Introduction

Wave Flow is an audio editor, so it can play, record, filter and modify a waveform, using lots of different functions, always following the WAV standard. Besides, as it has been designed for a Windows environment, it is done visually, easily and intuitively.

Those characteristics are obtained building a program that supports multiple windows or MDI documents (multiple document interface). Allowing the use of different windows simultaneously, the work that have to be done gets easier . If you have ever used a program not supporting MDI, you know that to interact between two or more waves, multiple sessions of the same program must be opened, but using a MDI program, there is no need to open more than one session of the program, because as many files as wanted can be opened from it, and they will be displayed in their respective child windows.

Once a sound is opened, it will be displayed in a child window, showing its characteristics in the status bar of each window. Clicking with the mouse on a child window will activate it. After selecting a part of the sound (keeping the left mouse button pressed over the sound), as many functions of the main program can be used.

The main program

The appearance and the use of Wave Flow, is very intuitive because it contains button bars, a menu and other visual elements. Besides, each button has a small hint (little help that appears when the mouse pointer is over a component a few seconds).

The bars that appear in the main program are the next:

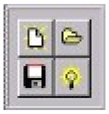
- Speed bar
- Options bar
- Play bar
- Selection bar

Besides, the menu options can be divided into:

- File menu
- Edit menu
- Tools menu
- Filter menu
- Miscellaneous menu

Speed Bar

Four buttons can be found in this bar. They allow you have a quick access to a set of file options that can also be found into the File menu.

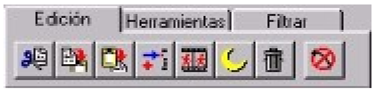


The available buttons are:

- New
- Open
-Save
-Save selection

Options Bar

This bar contains the buttons that allow a quick access to the most useful functions of the program. It is a multiple level bar, so more buttons in less space can be displayed. The buttons are classified into three levels: Edit, Tools and Filter.



The first level contains the speedbuttons that allow a quick access to the next functions: Cut, Copy, Paste, Mix, Repeat, Force silence, Delete and Undo.

In the second level (Tools), the next buttons can be found: Speed up/down, Efects, Reverb, Reverse, Normalize, Fade In, Fade Out i Amplify Volume.

In the third level (Filter) the next functions can be found: Advanced filter, Filter bank, Equalize, Low pass filter, Band pass filter, High pass filter, Median filter and Dynamic expander.

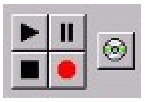
It is a filter that deletes the High frequencies, but allowing the pass of the low frequencies.

It is a filter that deletes the High and low frequencies, but allowing the pass of the medium frequencies

It is a filter that deletes the low frequencies (except for the offset component), but allowing the pass of the high frequencies.

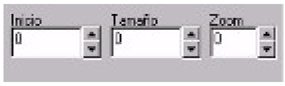
Play Bar

This bar contains the buttons that not only will let you play a part of a waveform, but also will let you pause, stop and record it.



The button with a CDROM Image (on the right of the Play Bar), shows a very easy and visual CD player, That will let you play any music CD.

Selection Bar



The selection bar, has three components. The first one (titled Begin), will let you display and change the beginning of the actual window, that can scroll through the waveform. The second one (Size), will let you change the selection size in samples. The Zoom edit box, will let you see and change the scale that will be used to display the waveform.

File Menu

You can use the commands on the File menu to manage the audio files in the program.

The commands on the file menu are the next:

- New
- Open
- Reopen
- Close
- Device information
- File information
- Save
- Save as...
- Exit
- Divide into multiple files
- Cd Player
- Save selection
- Print
- Print options

New

Creates a new wave (without any default format). This new wave will be represented as an empty window, where any wave can be pasted (previously cut or copied).

This option can also be called from the speebar.

Open

This command shows a dialog box that lets you choose an already created wave file (files with extension *.WAV). Once a file is selected, the program will open it and will display it as a wave.

This option can also be called from the speebar.

Reopen

This command shows the path and the name of the three last wave files that have been opened, putting them into a submenu. To open one of this files, select it clicking on its name, and the program will open it.

Close

Closes the waveform that is active. If the waveform has been modified in any way, the program will ask if you wish to save it. In that case the Save asdialog box will appear.

Device information

When this option is selected, a dialog box is displayed, which contains the characteristics and the information about the audio devices installed on the system.

File information

When this option is selected, a dialog box is displayed, which contains the characteristics and the information about the audio file that is active. Those characteristics can also be seen in the status bar of each window, and the path and the name of the selected file can be seen on the title of the its window.

path, number of channels, bits per sample and samples per second (sample frequency).

Save

To save a waveform there are two options:

-If the waveform of the active window has a file name, the waveform is saved on the old file. The old waveform will be replaced.

-If the waveform has no name (because it is new), the Save as... function will be called. You must also use this alternative function if you wish to change the name of the waveform file.

This option can also be found in the speedbar

Save as...

This command shows a dialog box where the final name of the file to be saved can be selected.

Exit

This command lets you close the actual session and exit the program allowing you to save the waveform files that have been modified .

CD Player

When this option is selected, a CD player is displayed on the bottom of the main window. This CD player can play, stop, pause, eject, move forward, rewind, move to next and last track, and shows information about the tracks and the time of the CD. It also lets you choose any track of the CD only by clicking on its button.

This option can also be found in the Playbar of the main program.

Save selection

Lets you save the piece of the waveform that is selected.

This option can also be found in the speedbar .

Print

Prints the waveform that is active with the configuration and values of the option Printer options. It doesn't print all the waveform, only the part that is displayed in the active window (including the selection, the selection cursor and the status bar of the window).

#Printer options

This command displays the dialog box of the default printer installed on your system.

Miscellaneous Menu

In this menu you will find miscellaneous functions that are not applicable to other menus.

The commands on this menu are the next:

- Colors
- Histogram
- Loop
- Zoom In/Out
- Audio Related Sites

Colors and Settings

Lets you change the colors of the Wave Flow screen (Line, WaveForm and Background). This option also lets you set Wave Flow as the default audio player/editor and as the default CD player.

Histogram

Lets you see the discrete probability function of the waveform and some extra parameters (mean, variance,...), that can be of some help in many cases.

Loop

Lets you select if the play of the sound will be cyclic or it will only be played once (default).

Zoom In/Out

Lets you select the actual zoom of the waveform using hotkeys (Ctrl++/Ctrl+-) with a Zoom Step of x2. You can also change the zoom with the zoom edit box or the zoom speedbutton placed under the zoom edit box.

Register

Wave Flow is a Shareware program. This means that if you like it after you have tried it or you are going to use it for commercial purposes, you must **register it**. I have put a lot of hard work to develop this audio editor, and I need money to continue developing new versions of this program.

There are two ways of registering Wave Flow 3.x:

a.-) Online Registration via Internet. You can visit Wave Flow website at http://www.waveflow.com and follow the instructions, or you can directly select Help Menu Register Online Registration.

b.-) Sending a letter with the money (a little fee of 25 US$) and the Registration Form to:

Xavier Cirac
c\ Batlles , 4
Premia de Mar , 08330
Barcelona
SPAIN

After receiving your registration mail, I will e-mail/mail you with your registration login and password so you can enter them to the program.

If you need multiple licenses of Wave Flow, you will have a very hard discount:

1 license:	25 US$
1-4 licenses :	23 US$ (8%)
5-10 licenses:	21 US$ (16%)
>10 licenses:	18 US$ (28%)

Thanks for registering Wave Flow.

Registration Form

Registration Form

Please Print the next form and fill in the information (if applicable) when you send me the registration letter:

Product: WAVE FLOW 3.x
Company: ____
Name: ____
E-Mail: ____
Address: ____
City: ____
State/Province: ____
Country: ____

Optional

Where did you find Wave Flow?

____

What are you using Wave Flow for?

____

Audio Related Sites (Internet)

This submenu will help you to find the most interesting audio related places (as well as some General sites) on the net. They are all rated and commented and they are perhaps the most useful sites to find software, help, tips, links, wav files...

Divide into multiple files

This option will be useful with the large wave files (>50 Mb) if Wave Flow seems to go too slow or gives you any memory error. Using this function you will be able to divide large wave files into pieces so Wave Flow can handle them.

Edit menu

The Edit menu not only has all the typical actions that all the clipboard functions that most Windows programs usually have in this menu (ex: cut, undo, copy, paste...), but also has some specific functions related to the audio processing programs (ex: rap, mix, insert and force silence...).

The functions in this menu are:

- Undo
- Cut
- Copy
- Paste
- Paste as new
- Mix
- Repeat
- Delete
- Trim
- Insert silence
- Force silence
- Select all
- Go to the cursor position

Undo

This option lets you undo the last action made to the waveform (the last modification), so there is the possibility to undo the mistakes that can be made.

Undo information is automatically save to a temporary file called wavedesh.tmp in your Wave Flow directory before you do anything that will change the waveform being edited.

Press Ctrl+Z to quickly undo the last operation.

Cut

Cut will copy the selection to a temporary file called wavecopy.tmp in the Wave Flow main directory, and remove it from the waveform being edited. Once cut, it can be Mixed o Pasted to other waveforms.

Use CTRL+X to quickly cut the selected wave.

Copy

This option realizes the same actions that the Cute2 option, but without deleting the selection. The selection will be copied to the temporary file called wavecopy.tmp to use it later with the functions Mix or Paste

Use CTRL+C to quickly copy the selected wave.

Paste

This options lets us insert the piece of wave that has been already copied or cut with the functions Copy o Cut. With this function different waveforms recorded in different moments can be added in the same waveform file. The selection will be pasted in the cursor position.

It is also accessible with the Ctrl+V shortcut.

Paste as new

This functions realizes the same that the Paste function, but creating a new child window where the selection already copied or cut will be inserted to. With this function we get a new file (initially without name), where we can edit the waveform without modifying the original audio file.

Mix

This function realizes a union of the actual waveform and the piece of waveform that has already been copied or cut. The result is a mix (a superposition) of the samples that we have been cut or copied and the samples of the actual waveform, starting on the cursor position.

This option is also accessible using the Ctrl+M shortcut.

Repeat

This function repeats the actual selection after the selection end.

This option is also accessible using the Ctrl+R shortcut.

Delete

This function lets us delete the actual selection of the active window. It is not the same that the Cut function, because now the selection will be loosed, not cut (only recoverable using the Undo option.

This option is also accessible using the Ctrl+Del shortcut.

Trim

This is the complementary function of Delete, because it deletes all the waveform except the actual selection.

Insert silence

This function inserts samples with an amplitude value = 0. This means that it inserts silence. The length of the silence inserted will be the size of the selection when this option is selected.

This silence will be inserted after the beginning of the selection, moving all the later samples after the end of the silence that has been inserted.

Force silence

This function takes the samples of the actual selection and force them to 0. This means that they will be silenced, so you will hear nothing.

Select all

This option selects all the waveform and displays the whole sound in the window, selecting it all. This means that the zoom, begin and size values of the selection barb will be modified.

You can also access to this function doubleclicking the status bar of the window that we wish to fully select.

This option is also accessible using the Ctrl+S shortcut.

Go to the cursor position

This function displays the waveform beginning in the sample that is written in the edit box called "Begin" placed in the selection bar of the main window of the program. This functions free us of having to look for the cursor all over waveform using the scrollbar manually.

This is a useful function when the zoom values are small.

Xavier Cirac

This is my name and I am the author of all you can see in this program. I have been working with some wave editors and I decided to build my own audio program, trying to get the best functions of all the other wave editors, and keeping off the useless options of all of them.

This software has been programmed using Borland Delphi 3.0© (a fantastic programming tool that gives a visual and easy way to make Windows programs).

I started programming this software in 1997 (March), and I am still working on it to get a better program. If you have any doubt about it, or find any mistakes or bugs please send me an e-mail to the address: <xevi@100mbps.es>. I will be very happy if you do so. If you wish to give me ideas for later versions, or give your opinion of something related to the program, you can mail me too.

To get a description of the program characteristics please click the next label:

<u>Wave Flow</u>

Filter Menu

Using this menu, you can access to all the filters (time filters and frequency filters), and all the functions related to the frequency domain. The functions that you can find in this menu are the next:

- Filter bank
- Advanced filter
- Median filter
- Improved median filter
- Average filter
- Tophat   filter
- Add noise
- Equalize
- Dinamic expander

Filter bank

This option shows a dialog box where a list of some predefined filters can be found. The values of this filters are fixed but they are the typical filters that you can find in any audio editor, so if you wish to configure any of them, you should use the function Advanced filter. Those filters, will delete or increase some frequency components of the sound (low, medium or high frequencies) to get different effects.

This function eases the use of the most common filters.

The predefined filters that you can choose in the selection list of the dialog box are:

- Low pass filter 2K
- Low pass filter 5K
- Low pass filter 10 K
- High pass filter 60 Hz
- High pass filter 100 Hz
- High pass filter 1K
- Bass Boost
- Bass cut
- Treble Boost
- Treble cut
- High Boost
- High cut

You can quickly access this functions using Ctrl+B.

Advanced filter

This is, perhaps, one of the most important functions in this program. It lets us realize a personalized filter, because with this function we can configure a set of parameters that will modify the response of our filter, making it more or less selective. When this option is selected, the next dialog box will appear:

At the top of the dialog box a frequency representation of the Fourier transform (FFT) of the peace of waveform selected will appear. Using the buttons of its right, we can select if we prefer a linear-linear representation, or a semi-logarithm representation (Y axis in logarithm scale and X axis in linear scale). This frequency representation will guide us to realize our filter. The rang of representation is: 0 Hz (offset component) to half the sample frequency (due to the Nyquist theorem).

Under the graphic we will find two trackbars that will let us select the low cut frequency and the high cut frequency of the filter (this response of this filter is represented with a red colored square on the frequency representation).

The selection box on the right of the dialog box, will be used to select the action that we wish to realize with the selected range of frequencies. This functions are:

- **Delete:** Deletes the frequency components included into the selected range (force them to zero).
- **Boost** : Increases the value of the frequency components included into the selected range. The ratio factor that will be used, can be selected with the small trackbar situated under the selection box.
- **Decrease:** : Decreases the value of the frequency components included into the selected range. The ratio factor that will be used, can be selected with the small trackbar situated under the selection box.
- **Band pass filter:** Using this function, all the frequency components that are **not** included into the selected range will be forced to zero. This is the default option.
- **Window Size:** You can select the size of the FFT used in the filtering transformation (>Window Size means >Spectral ressolution & >Processing Time).
- **Overlap:** You can choose the overlap into transformation windows.
- **Window Type:** It shows the available windowing functions (Rectangular, Hamming, Hanning, Barlett & Blackman)

We can see that combining all the options of this function, all the filters of the option Filter bank, can be done (and many more!).

You can quickly access this function using Ctrl+F.

Median filter

This option realizes the 3x3 median filter of the selected peace of waveform.

This filter realizes the median of the actual, anterior and posterior samples and assigns it the value of the actual sample. The median of three numbers is the value that is neither the lowest nor the highest.

The median filter is used to delete the scratch noise.

Improved median filter

This option realizes the 3x3 median filter of the selected peace of waveform. An improved median filter is a median filter with a threshold (under this threshold the signal will not be modified).

The median filter is also used to delete the scratch noise, being a little bit "cleaner" the median filter.

The scratch is a high frequency noise due to the saturation of the signal. For example, one of its causes can be the little jumps that the record player needle does when playing a record.

Average filter

This option performs the 3x3 average filter of the selected waveform. To do it, the arithmetic average of the actual, past and next sample is done, and the result of this operation is assigned to the actual sample.

In fact, this is a low pass filter because it deletes the high frequency changes.

Top hat filter

This option performs the 3x3 top hat filter of the selected waveform.

A top hat filter is a very useful filter to delete the scratch noise because it is a filter that only deletes this kind of noise, without modifying the signal that has no noise. It is perhaps even better that the median filter, because it is cleaner with the parts of the waveform that don't have scratch.

Add noise

This function add random noise to the selection of the actual waveform. The result that you can hear, is that a fried noise is added to the signal.

Equalize

This function shows a dialog box where an octave equalizer is displayed, whose attenuation values can be selected for each octave using a mobile trackbar. An equalizer is useful to attenuate or increase some frequency ranges, to increase the sonority of the waveform.

The mobile trackbars scale is in dB (decibels), and the frequency scale is in Hertz's (Hz).

This option is also available using the Ctrl+Q keys.

Dynamic expander

This option lets us perform an expansion of the dynamic range of our signal (number of dB of intensity variation that the emitter can generate), and in some cases it is useful to add richness to our waveform.

When this option is selected, a dialog box is shown, where the ratio of increase or decrease can be selected. This ratio is the value of the filter that will be applied to the waveform to expand it. If this value is greater than 1, this function will actuate as an expander, but if it is lower than 1, it will actuate as a compressor. If a too high value is used, the signal can be saturated, so the effect that we wished to get won't be reached.

Thanks

This program has been done thanks to the help of : Carmina Boter, Eloi Solà, Léonard Janer and Màrius Flaquer.

Tools menu

In the Tools menu you will find the general functions that will let you modify the waveform, adding lots of useful effects.

In the Tools menu you will find the next functions:

- Speed Up/Down
- Resample
- Reverse
- Invert channels
- Travelling stereo
- Normalize
- Fade In
- Fade Out
- Amplify volume
- Eco
- Reverb
- Effects
- Remove Offset

Speed Up/Down

This option displays a dialog box where some options can be configured, making the waveform faster or slower. This is performed resampling the waveform adding or taking off samples by interpolation. For example: If the speed is set slower, this will mean that the final number of samples will be increased. The final number of samples of the selection, can be easily selected through the edit box called 'Estimated number of samples'. This is very useful because it lets you synchronize the speed of two waveforms, only knowing the pitch or the period between two drums hits of one of them .

In this dialog box you will find as well the actual length of the selection (in samples), and the increase or decrease percent. All this information will determine the final length of the selection.

This option is also accessible with the Ctrl+Alt+V Shortcut

Resample

This option shows a dialog box that will let you change the actual format of all the active waveform.

In this dialog box you can find the actual format of the file of the active window, and three selection lists that will let you change the final format of the file. You will be able to change the number of samples per second, the number of bits per sample and the number of channels.

Once all the modifications have been done, the waveform will be shown with the new format.

Reverse

This function will let you change the order of the samples of the current selection. Once this function is called, the last sample of the selection becomes the first, and the first one becomes the last. This is makes the same effect that playing a record on the inverse direction.

This option is also accessible using the Ctrl+I shortcut.

Invert channels

This function is only accessible when the current waveform is stereo, because it performs an interchange between the two channels (left and right).

Travelling stereo

This function is only accessible if the current waveform is stereo.

This effect increases the volume of one channel while decreases the volume of the other one. It creates the effect that the emitter of the sound is moving from one side to the other.

By selecting this function, a dialog box will be displayed, where the direction of the movement can be selected (left to right or right to left). The percent of increase and decrease can also be configured.

Normalize

This function increases the volume of the actual selection of the active waveform, until its maximum value reaches the maximum value that the format allows. This function will increase the volume of your waveform, but without saturating it.

Fade In

Fade In will performs a progressive increase of the volume of the waveform making the initial volume lower than the final one.

This function will display a dialog box with some options to configure. First of all, there is a track bar that will let you select the percent of the initial decrease of the volume of the waveform (the final percent will be 100%). Secondly, you can also select the kind of function that will be used to increase the volume (linear, exponential or logarithm).

This option is also accessible using the Ctrl+Alt+I Shortcut.

Fade Out

Fade out is the complementary function to Fade in, because it performs a progressive decrease of the volume of the selection of the current waveform, beginning with a 100% of volume and ending with the percent that you can select using the track bar.   You can also select the kind of function that will be used to increase the volume (linear, exponential or logarithm).

This option is also accessible using the Ctrl+Alt+O Shortcut.

Amplify Volume

This function is useful to change the amplitude or volume of the current selection. You can change the percent of increase or decrease using the either the track bar or the manual configuration (writing a number into the edit box). A percent of 100% will leave the waveform with the same volume. Selecting a higher value, the volume will be increased, and selecting a lower value will decrease it.

This option is also accessible using the Ctrl+Alt+A Shortcut.

Echo

This function adds an echo effect to the waveform. You can choose between a simple echo (only one repetition) or a multiple echo (various repetitions). If you select "multiple echo", if can also configure the number of repetitions to perform.

This option is also accessible using the Ctrl+Alt+E Shortcut.

Reverb

This option displays a dialog box where the reverb parameters can be configured:

- *Reverb time*: Is the total delay time of the reverb. As high is this value, as big is the room you wish to simulate.

- *Initial delay*: It is the delay between the direct sound and the first reverb rebound.

- *Reverb delay*: it is the delay between the first reverb rebound and the big group of reverb components.

- *Density*: It means the number of reverb rebounds that will be generated. As high is this parameter, as dense and unintelligible will be the result sound.

This function has also some predefined options representing the most common rooms, to easy its configuration.

This function is also accessible using the Ctrl+Alt+R shortcut.

Effects (FX)

Wave Flow, has an small effects bank, that includes some curious effects and some of the effects given with the digital effects processors. They are:

- **Telephonic Line**: This effect performs a band pass filter using the telephonic bandwidth (between 300 Hz and 3400 Hz).

- **Robotic voice**: This effect simulates the voice of a robot.

- **Distortion**: This effect makes a distortion to the sound. All the samples whose value is higher than a configurable threshold are substituted by random noise added to the selected threshold. As low is the threshold you choose, as distortion you will get.

- **Compressor**: This effect compresses the dynamic range of the signal. All the samples whose value is higher than a configurable threshold are compressed by a configurable ratio. As low is the threshold you choose, as compression you will get.

- **Expander**: This effect expands the dynamic range of the signal. All the samples whose value is higher than a configurable threshold are expanded by a configurable ratio. As low is the threshold you choose, as expansion you will get. Another expander is the dynamic expander

- **Noise Gate:** This effect is used to remove the background noise (that is currently a low intensity noise). To get this effect, all the samples whose value is lower than a configurable threshold, are forced to zero.

- **Quantifier:** This effect divides the signal in a few values, rounding the value of the samples to the nearest one. For example, if you have a 8 bits sound (255 different levels), and you can quantify it to 9 different levels (0, 32, 64, 96, 128,160, 192, 224, 255). This effect adds a quantification error due to the loss of quantification values.

- Metallize: This effect adds a metallic effect to the sound by periodically forcing some values to zero (this means that some high frequency components are created). Depending on the period used and the number of samples forced to zero, you will have a Soft metallitzation or a Hard metallitzation.

- **Chopper**: This function inserts periodic blank spaces into the sound (samples whose amplitude is 0).

This function is also accessible using the Ctrl+Alt+E shortcut.

Remove Offset

This function removes the continuous component of the selection of the active waveform. This is done by calculating the average of the selected range. It has no any audible result, but it is useful if other functions like digital filtering have to be used.