

GOLDWAVE.ZIP ("the package") includes the following software and documentation:

GOLDWAVE.EXE
GOLDWAVE.HLP
GOLDWAVE.WRI
README.TXT
REGISTER.TXT
BWCC.DLL

The package is copyright 1993 by Chris S. Craig.

You may copy and distribute the package through BBS and ftp sites. Only the GOLDWAVE.ZIP file may be distributed or copied.

You are prohibited from:

- charging a fee or requesting donations for the package;
- distributing or including the package in commercial products;
- modifying the package .

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

All trademarks/registered name acknowledged.

4.2 Contacting the Author

I would appreciate any registrations, comments, suggestions, and bug reports. During May '93 to August '93 and January '94 to April '94, I will be completing the last two academic Terms (7 and 8) of my university degree. I will do my best to provide user support, but I may not have time to answer all questions.

Important:

Please do not ask programming or development questions such as "Can I have the source code?", "What does your sound class structure look like?", "How can I make an oscilloscope?", etc. These type of questions can take hours to respond to. Therefore, I will only answer questions relating to the use of GoldWave.

You can send e-mail to:

`chris3@garfield.cs.mun.ca`

I prefer e-mail. Be sure to give a valid return address.

Regular mail/registrations can be sent to:

Chris Craig
P.O. Box 51
St. John's, NF

CANADA A1C 5H5

Thank-you for using GoldWave.

4.3 Registering

If you think GoldWave is a useful utility, a registration of \$30 Canadian (only \$25 US) would be appreciated. You will receive a password that will register your copy of GoldWave in your name and remove any registration messages.

If you send more than \$59 Canadian, you will receive the next version of GoldWave when it is available.

To register, fill in the REGISTER.TXT file, then print it out and mail it along with your check or money order to the address in the previous section.

4.4 Reporting Problems and Bugs

This is my first Windows program as well as my first major C++ project. This is also the first release (GoldWave v1.00), so there may be some problems. See Contacting the Author for more information.

Known problems:

- Some audio drivers report the current play/record position incorrectly. This means that the oscilloscope will be inaccurate.
- If you try to delete a single sample, 2 samples will actually be deleted.
- No cross-fading is performed on editing boundaries.
- No anti-aliasing filter is applied when paste or mix down-samples.
- Oscilloscope graph is inverted (its quicker that way).

4.5 Trouble Shooting

Problem:

Cannot play sounds GoldWave or audio device/driver is incorrectly installed.

Solution:

Make sure a WAVE driver is installed in the Control Panel's Drivers applet.

Check that Window's Sound Recorder can play sounds. If it doesn't, the

driver is not installed correctly. Make sure that your audio device is capable of playing the sound. For example, an 8-bit audio device cannot play a 16-bit sound.

Problem:

Cannot record sounds

Solution:

See above.

Make sure your audio device/driver can record sounds.

The sound may be paused. Click on the stop button.

Problem:

Cannot use the stop or pause button

Oscilloscope does not work

Solution:

The audio driver is synchronous. This means that Windows (and GoldWave) loses control until the sound has finished playing.

Oscilloscope is not active (**Options | Settings**).

Problem:

Oscilloscope is out of sync

Solution:

Many audio drivers return inaccurate 'current' positions. Make sure you have the most up-to-date device driver.

Problem:

Periodic popping or clicking while playing a pure sine wave

Solution:

Most audio devices/drivers make periodic pop/clicks between DMA transfers and/or memory boundaries. It is most noticeable when playing pure sine waves.

Pops and clicks can occur at the beginning or ending of a sound if the first or last sample is not 0 (silence). Fading in/out a small block can sometimes fix this.

4.6 Revision History

v1.00:

First release. Includes: mono/stereo, 8/16 bit, uLAW, independent channel

editing, up to 5 sound windows, detached Scope and Controls window, tool bar, information (attributes) bar, MDI, expression evaluator, and several effects.

4.7 Future Versions and ScopeTrax

Here's a list of possible future enhancements to GoldWave:

- Full keyboard support for moving start/finish markers
- Sampling rate conversion command, instead of having to use paste
- More DSP/FFT effects (Pitch shift, time compress/expand, filters, etc.)
- Add Print command for sound graphs
- OLE support
- Direct editing with the mouse
- Support for ADPCM, unsigned 16-bit, etc. file formats
- More device control (loop, play block, window, all)
- "Save as" saves files with any format
- Better on-line help
- etc...

ScopeTrax Note:

ScopeTrax (released March '92) is a sound player for DOS that has a true real-time oscilloscope and the ability to play clearly through most PC-Speakers. I had planned to make new versions of ScopeTrax, but because of several limitations with DOS (640K barrier, lack of standard sound drivers, etc.), GoldWave may have to replace ScopeTrax. If a larger number of people disagree, I will try to make a new version of ScopeTrax. I still have not received enough donations to afford a new Sound Blaster Pro or SB16 ASP developer kit.

4.8 Acknowledgements

I thank the following people for their suggestions and support (ScopeTrax):

Rich Mintz
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Randal P. Ford

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Appendix A: Audio Attributes/Terminology

A.1 Digital Sound Attributes

A digital sound (sampled sound) is composed of thousands of pieces of data, called samples. Each sample holds the loudness (amplitude) of a sound at a given instant in time. This is similar to computer graphics where each point of light (pixel) has a certain brightness and location. All these points combine to make a picture. In digital sound, all the samples combine to make a sound. There are several attributes that determine the quality and quantity of digital sound. They are: **sampling rate**, number of **bits**, and number of **channels**.

The **Sampling rate** is the number of times, per second, that a sample is recorded. It is measured in Hertz (seconds⁻¹, Hz). A high sampling rate will yield a high quality of digital sound in the same manner that high graphics resolution will show better picture quality. Compact disks, for example, use a sampling rate of 44100Hz, whereas some telephone systems use a rate of only 8000Hz.

The rate to use depends upon the type of sound and the amount of memory and disk space you have available. Higher rates consume larger quantities of memory. In the above example, the compact disk may require 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 a table at the end of this section.

The number of **bits** determines how accurately the amplitude of a sample is recorded. The two most common are 8-bit and 16-bit formats. In an 8-bit sample, there are 256 different levels of amplitude possible. 16-bit samples have 65536 levels. To compare the difference, let's say that you are a teacher grading tests and you can use one of two marking schemes. In scheme #1, the mark is out of 10. In scheme #2, the mark is out of 1000. All marks must be rounded off (no fractions

allowed). If a student gets two thirds of the questions right, then in scheme #1, the grade will be 7 out of 10. In scheme #2, the grade will be 667 out of 1000. Obviously, scheme #2 is more accurate. In digital sound, low levels of accuracy can cause quantization errors, as discussed in the next section.

One drawback of using 16 bits is that it uses twice as much storage space. This is shown in a table at the end of this section.

The number of **channels** also affects the quality and quantity of digital sound. A single channel sound, referred to as a mono sound, contains information for only one speaker (i.e. similar to AM radio). A two channel sound (stereo sound) contains data for two speakers (i.e. similar to FM stereo). Stereo sounds require twice as much storage as mono sounds.

Format/Attributes

Quality/Sound type

Quantity, Storage/second

11025Hz, 8-bit, mono

Fair quality. Good for speech and low pitch sounds.

11025 bytes

11025Hz, 8-bit, stereo

Fair quality stereo.

22050 bytes

11025Hz, 16-bit, mono

Less noise.

22050 bytes

11025Hz, 16-bit, stereo

Stereo, less noise.

44100 bytes

22050Hz, 8-bit, mono

Good quality. Good for music and relatively complex sounds.

22050 bytes

22050Hz, 8-bit, stereo

Good quality stereo.

44100 bytes

22050Hz, 16-bit, mono
Very good quality.
44100 bytes

22050Hz, 8-bit, stereo
Very good quality stereo.
88200 bytes

44100Hz, 8-bit, mono
High quality. Good for all sounds.
44100 bytes

44100Hz, 8-bit, stereo
High quality stereo.
88200 bytes

44100Hz, 16-bit, mono
Excellent quality.
88200 bytes

44100Hz, 16-bit, stereo
Excellent quality stereo. Large storage requirements.
176400 bytes

Amplitude vs. Volume Note

Amplitude is the height of a wave. Volume is amplitude measured in dB, which is a logarithmic scale.

A.2 Problems with Recording

There are five potential problems when recording sound: aliasing, clipping, quantization, internal noise, and DMA/memory boundaries.

Aliasing occurs when the sampling process does not get enough data to correctly determine the shape of the sound wave. The digital sound will have high or low pitch tones that never existed in the original sound (See Appendix B, end of Signal Generation section). This problem can be eliminated by using higher sampling rates (or by using anti-aliasing filters).

Clipping errors occur when the sampled amplitude is outside the range of valid values. For example, if the range is -1.0 to 1.0, and a value of 1.2 is sampled, then this value must be clipped to 1.0. A way to reduce these errors is to lower the input volume when recording.

Quantization errors occur when the sample is rounded to the nearest level of amplitude. This can be explained by using the "marking schemes" example of the previous section. The number two thirds (2/3) is represented by 7/10 in scheme #1. This gives a quantization error of:

$$| 7/10 - 2/3 | = 1/30$$

Similarly, in scheme #2, the quantization error is:

$$| 667/1000 - 2/3 | = 1/3000$$

Clearly, scheme #1 has the smallest error. Therefore, using 16 bits instead of 8 bits is a good way to reduce quantization errors.

The other two recording problems deal with computer hardware. To minimize internal noise, make sure your audio card is installed as far away from your graphics/monitor adaptor card as possible. If you use a microphone, keep it away from your monitor and computer fan. Remember to use shielded cables.

DMA and memory boundaries can cause loss of quality during recording and playback (pure sine waves rarely play without some "defects"). When recording at high sampling rates (in DOS), use a RAM disk. Unfortunately, Windows offers little control over when disk accesses will occur. Some of the latest audio cards and drivers may reduce this problem.

Appendix B: Expression Evaluator

B.1 Overview

The Expression Evaluator is a versatile tool for manipulating and generating audio data. After you select **Expression evaluator** from the **Effects** menu, you are presented with a dialog (a window) that is similar in appearance to a calculator.

On the left side, below the number buttons, is the **Destination** box. The destination is defined as the sound window where results of the evaluation will be placed. The box's drop-down list contains a list of all sound windows in the form "wavex", where x is the number on a sound window as given in its title bar. For

example, a sound window with the title "3 - HELLO.SND" would appear as "wave3" in the list. By default, the destination is set to the active sound window (the one on top). You can change the destination, if more than one sound window is opened, by using the up and down cursor keys or by selecting it with the mouse from the drop-down list.

A large Expression box is located in the middle of the dialog. This is where the expression is entered. To enter an expression, you can either type it in or click on "calculator" buttons with the mouse. A list of valid operations and functions is given in a following section.

At the bottom left side of the dialog are two boxes where the initial values of the variables 'n' and 't' may be entered.

After you have specified the destination, expression, and initial values, choose the OK button (or just press the "enter" key) to begin evaluation. If you entered an expression incorrectly, a message will be displayed in the **Message** box. The **Message** box also gives the progress during evaluation. Since the evaluation process takes time, you can stop it at any time with the Cancel button. Pressing the Cancel button a second time will close the Expression Evaluator dialog.

B.2 Evaluation Range, Variables, and Constants

Expressions are evaluated over the destination sound window's block. You must set start and finish markers before you open the evaluator dialog (Refer to Editing Overview for more information). Let's assume we have the following before opening the dialog:

Title bar: "1 - Untitled"
Sampling rate: 8000Hz
Start marker at 0.5 seconds
Finish marker at 1.2 seconds

After we open the dialog, we set:

Destination: "wave1"
Expression: "sin(t)"

The initial value for t is automatically set to 0.5. If you now choose OK, the expression "sin(t)" will be evaluated from t = 0.5 to t = 1.2 in steps of 1/8000 seconds. You will get:

wave1(0) = sin(0.500000)

wave1(1) = sin(0.500125)
wave1(2) = sin(0.500250)
...
wave1(3) = sin(1.200000)

Time Variable and Constant

The variable "t" is the time variable and, as demonstrated above, it changes throughout the start/finish range. The step of change is determined by the inverse of the destination's sampling rate. This inverse is assigned to a special constant, "T". In the above example, $T = 0.000125$ (i.e. $1/8000$). Constant "T" and variable "t" are automatically reset when a new destination is selected.

Sample Index Variable and Total Constant

The variable "n" is the sample index variable. It is always an integer value. During evaluation, this variable is continuously incremented in steps of 1. The sample index variable is used whenever an operation must be performed on an existing sample. A value of $n = 0$ (the initial default value) is the index of the sample at the start marker's position of the given signal. For example, the expression

$$\text{wave2}(n) + \text{wave3}(n+100)$$

takes the wave2 sample at wave2's start marker's position and adds it to the wave3 sample 100 samples after wave3's start marker's position. If a sample index is outside the wave's block, the value is 0 (i.e. $\text{wave2}(n-50) = 0$ while $n < 50$).

You must be careful when indexing signals with different sampling rates to get a desired effect. Assume wave1 is a voice recorded at 11025Hz and wave2 is music recorded at 22050Hz. If you want to sum (superimpose) the two signals with wave1 as the destination, then the expression

$$\text{wave1}(n) + \text{wave2}(n*2)$$

must be used (ideally, wave2 would have to be low-pass filtered first). Whereas, if wave2 is the destination, the expression should be

$$\text{wave1}(n/2) + \text{wave2}(n)$$

A constant "N" is set to the total number of samples in the evaluation range. It has several uses, such as reversing a sample. Assume wave2 is a new sound that has the same sampling rate and length of wave1. By setting the destination to

wave2 and using

wave1(N-n-1)

wave2 will be the reverse of wave1.

Conversion Between Variables

$$n = (t - \textit{initial value})/T$$

$$t = n*T + \textit{initial value}$$

B.3 Valid Operations and Functions

The following table summarizes the operations and functions included in the evaluator. Note that although the "%" operator is not shown in the evaluator dialog, it is a valid operator.

Label	Operation, function
(,)	Parenthesis
+, *, -, /	Add, multiply, subtract (negate), and divide
%	Modula operator (remainder)
^	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
log, ln	Log base 10, natural logarithm
exp	Exponential base e
step	Unit step (0 for $t < 0$, 1 for $t \geq 0$)
int	Integer value

rand(n) Random number between 0 and n
wavex(n) Signal value at n

B.4 Signal Generation

Generation expressions of several signals are listed below. Words given in italics are values that you must enter.

Type: Sine wave

General Expression:

$$\sin(2*\pi*frequency*t)$$

Examples:

Middle C:

$$\sin(2*\pi*261.7*t)$$

1000Hz and 750Hz tones:

$$(\sin(2*\pi*1000*t) + \sin(2*\pi*750)) / 2$$

Type: Square wave

General Expression:

$$\sin(\pi*\text{int}(2*frequency*t))$$

Examples:

1000Hz tone:

$$\sin(\pi*\text{int}(2*1000*t))$$

Middle C:

$$\sin(\pi*\text{int}(2*261.7*t))$$

Type: Saw (triangle) wave

General Expression:

$$1 - 2*\text{abs}(1 - 2*frequency*t\%2)$$

Example:

200Hz tone:

$$1 - 2*\text{abs}(1 - 2*200*t\%2)$$

Other examples of signals:

Noise:

$$1 - \text{rand}(2)$$

Exponential decay of 500Hz cosine wave to 50%:

$$\cos(2*\pi*500*t) * (0.5*\exp(-t) + 0.5)$$

Slow sweep (up to 20kHz):

$$\sin(2*\pi*160*(t\%5)^3)$$

The Sweep is a great way to see and hear aliasing if the destination signal has a low sampling rate. The ideal output should be a single tone increasing in pitch until it is nearly outside the range of human hearing. In reality, you will be able to hear two or even three tones, some rising in pitch and some falling!

B.5 Filters

One of the easiest ways to create digital filters 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 be used in the expression evaluator.

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 (1/4 the sampling rate). Enter:

$$[b,a] = \text{butter}(4, 0.5)$$

The result should be:

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 GoldWave, let's assume that the sound to be filtered is in the sound window titled "1 - SOUND.SND".

- 1) Use **File | New** to create a new sound window with the same sampling rate as "1 - SOUND.SND". The new sound window will be titled "2 - Untitled".
- 2) Make sure the length of the new sound is as long (or longer) than the original. If it is not, use **Edit | Insert space** to increase its length.
- 3) Use **Effects | Expression evaluator** to open the expression evaluator dialog.
- 4) Set the destination to wave2.
- 5) Enter the following expression:
$$\text{wave1}(n)*0.0940 + \text{wave1}(n-1)*0.3759 + \text{wave}(n-2)*0.5639 + \text{wave1}(n-3)*0.3759 + \text{wave1}(n-4)*0.0940 - \text{wave2}(n-2)*0.4860 - \text{wave2}(n-4)*0.0177$$
- 6) Choose OK (You might want to get a coffee while you wait).