Getting into the details

WAVELAB 7

Audio Editing And Mastering Suite

steinberg

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Chapter 1

WaveLab 7 Help

Welcome to the WaveLab Help browser. You can expand the tree structure in the panel on the left (under the Contents tab) to find individual Help topics. You can also use the Index and Bookmarks tabs, in addition to a powerful Search feature (under the Search tab).

Please see Getting Help for more ways of using the Help browser and other help features available throughout WaveLab.

For a quick overview of the features provided in WaveLab see Features Overview.

Top-level topics

Getting Help

Using the interface

WaveLab Concepts

Audio File editing

Audio Montages

Batch Processing

Podcasts

Control Window

Master Section

Master Project

Interface Elements

Audio Plug-ins

Utilities

How do I...

Glossary

Chapter 2

Getting Help

WaveLab's detailed help system makes it easy to look up interface features and get information from within the program. There are three main types of help available:

What is this?

Access this by clicking on the icon, or the question mark icon in a dialog window titlebar, or by holding down **[Shift]**+[F1]. Once activated, the cursor changes to a question mark. Click on an interface feature for information on what it does and details of any settings/-values/functions involved. This is a quick way to learn about the more detailed features of WaveLab.

When a Context menu is open, holding down **[Shift]**+[F1] allows the "What is This?" mode to activate while the menu is open, and this is actually the only way to get help for Context menu features.

It is worth noting too that "What is This?" help is sometimes available for sub-parts of controls, such as List headers.

Tooltips

Hold your cursor over a button to show a brief description of its function wherever available.

Help window

This window provides you with detailed information on WaveLab's features and functionality. You can access this help window in several ways:

- via the Help menu. See Help menu.
- from each dialog window provided with a Help button. This opens the Help window at the relevant entry.
- by pressing [F1]/[Command]+[?]. If a window is active, the Help window opens to the relevant entry. If a dialog box is open, the dialog is described.

Related topics

Help menu

Command bars

2.1 Help menu

The Help menu gives you a convenient point of access to WaveLab's Help system.

- **Help about the active window...**

  Select this (or press [F1]/[Command]+[?]) to give help on the active window. If a dialog box is open, the dialog is described.

- **Contents...**

  This lists the Help contents as a "tree structure" in the Contents window. Like many WaveLab windows, you can make this a floating window, if you want to park it in a convenient place on screen.

- **Index...**

  This accesses an Index to Help entries. Type in the "Look for:" box, or scroll down the alphabetical list. If there is more than one entry for the Index item, you'll be given a choice of entries to be displayed, in the "Choose Topic" box.

- **Search...**

  Type the name of the topic you wish to find in the "Search for:" box. If more than one topic is found to match the search string, a list is displayed: click on the most likely entry to display the topic. If there are too many entries, you can refine the search by clicking on [+] "Advanced search".

- **Bookmarks...**

  Click the [+] Add button to add the displayed topic to the list of Bookmarked favorites. Now you can return to the entry speedily.

- **What is this?**

  This gives a short "Tooltip" explanation by pointing and clicking on the interface feature you need information on. This can also be used to get help on any menu item. Activate the question mark icon by selecting "What is this?" from the Help system; using **[Shift]**+[F1]; or by selecting from the command bar button.

- **Steinberg on the internet** - This fly-out menu opens a window on your browser with the appropriate URL to support, update, upgrade or register WaveLab. You can also access the Steinberg Home Page.

Related topics

Getting Help

Chapter 3

Using the interface

This section gives information on interface features that may not be obvious and which may help to speed up your workflow, especially as your experience with WaveLab increases.

WaveLab is incredibly flexible in the ways that you can set up your working environment. Interface elements can be arranged to suit your workflow, the type of project you are creating and the size and number of your displays.

You can move, reorder and detach windows and tabs from frames and tear off groups of buttons in the Toolbar. You can slide and quickly partition different panes (areas) of the WaveLab window.It's easy to set up different layouts for different projects and switch between them. WaveLab has distinct Workspaces where you can work on a "session". Each Workspace lets you collect a number of files that you want to keep together. This could be all of the Audio Files used in a single song for instance, or all the Audio Files that go to make up a Podcast episode.

Tip: If you're a user of Nuendo (a Steinberg product), don't forget that you can drag files directly from within WaveLab.

Related topics

About Tool Windows

Adjusting Envelopes

Command bars

Context menus

Docking windows

Double clicking

Dragging operations

Playback shortcuts

Select-clicking

Shortcut system

Sliders

Status Bar

Switcher window
Tab Groups
Transport controls
Value editing
Workspaces
Zooming and scrolling

3.1 About Tool Windows

Throughout WaveLab there are various tool windows available that enable you to view, analyze and edit the current active document. Generally the content of a tool window is synchronized with the active document, with the exception of the audio meters which can display information related to other documents.

There are two types of Tool windows:

- **Specific tool windows** - these are dedicated to a given Workspace type. The File Browser and Marker windows are examples of specific tool windows.
- **Shared tool windows** - Only one instance of these windows can exist at any one time in WaveLab, and it can be shared between Workspaces. The Master Section is an example of a shared window and as such there is only ever one instance of it.

All tool windows can be accessed via the Workspace menu. See Docking windows for information on docking and undocking tool windows.

Specific Tool windows

Specific tool windows are specific to the current Workspace and are dedicated to that Workspace type. They perform useful tasks in the active Workspace. The File Browser and Marker windows are examples of specific tool windows.

Specific tool windows can be docked and un-docked anywhere within their Workspace, and saved in your custom layouts. They can also float as standalone windows in a specific Workspace.

Shared tool windows

Shared tool windows are shared between certain Workspaces and are independent of any one Workspace. Like specific tool windows, they can be docked and un-docked, and saved in your custom layouts. The Master Section is an example of a shared window.

The main difference with Specific tool windows is that only one instance of a Shared tool window can exist at any one time in WaveLab. When you open a shared tool window that is referenced in another Workspace it will "move" from the other Workspace to the active one.

A shared tool window can be moved and docked exactly like a specific tool window, however in addition Shared tool windows can "float as an independent window", that is, independent of any Workspace. Each shared tool window has a menu entry to allow this floating

mode.They can also be placed in the Control Window, in a tab group. Each shared tool window has a menu entry to allow this ("Place in Control Window").

Differences between Mac and Windows

Floating windows behave slightly differently on Apple Mac and Windows, due to some basic differences between the platforms:

- **Mac** - On the Mac, a tool window is always on top of all other windows and a floating window remains visible even if its dependent Workspace is not active or is minimized. If WaveLab is no longer the active application, all its floating windows are hidden.
- **Windows** - A floating window is hidden when its dependent Workspace is minimized or covered by another window. If WaveLab is no longer the active application, all its independent floating windows are hidden.

Saving a windows location

Once you have your tools windows, command bars and tab groups set up the way you wish to work, you can save their location in the current Workspace by choosing **Workspace** > **Layout** > **Save as...** and saving a preset; or you can save the current layout as the default layout by choosing **Workspace** > **Layout** > **Save current layout as default**.

If you wish to save the location of all Workspaces, then use the global option by choosing **Global** > **General Window Layout** > **Save as...** and saving a preset; or you can save the current layout as the default layout by choosing **Global** > **General Window Layout** > **Save current layout as default**.

Related topics

Workspaces

Shared tool windows

Specific tool windows

3.2 Adjusting Envelopes

At various places in WaveLab, envelopes are used to constrain the way effects are expressed. Some offline processes such as Level Envelope, use envelopes to adjust an effect over time. You can edit the profile of these envelopes quickly and easily.

Adjusting an envelope

Edit the envelope using the envelope editor window in the following ways:

- Create and erase points by double clicking.
- Select points by clicking on them.

- To select more than one point, press **[Ctrl]/[Command]** when clicking - you can then move them together. Selected points don't have to be contiguous. You can also click anywhere in the window and drag the cursor over the envelope points to select them.
- You can drag any segment to move the whole envelope profile up and down.
- You can drag a single segment vertically by pressing **[Ctrl]/[Command]** and move the segment up and down.
- You can drag a single segment horizontally by pressing **[Ctrl]/[Command]** + **[Shift]** and move the segment left and right.

You can also use the controls at the top of the editor window to delete or reset points in the envelope. Use the envelope smoothing button to toggle the envelope points between a linear (polygonal) or a curved path. The envelope is not applied until you click "Apply".

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Level envelope

Time stretching

Pitch correction

Pitch bend

3.3 Command bars

Commonly used tools, shortcuts and commands are represented by "Command buttons", each with their own icon. Related buttons are grouped into various Command bars. You can dock Command bars to any window edge and rearrange them. Each Workspace has an appropriate set of command bars that are available to be displayed. All the commands accessed via the Command buttons can also be found in the menus.

Hiding and displaying Command bars

To view a list of available command bars right-click on an empty part of the top edge of the Workspace. More precisely, on Windows computers right click anywhere along the window's top menu, on Apple Macintosh computers right click in the window's caption bar. This displays the Peripheral Windows context menu, which you can use to select which Command bars are displayed.You can also choose which individual Command buttons are displayed from the Customize commands dialog.

To move a set of Command bar tools

Drag on the left side of the command, or at top for vertical command bars - the pointer changes to ✣ when you have "grip". Drag the group of Commands to any side of the appli-

cation window and release the mouse button. The application window and other Command bars will automatically make space for the Command bar in its new position.

By default, Command bars are docked and don't float. You can make them float individually or not, by right clicking and selecting the desired option from the menu. Once the option has been selected, move the command bar.

You can drag Command groups below an existing row to form a new row (or next to an existing column, to form a new column). You can also reorder their position within a row or column, to customize your layout.

If a Command bar does not have room to display all of its icons, it will have a double right (or down) arrow >>. Click on the double arrow to reveal any obscured Command buttons.

Finding out what a Command button does

Hover over an individual Command button to display its tooltip. If no tooltip appears, make sure "Show tips when mouse stays over buttons" is activated in **Preferences...** > **Global Preferences** > **Display tab**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Global Preferences

Customize Commands

Context menus

3.4 Context menus

Throughout WaveLab there are various Context menus available. These menus group together a range of commands and/or options specific to the window you are working in. They appear when clicking with the right mouse button in certain places and may be useful for speeding up your workflow.

Some Context menus have colored title bars (these were formerly known as "Speed menus" in WaveLab 6), and their title indicates the part of the interface they relate to.

Using Context Menus

To bring up a context menu, right-click in the desired area. This is normally somewhere along the edges of a window, or in some cases anywhere on the window itself or its title. Right-clicking on a File tab, for example, shows a context menu with some relevant file options. Right-clicking on the waveform window header brings up the Time Ruler context menu allowing you to access a number options for changing the Time Ruler display format.

Tip: Most context menu commands can be found on the normal menu bar but **some unique commands are only found in context menus**. As they are not always obvious, when searching for a function, it may be worth checking for a Context menu in the window in which you are working by right-clicking on it, or around its edges.

Related topics

Shortcut system

3.5 Docking windows

All of WaveLab's tool windows can be used as docked windows or as stand-alone "floating" windows. They can be freely dragged around and docked at various locations. Command bars can also be freely moved around and docked along the edges of most windows. See Command bars for more information.

Undocking a window

Double-click on the title bar. This works with all tool windows, Specific and Shared.

About Specific tool windows

To undock a Specific tool window use one of the following methods:

- double-click on the title bar.
- click on the small double window icon which is located at the top left hand corner of the window.
- click and drag the window by its title bar.

Once un-docked, you can freely drag the window around by clicking and dragging on the window's title header bar.

To dock the window again, drag it to any location where you are visually prompted to release the window. Around the edges and the top of the Workspace is where this normally occurs. Most windows can be docked horizontally or vertically, but some can only be docked in one direction, because of the nature of their contents. To dock the window again, you can also double-click in the title bar.

Tip: To move a window and prevent it from docking, hold down **[Ctrl]/[Command]** before you begin to undock the window. This will avoid the window docking again and allow you to place it anywhere. Once you begin to drag, you can release **[Ctrl]/[Command]**.

About Shared tool windows

Shared tool windows such as the meters and the Master Section are slightly different to other tool windows as there is only ever one of them available and it can only appear in one Workspace at a time. As such, when you open a shared tool window from the menus in another Workspace it undocks and moves from its original Workspace. An empty tab container with a vertical title bar will remain in its previous Workspace.

To retrieve a window again from another Workspace use the button at the top left of these empty containers. For example, if you have the VU Meter displayed in the Audio Montage

Workspace and you wish to display it in the Audio File Workspace, you can simply click the blue button and the VU Meter window will move back into its tab container.

You can choose to deactivate the default moving behavior by deselecting **Workspace** > **Auto move shared tool windows**. Shared tool windows do not then move when you switch between Workspaces.

Docking/Undocking in the Control Window

Certain windows (those with a Window and Global menu in their title header) can also be arranged in the Control Window. These are placed centrally - slightly differently to how they behave in the other Workspaces. For more information on docking/undocking windows in the Control Window see Control Window.

Related topics

Tab Groups

About Tool Windows

Command bars

Control Window

3.6 Double clicking

A double click is the act of pressing the mouse button twice quickly without moving the mouse.

Throughout WaveLab you can achieve the following by double clicking in certain places:

- **Create a new empty document** - double click on the empty part of a tab bar. See Tab Groups for more information.
- **Edit a marker name** - double click at the right side of a marker's icon (in the Audio File or Montage Workspace) above the time ruler.
- **Select a region (Audio File Workspace)** - double click to select the whole Audio File. Where markers exist, the region between the markers is selected. If markers exist and you wish to select the entire file you can triple-click to select the whole file.
- **Set zoom to display entire waveform (Audio File/Montage Workspace)** - double click on the horizontal zoom wheel to reset the zoom to display the whole Audio File.
- **Set zoom to default value (Audio File/Montage Workspace)** - double click on the vertical zoom wheel to reset the zoom to its default value.
- **Center the waveform (Audio File Workspace)** - double click on the vertical scroll bar to center the waveform around the horizontal axis.

- **Start playback (Audio File/Montage Workspace)** - double click in the time line to start playback from that point.

In addition to these specific examples, in various places throughout WaveLab, double-clicking can be used to begin editing certain values. Depending on the context the type of editing triggered will vary. In most list views for example, double-clicking in a cell will allow you to edit its value.

Related topics

Tab Groups

Select-clicking

3.7 Dragging operations

WaveLab makes much use of "drag-and-drop" techniques to perform various operations, some of which can't be performed otherwise. These are referred to as drag operations in this documentation.

To drag an object, press and hold down the left mouse button while over the object and drag it to the desired destination. Drop the object by releasing the button.

Many types of objects can be dragged between different source and destination locations including files, text, Clips, the playback head and markers.

Tip: This help entry describes some of the most important drag and drop operations. There are many other possibilities, especially within the various list views for example where you can possibly drag from the list to an external window, drag a file into the list (to import), or reorder, these vary of course according to the context). Therefore we encourage you to try drag and drop wherever you feel it may make sense.

Within Workspaces

Drag objects within and between Workspaces to perform the following functions:

- **Docking a tool window** - drag a window's title bar to any side of the Workspace, beside or above an existing tool window, to dock the window at a new place.
- **Moving a Command bar** - drag the bar grip at the left-hand end of a command bar and reposition it. See Command bars for more information.
- **Moving a Document Tab** - drag the tab to a new location within its own tabbed group to reorder.You can also drag a tab to another group of tabs.
- **Open a file** - drag a compatible file from the WaveLab's File Browser, from the operating system's file browser or from another application to the tab bar.
- **Insert a file in another Workspace** - drag an Audio File by its title bar tab between the Audio File and Audio Montage Workspaces. You can also use the Switcher window to drag any object to another Workspace. Drag the object over the corresponding Workspace icon in the Switcher window and wait until the new Workspace becomes active. Then you can release the file in the target Workspace.

Within the Audio File Workspace

Drag objects within the Audio File Workspace to perform the following functions:

- **Paste an Audio File** - drag a document's title bar tab or document button onto the waveform area of another file, to copy and paste its audio content. You can also drag a compatible file directly from the File Browser, the operating system's file browser, or from another application.
- **Delete/Move a marker** - drag the marker along the time ruler to change its position. Drag while pressing the **[Shift]** key to create a copy of this marker. Drag above the time ruler to delete the marker.
- **Copy an audio selection** - drag a selected region of audio onto the waveform area of the same file, or of another file. Left, right or both channels can be copied.
- **Create a new file from an audio selection** - drag a selected region of audio onto the empty space of a tab bar to create a new copy of the audio. If dragging happens right after double clicking, the selection is quantized and extended up to the marker's "region" under the mouse.
- **Modify a selection** - drag left or right to change the extent of the selection.
- **Move the waveform** - drag anywhere above the time ruler to move it horizontally.
- **Create a marker from selected text** - drop text that you have selected in an external application onto the time ruler to create a generic marker there. The text becomes the marker's name.
- **Scroll the waveform** - drag the cursor to the left or right sides of the waveform area to display the region before or after.
- **Mix to Mono** - by holding down [Control + Alt]/[Command + Alt] while dragging the file to a new tab, you can quickly mix a stereo file to mono.
- **Convert from Mono to Stereo** - by holding down [Control + Alt]/[Command + Alt] while dragging the file to a new tab, you can quickly convert a mono file to stereo .

Tip: When aligning some objects in the waveform area, labeled prompts will help you to snap to other locations in the waveform (for example, the end points of Audio Files, markers or cursors).

Audio Montage Workspace

Drag objects within the Audio Montage Workspace to perform the following functions:

- **Paste an Audio File** - drag a document's title bar tab or document button onto the Montage area of another file, to copy and paste its audio content. You can also drag a compatible file directly from the File Browser, the operating system's file browser, or from another application.

- **Delete/Move a marker** - drag the marker along the time ruler to change its position. Drag while pressing the **[Shift]** key to create a copy of this marker. Drag above the time ruler to delete the marker.

- **Create a marker from selected text** - drop text that you have selected in an external application onto the time ruler to create a generic marker there. The text becomes the marker's name.

- **Scroll the Montage** - drag the cursor to the left and right sides of the Montage to display what comes before or after.

- **Paste a Clip** - drag a document's title bar tab or document button onto the Montage area to copy and paste its audio content. You can also drag a compatible file directly from the File Browser, the operating system's file browser, or from another application.

Within the Podcast Workspace

You can drag episodes in the Episodes list to reorder them.

Within the Master Section

You can drag effects between different Effects slots to change the order of processing (which takes place from top to bottom).

Within the Batch Processor Workspace

- **Change the order of effects** - you can change the order in which plug-ins are processed by dragging plug-ins within the Audio plug-in chain window.

- **Add to batch process** - you can drag a file (via its tab) to the batch conversion tool or batch processor, for "instant" processing.

Related topics

Docking windows

Switcher window

3.8 Playback shortcuts

Playing back audio is a common repetitive task in WaveLab. In addition to the playback command buttons in the Transport controls, there are a few extra commands that can only be accessed via special keyboard shortcuts. These shortcuts can be used even when the audio window is not the active window. These special keyboard shortcuts cannot be customized. They are:

- **Space bar** - Start/Stop

- **0** or **F7** - Stop. When pressed while already stopped, the cursor jumps to the previous playback start position. A further "0" returns the cursor to the start of the file.
- **Enter (Numeric Pad)** or **F8** - Start
- **[Ctrl]/[Command] + F6** - Plays from the beginning to the end of the selection, if any. Where there are markers it plays between them. If there are no markers it plays from the beginning to the end of the Audio File.
- **[Win]/[Ctrl] + F6** - As above, but playback starts with a preroll **before** the region and stops at its end.
- **[Shift] + F6** - As above, but playback ends with a postroll **after** the region.
- **[Win]/[Ctrl] + [Shift] + F6** - As above, but playback starts with a preroll **before** the region and ends with a postroll **after** the region.

Related topics

Transport controls

3.9 Select-clicking

Select-clicking is the act of selecting an object with the mouse and while it is selected, clicking again. This is a common interface interaction when renaming a file or folder in your computer's native file browser.

Throughout WaveLab there are places where you need to select-click fields to edit values. In some lists such as the Basic Audio CD window, you need to select-click cells to edit a track name or ISRC code. Generally if a value can be edited, a double click selects all characters in that cell and allows you to begin editing; if not, then try select-clicking the cell instead.

Related topics

Tab Groups

Double clicking

3.10 Shortcut system

Throughout WaveLab there are many predefined shortcuts and custom commands that can be used to speed up your workflow. The offline process "Fade-in", for example can be triggered from its menu entry, from its command bar icon, using a custom key sequence, and you can use MIDI events as well. Each shortcut is restricted to its specific Workspace, which means you can re-use the same shortcut in different Workspaces. When a command is available from more than one Workspace, then the shortcut applies to all Workspaces and cannot be reused.

Not all commands are restricted to a Workspace, for example, all Master Section commands are global to the application, and Record commands (Record dialog) are active only when the Record dialog has focus. In addition, a few specific Global shortcuts can be defined from **Preferences...** > **Global Preferences** > **Global shortcuts tab**.

Shortcuts are defined and can be edited in the Customize commands dialog. There are various places where you can access this dialog, and each dialog restricts editing of the shortcut to the related context. You can generate a summary of custom commands in a Print preview or HTML format from the dialog's "Summary" button.

There are several types of shortcuts available in WaveLab:

Key sequences

You can define a sequence of up to four keys to trigger a desired action. Note that these keys must be pressed/released one after the other, and the action is performed when the last key has been pressed. Key stroke sequences give you a very large choice of combinations, which is useful in an application with a large feature set. There are however possible pitfalls, if for example:

Shortcut #1 is defined with the sequence: Control+D, Control+E

Shortcut #2 is defined with the sequence: Control+D

The application watches for key strokes, and when Control+D is pressed, Shortcut #2 won't be triggered because Control+D is also used in a longer sequence (the application can't know if it should wait for Control+E or not).

Note: A few very basic shortcuts cannot be redefined. If this is the case, the relevant edit field is disabled. On the Mac, only the first key of native menus can be displayed.

MIDI event sequences

This is the same as with keyboard sequences, but using MIDI events sent from an external MIDI device instead.

Keywords

Unless you use them frequently, key sequences are not always easy to remember but keywords normally are. In WaveLab you can define any unique keyword to trigger a command. Type the keyword in the dedicated "Keyword" field in the "Definition of shortcut(s)" dialog, and click OK.

To trigger the command, enter the keyword in the Keyword command bar, found in all Workspace, and press Return. This command bar keeps a list of recent commands, to improve performance.

Keywords are useful in conjunction with presets, because a specific preset is easier to remember from a keyword (for example, "norm-1" to trigger a "Normalize to -1dB" process, if such a preset has been set).

Adding menu items to a Command bar button

Each Workspace has a number of default Command bars defined, but you can add further menu items to customize Command bars.

Each menu command with an icon can be potentially added to a Command bar. Commands are internally grouped by functionality, so the button always appears in its related command bar (File commands appear in the File command bar, for example).

Add a menu command using the Customize commands dialog. Each command has a check-box in the "Bar" column.

Other shortcuts

There are many other shortcuts in WaveLab that are accessed using the mouse. These are described in Double clicking, Dragging operations and Tab Groups. Clicking the right button on the mouse in certain places will produce a Context menu which offers quick access to relevant commands.

Related topics

Customize Commands

Definition of shortcut(s)

Double clicking

Dragging operations

Tab Groups

Context menus

3.11 Sliders

At various places in WaveLab, slider controls are available to change parameters. There are a number of ways to change a slider's value:

- Turn the mouse wheel over the slider (no click is required). Press **[Ctrl]/[Command]** simultaneously to scroll faster. This modifier also applies to the zoom wheels.
- Left click on the button to move it.
- Left click outside the button to move the button directly to that position.
- Right click before or after the button, to move the button step by step. Keep the mouse button pressed to automatically step to the next value.
- Middle click anywhere to reset the slider to the default value (if a default is available).
- Left click and press **[Ctrl]/[Command]** anywhere to reset the default value (if a default value is available).
- Double click on the button to reset the default value (if a default value is available).

Related topics

Adjusting Envelopes

3.12 Status Bar

The Status bar is normally displayed at the bottom of the screen and shows information about the active window using the units specified in the rulers.

The Wave and Mouse cursor time positions are shown using the same units of time as in the rulers and dialog boxes. Level positions are always shown in dB. This information is updated depending on the cursor position and whether you have a region of audio selected. In the case of the Audio File Workspace it gives the following information about the Audio File. From left to right:

- **Time/Level (dB)** - at the mouse cursor position.
- **Wave cursor Time** - displays the time at the position of the Wave cursor. This information changes when you click to reposition the cursor. Click on this status bar field to show the Cursor position dialog, which you can use to accurately position the cursor.
- **Audio selection indicator** - displays the length of the current selection, or the total length of the Audio File if there is no selection. Right-click on this to display the entire audio range.In WaveLab Left-click also opens the Audio Range dialog, so you can define or refine a selection.
- **Zoom indicator** - displays the current zoom factor. Click this to edit the zoom factor; right-click to display a pop-up menu (equivalent to selecting **View** > **Zoom**).
- **File format details** - displays the number of audio channels, bit resolution and sample rate; Click to display Audio properties dialog.
- **Document button** - allows you to click and drag the current Audio File into another document (for example, in the Audio Montage Workspace). This is equivalent to dragging the file Tab.
- **Sample Key indicator** - indicates the key of the current Audio File (if defined) and opens the Sample Attributes window. The status bar also indicates the progress of some background operations, such as rendering an effect. The operation can be paused or canceled using the buttons provided.

The "Play through Master Section" buttons and allow you to completely bypass the Master Section for playback, or not, with a setting for each file (Audio File and Montage).

There are also the Save/Restore Master Section presets buttons, which allow you to store/restore these settings for each file. See Save Master Section Preset.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Time Format dialog

Zoom factor

Audio properties dialog

3.13 Switcher window

This small floating window allows you to switch between each of the main Workspaces in WaveLab, and also to open and create documents.

It is visible at all times and floats above all other windows. You can choose which Workspace buttons are visible and adjust its transparency in Global Preferences. You can resize, rotate and move the Switcher window anywhere on your monitor(s).

- It is useful to use a file in a different Workspace. For example, you can drag an Audio File from the Audio File Workspace over the Audio Montage icon in the Switcher window and wait until the Audio Montage Workspace becomes active before inserting the file in a montage.
- To switch between Workspaces: click on a Workspace button and if the Workspace is not already open, a menu opens and offers several options.
- If two instances of the same Workspace are open, you can toggle between them by repeatedly clicking on the appropriate icon.
- To create a new document in any Workspace: click + **[Ctrl]/[Command]** on a Workspace button.
- To open a document in any Workspace: click + **[Shift]** on a Workspace button.
- To display a menu listing the files used recently in a particular Workspace, click on any Workspace icon and hold the button down. This means that with just one click you can access the history of a document from any Workspace. From the same menu, you can also create a new document or open the file selector box. If you click on the icon of the active workspace, you don't need to hold the button down. Alternatively, you can right click to show a menu instantly.

Related topics

Workspaces

Docking windows

3.14 Tab Groups

A tab group is a region within a Workspace that contains windows that can be activated by a tab, each with its own content and title bar. Only one tab can be active at a time.

In an editing Workspace, a tab group contains one or more documents (e.g. Audio Files).In the Control Window, a tab group contains one or more shared tool windows. In WaveLab it is possible to create more than one tab group.To activate a tab, simply click on the title bar of the tab you wish to work on.

Tip: Because it can be useful to quickly display multiple files side by side, the Audio File Workspace has a **Tile** mode that can be activated from the **Tabs** menu. To make this possible, a single tab group must exist.

Using Tabs for editing audio

Tabs have a variety of useful behaviors depending on the type of window contained by the tab group.Tabs that display meters and other tool windows can be dragged and moved around in the Control window.Tabs that contain Audio Files (in the Audio File Workspace) symbolize a whole Audio File and where applicable allow you to perform many other useful functions:

- **Reorder the tab bar** - by dragging and dropping the tab at a new position on the tab bar.
- **Move it to another Workspace** - by dragging and dropping the tab in the editing window of another Workspace.
- **Paste it into an Audio File** - by dragging and dropping the tab onto the waveform display at the cursor position you want to insert the copied audio.
- **Paste it into another application** - by dropping the tab into the relevant window of the other application. The result depends on the target aplication.

Tip: You can also achieve the same actions by dragging the Audio File from the document button on the status bar at the bottom right of each window.

Other useful functions can be accessed using tab groups:

- **Create a new empty document** - by double clicking on an empty part of the tab bar (on the right normally). The new file is created by using the active file as template (for example, Sample rate).
- **Display the document file path** - by keeping the mouse over a tab for a small moment. This displays a popup that indicates the full file path of the associated document.
- **Access additional tab functions** - by right clicking on a tab.
- **Access the Recent file list** - by Right-clicking in the empty tab area.
- **Move a file between tab groups** - by dragging a tab from one tab group to another.
- **Copy a file** - by holding down **[Ctrl]/[Command]** to create a copy by dragging the file elsewhere in the tab area.
- **Mix to Mono** - (Audio File Workspace) by holding down [Control + Alt]/[Command + Alt] to mix a stereo file to mono while dragging the file to a new tab.

- **Convert from Mono to Stereo** - (Audio File Workspace) by holding down [Control + Alt]/[Command + Alt] to convert a mono file from mono to stereo while dragging the file to a new tab.

- **Quickly switch between tabs** - by moving the mouse wheel while over the tab bar. This switches the focus between the different tabs.

Tip: An option in global preferences allows you to hide the title bar if there is only one tab open **Global preferences** > **User Interface** > **Show tab if there is a single window**.

Related topics

About Tool Windows

Docking windows

3.15 Transport controls

The transport controls allow you to control playback within an Audio File or montage, and to open the record window. It can be accessed via the Transport menu or the Transport toolbar. See Command bars for more information.

The Transport menu gives access to (and shows keyboard shortcuts to) the same functions that are available in the transport toolbar.

The Transport toolbar gives quick access to most of these functions. If the toolbar is hidden, select **Workspace** > **Command bars** > **Transport commands**. Transport functions operate on the current Audio File.

The basic transport functions are:

- **Loop On/Off**
  - Toggles looping on/off for the currently selected Audio File or montage.

- **Start of file / End of File**
  - Move the cursor to the beginning or end of the currently selected Audio File.

- **Skip forward / Skip backward**
  - Skip the cursor forwards or backwards within the currently selected Audio File.

- **Play**
  - Begin playback of the current Audio File.

- **Stop**
  - Stop playback of the current Audio File. A second click moves the cursor to the beginning of the last start position. A further click returns the cursor to the start of the file.

- **Record**
  - Opens the recording window from which you can start recording.

Additional transport functions such as start, skip and stopping playback options are also available from the transport command bar. In WaveLab additional playback speed and jog/shuttle shortcuts are also provided.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Playback speed

Recording Dialog

Command bars

3.16 Value editing

At various places in WaveLab, numerical values can be edited using a combination of text fields and spin controls.

These values are sometimes made of several parts, for example, 12 mn 30 sec 120 ms. Each value can be edited using the following methods:

- With the value selected, type a new value.
- Use the up and down arrow keys to change the value by one unit at a time.
- Use the page up and page down keys to change the value by several units.
- Use the home and end keys to jump to the maximum and minimum values.
- Use the left and right arrow keys to move from one part to another.
- Click on the small button arrows (the spin controls), to change the value.
- Move the mouse cursor over the desired section. Now use the mouse wheel to change the value, **without** clicking. The hovered section will be highlighted when the wheel spins.
- Move the mouse cursor over the part you wish to change. Click and drag up and down to change the value.

Tip: Right clicking on the edit control may display a context menu, where additional functions are available.

Related topics

Sliders

3.17 Workspaces

A Workspace is a window that provides an editing and playback environment for a particular audio document type. Each type of Workspace presents the functions related to the specific file types it is made for; this reduces clutter in the user interface.

In WaveLab there is a Workspace for each main type of file to edit:

- **Audio File Workspace** - for viewing and editing Audio Files. See Audio File editing
- **Audio Montage Workspace** - for assembling and editing audio montages. See Audio Montages
- **Batch Processing** - allows you to process a list of Audio Files with offline effects, VST-plug-ins and Master Section presets. Each file is processed and then outputted to a folder of your choice. You can run batches repeatedly if required. See Batch Processing.
- **Control Window** - for hosting and organizing tool windows (e.g. meters), especially in a multi-monitor setup. You can choose not to use it if you do not find it convenient, since removing the Control Window does not remove any features of WaveLab. See Control Window.
- **Podcast Workspace** - for preparing and uploading Podcasts. See Podcasts

A Workspace is highly customizable, to match your workflow, experience and screen setups (which may vary from a studio-based multi-screen DAW to a single-screen laptop on the road). A Workspace can appear in a wide variety of styles, from a simple window with a single menu in its most uncluttered form, to a sophisticated arrangement of Command bars, tool windows, Tab groups and active meters.To meet the varied needs of users, custom layouts can be saved (using **Global** > **General window layout** > **Save as...**) and restored from a menu or shortcut. When a file is opened from a given Workspace, it is added to the active tab group of this Workspace.If a file is opened from the application Workspace, a new Workspace is created for it. You can have several instances of the same Workspace type open simultaneously (for example, you could have two Audio File Workspaces open with different layouts). However, for performance reasons it is generally better to have just one instance of a Workspace type open at a time. You can drag files between Workspaces if they both share compatible formats. For example, you can drag an Audio File from the Audio File Workspace and drop it into the Audio Montage Workspace by using its tab bar or its document button (this is located on the status bar at the bottom right of the Workspace). For more information on using shortcuts to speed up your workflow in different Workspaces see Dragging operations and Tab Groups.

Elements of a Workspace

Each Workspace is comprised of:

- A **Menu bar**. Each Workspace has a different Menu bar, but certain menus are shared among all Workspaces and each menu can be customized in various ways. The Workspace menu has a sub-menu to show/hide the available Command bars and tool windows. Alternatively, you can right-click in the menu (Windows) or caption bar (Mac) area to access the Peripheral Windows Context menu.

- One or more **Command bars**, for quick access to functions via buttons. Command bars can be customized extensively. See Customize Commands.

- **Tab groups**, to host the document(s) to edit. This is a central part of the Workspace: you can reorder the tab bar, move it to another Workspace, create a new empty document, display the document filepath and access other functions by right-clicking. See Tab Groups.

- A set of **Specific tool windows**. The specific tools offered vary according to the Workspace; their display can be turned on or off individually, and the windows can float or be docked. See Specific tool windows.

- A set of **Shared tool windows**. The shared tools offered vary according to the Workspace; their display can be turned on or off individually, and the windows can float or be docked. See Shared tool windows.

Tip: To switch easily between different Workspaces use the floating Switcher window.

Related topics

Audio File editing
Audio Montages
Podcasts
Control Window
Batch Processing
Customize Commands
Specific tool windows
Shared tool windows
Command bars
Tab Groups
Docking windows
Switcher window

3.18 Zooming and scrolling

In WaveLab you can use several mouse and keyboard shortcuts for scrolling and zooming in the Audio File and Audio Montage Workspaces.

Scrolling with the Mousewheel

- With the cursor in the wave window you can scroll the view horizontally using the mouse wheel.

Zooming with the Mousewheel

- With the cursor in the wave window hold down **[Ctrl]/[Command]** while rotating the mouse wheel. The view will zoom horizontally; if you hold down **[Shift]** the view will zoom vertically.

Activating the zoom tool

- Activate the zoom tool by holding down **[Ctrl]/[Command]** and click on the waveform (**[Ctrl]/[Command]** must be pressed first).

Zooming with the Mouse

- Holding your cursor over the Time Ruler and moving the mouse up and down also zooms the view horizontally. Hold down the **[Shift]** key to do the same thing while maintaining the edit cursor in the same position.

- Zoom by dragging the thumb edges of the scroll bars in the Audio File and Montage windows.

- Zoom by acting on the wheel controls in the bottom right corners of the Audio File and Montage windows, by click and drag or using the mouse wheel.

- Zoom and scrolling can be undone or redone by the **View** > **Navigate backwards/-forwards** commands.

Changing the Zoom level

- you can make quick changes to the zoom level using the View All or Zoom In 1:1 buttons in the control bar. Scrubbing the vertical or horizontal Zoom Wheel (located in the bottom right hand corner of Main wave window) also alters the zoom level.

Related topics

Shortcut system

Chapter 4

WaveLab Concepts

WaveLab is an incredibly powerful tool and has many features which can be used for a wide variety of audio editing, mastering and mixing purposes. This section provides you with links to some of the main concepts used throughout WaveLab. Understanding these concepts can help you to get the most out of WaveLab.

Related topics

Features Overview

Analysis

Authoring CDs and DVDs

Markers

Metering

Offline processing

Presets

Rendering

Scripting

The Spectrum display

Spectrum editor

About Regular Expressions

4.1 Features Overview

WaveLab 7 Pro - Audio Editing and Mastering Suite

WaveLab 7 Pro is the all-in-one solution for professional mastering, high resolution multi-channel audio editing, audio restoration, sample design, radio broadcast and podcasting work, right through to complete CD/DVD-A production. For some time, WaveLab has been the standard application for digital audio editing and processing due to its outstanding flexibility and pristine audio quality. Version 7 of the application adds a dazzling collection of

features to satisfy the needs of top professionals and audio enthusiasts.

- Sample accurate time and spectrum audio editing
- Now available for both the Mac and PC platforms, ideally suited to high performance multi-screen setups
- Excellent performance and outstanding audio quality with support for all common bit-rates: 8-, 16-, 20-, 24- and 32-bit floating point resolution, at up to 384 kHz.
- Powerful Audio Montage for simultaneous non-destructive editing across several tracks
- Red Book-compatible CD mastering and Audio CD/DVD-A authoring and burning
- Comprehensive suite of real-time metering and analysis tools, including global analysis, 3D-spectrum analysis, VU meter, spectroscope, oscilloscope, phasescope and a powerful Spectrum editor
- A large library of plug-ins are provided, including the entire Steinberg collection, Sonnox audio restoration and batch specific plug-ins.
- An unlimited number of audio lanes, with unlimited file size (over 2GB)
- Extensive batch file processing and full scripting for professional users
- A totally flexible and customizable GUI: store master project custom file sets, save multiple Workspace layouts and tabs. Menus, keyword commands and MIDI shortcuts may all be customized.
- Suite of offline processors provided, including effect morphing, pitch bend, error detection and correction and high-end DIRAC ©time stretching and pitch correction
- Full range of different marker types for looping, CD creation, tagging errors, etc.
- Full range of tools: Auto split, Audio File comparison, signal generation, programming utilities, etc.
- Author and publish podcasts directly from within the application
- All major Audio File formats are supported, including WAV, AIFF, AU, MP3, MP2, OggVorbis, Windows Media 9, AES-3, RIFF64, Sony Wave64, Raw.

4.2 Analysis

WaveLab provides you with a comprehensive set of tools for analyzing your audio and diagnosing any errors. You can view your Audio File in many different ways using the suite of audio meters, via its frequency spectrum, or even in three dimensions. There are also several tools to interrogate any sample of your audio and find any errors or anomalies.In WaveLab

you can even compare two Audio Files with the Audio file compare tool and view audio in a Spectrum or Loudness view for greater precision when editing. Below you can find links to some of these tools used in WaveLab. Most of them can be found in the **Analysis** menu. The meters can be found in the **Meters** menu or via the **Workspace** > **Shared tool windows**. More specialized tools also exist as part of some offline processing dialogs, such as the "Find current peak level" command in the Change level dialog, or the "Eliminate DC Offset" dialog, for example.

Related topics

Global analysis

Audio file comparer

3D Frequency Analysis

Loudness distribution

Metering

The Spectrum display

Audio File editing

Offline processing

4.2.1 Global analysis

This dialog allows you to perform advanced analysis on your audio to identify areas with specified properties. It can be used to find problem areas such as glitches or clipped samples, or just to check general information such as the pitch of a sound.

How it works When you analyze a section of an Audio File, WaveLab scans it and extracts information which it displays in the dialog. It also "pin-points" sections of the file that meet specific characteristics, for example, sections being very loud or almost silent. You can then browse between these points, set markers or zoom in on them.

Types of analysis On most of the tabs, you will find settings determining exactly how the analysis will be performed. Each tab focuses on a particular analysis area:

- **Peaks** - this tab is for finding individual samples with very high decibel values.
- **Loudness** - this tab is for finding sections that are perceived by the human ear as louder or weaker in volume. WaveLab uses an accurate method (RMS - Root Mean Square), to measure a consecutive section of samples and then average their value.
- **Pitch** - this tab is used for finding the exact average pitch of a section of audio. The method works best on monophonic material (single notes, without chords or harmonies) and assumes the analyzed section has a relatively stable pitch. As a general rule try to analyze the sustain portion of a sound, rather than the attack.
- **Extras** - this tab is for informing you about DC offsets and the significant bit resolution of the file. This can be useful, for example, if you wanted to check whether a 16 bit file really uses 16 bits (or if it was actually recorded with only 8 bit resolution and then upscaled to 16 bits).

- **Errors** - this tab is to help you find glitches and sections where the audio has clipped (gone beyond 0 dB).For more control over the analysis, use Error detection and correction. **Checking and browsing the results** Checking the results on the Pitch and Extra tabs is simple, since only one value is returned for the whole section of analyzed audio. For all the other tabs, the analysis methods returns a series of exact points ("hot points") in the file or section of audio. Use the relevant buttons in each tab to select the "hot points" you wish to look at in more detail. Using the slider you can then browse or skip between these points, adding markers or focusing the waveform display at that particular point in the file.

You can access this dialog in the Audio File Workspace via **Analysis** > **Global Analysis...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Error detection and correction

3D Frequency Analysis

4.2.2 Audio file comparer

This analysis tool allows you to compare two Audio Files for differences between them. It can create a "delta file" that contains these differences. It can also drop markers at positions where differences are found.

This can be useful for:

- Judging the effect of using an equalizer by comparing the file before and after. The delta file will show what was added or removed.
- Check the noise added by a processor.
- Compare two digitally recorded files to check for dropouts.

Note - Make sure that the two documents you want to compare are open.

You can access this dialog via **Analysis** > **File comparator...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

4.2.3 3D Frequency Analysis

This function in WaveLab allows you to view a wave file in both the frequency and time domains. Although a wave display (time domain) tells you a lot about where one sound starts or ends in a file, for example, it doesn't tell you anything about the timbral contents of the file. A frequency graph (frequency domain) allows you to investigate the individual frequency components of an Audio File. With the addition of the time dimension you can follow these frequencies in your Audio File over time and pick out individual snare hits or

vocal notes in a song, for example. The graph used in WaveLab is sometimes called a 3D-Spectrogram.

Displaying and viewing the graph

You can choose to analyze a selection of audio or an entire file. If you select a stereo recording, a mix of the two channels will be analyzed. Once you have your region selected or no region (in the case you wish to analyze the whole file) choose **Analysis** > **3D Frequency Analysis...**. The graph window will appear.

There is a scrub wheel that allows you to rotate the view around and a settings button that displays the range of frequencies currently displayed. Click on this button to open the settings dialog and edit the frequency range displayed, as well as to access other options.

Tip: The length of the selection affects the accuracy of the analysis. For short selections the result will be very detailed. For longer selections (over a minute or so) the results are generally less detailed as the harmonic content may vary between "measuring points", and as such are not included in the graph. You might for example make a separate analysis of the attack (beginning) of a sound, since the most drastic variations usually occur there.

Some example uses for 3D Frequency Analysis

The 3d Frequency Analysis graph in WaveLab can be used for many purposes, for example:

- to see how the frequency spectrum is distributed in a mix.
- as a basis for EQ-ing, so that you know which frequencies to reduce or boost.
- to see which parts of the audio spectrum a certain background noise occupies (for removing by filtering).
- for educational purposes - these graphs tell you a lot about how different sounds are "built".

For detailed spectrum analysis, the 2D Spectrometer remains the best choice, as it provides more accurate viewing and more options.

Related topics

Frequency range

Metering

4.2.4 Loudness distribution

This dialog allows you to measure the most frequent loudness values found in an Audio File (this is not the same as average loudness). The graphic peaks found by the analysis represent these values.

The tool gives an answer to the question: "How often does a given loudness (vertical scale, in dB) appear in the whole file?". The percentage figure is relative to other peaks. Some simple examples:

- 1. If, for instance, you have a sine of 0 dB / 2 second followed by a sine of -6 dB / 2 seconds. This means the global audio material has as much 0 dB material as - 6dB material: you will see 2 peaks (0 / -6 dB), each at 100%.
- 2. If, for instance, uou have a sine of 0 dB / 1 second followed by a sine of -6 dB / 3 seconds. This means there are 3 times more -6 dB material than 0 dB material. You will see a peak of about 33% at 0 dB, and another peak of 100% at -6 dB.

This analysis is useful for learning about how the loudness of your music is distributed.

You can access this dialog in the Wave File Workspace via **Analysis** > **Loudness distribution...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

4.3 Authoring CDs and DVDs

WaveLab provides extensive tools for authoring all types of audio and data CDs or DVDs. Depending on your connected optical hardware you can burn CD-R, CD-RW and DVD-R/DVD+R disks. WaveLab allows you to author professional audio CD and DVD-Audio disks with full control over track indexes, CD-Text, PQ and ISRC/EAN codes etc. You can validate your disks before burning to make sure they conform to valid CD standards (such as the Red Book standard) or choose to author using newer formats with CD-Text support or gapless tracks. When it comes to outputting your final disk you can choose to output your audio CD as a DDP image or burn it straight to optical media. If you are creating a DVD-A, you can output to AUDIO_TS and VIDEO_TS folders or burn a DVD straight from within WaveLab. You can also burn a data CD or DVD directly from the application or save them as ISO disk images.

Important note for Mac OSX users

When using the CD-R/DVD-R drive on Apple OSX computers for importing or burning it is important to note the following. If you insert a optical CD or DVD disk in the drive while the drive is not selected in WaveLab for import or burning, WaveLab will not be able to "see" the drive. The disk drive is "taken over" by other applications under Mac OSX and is not seen by WaveLab when this happens. To avoid this occurring, make sure there is no disk in the drive when opening any CD window within WaveLab.

Audio CD authoring basics

You can author an audio CD from within both the Audio Files and the Audio Montage Workspaces. From the Audio Montage Workspace you can additionally create DVD-Audio disks and you have a more comprehensive set of tools for creating audio CDs and DVDs. Many of the tools provided in WaveLab also work together in conjunction to enable you to create your own workflow for audio CD authoring.

Track markers and CD authoring

In WaveLab one of the main concepts to understand about audio CDs is the use of markers to define tracks, and the breaks between tracks. Each track must at least have a track start and track end marker. Some of the various authoring tools provided by WaveLab will generate these track markers for you automatically but you can also manually define, modify and edit them using the marker window or directly in the main editing window. You can also use other marker types to determine where track indexes are and to cross fade two tracks together seamlessly to create a gapless CD. See "CD/DVD-A authoring markers" in Marker types for more information on the specific function of different markers.

Authoring in the Audio Montage Workspace

In the Audio Montage Workspace you can use the following tools to author your audio disks:

- **CD window** - this window contains a variety of tools for authoring and burning CDs with a great deal of precision and control. It even has a CD Wizard to create CD tracks quickly from your montage Clips. This window provides full control over your audio CD. See CD window for more details.

- **DVD Audio window** - use this window for authoring and burning DVD-Audio disks. It uses Audio Montage files to create tracks for your DVD-A disk. See DVD-Audio window for more details.

- **Marker window** - use this window in conjunction with the CD window to edit CD track names and times. You can also drag markers in the main Audio Montage editing window to adjust start/end times and double-click on a marker's name to edit its marker/track name.

Authoring in the Audio File Workspace

In the Audio File Workspace there are also some tools to allow you to create and author audio CDs, these include:

- **Basic Audio CD** - use this window to quickly create an audio CD from individual Audio Files. Any track markers you have previously created will be used to define tracks or WaveLab will automatically define track markers for you for each Audio File if you haven't created any. If you wish to have more control over your audio CD, you can export your Basic Audio CD session to the Audio Montage Workspace. You could do this for example, if you wanted to create a DVD-Audio disk or wanted to add and adjust cross fading between tracks. See Basic Audio CD window for more details.

- **Marker window** - use this window to add and edit CD track names and times to an Audio File. You can also drag markers in the main Audio Files editing window to adjust start/end times and double-click on a marker's name to edit its marker/track name. Your markers are saved along with Audio Files (unless you deactivate this option in preferences) and are used by WaveLab for example, if you import the file later to create a Basic Audio CD.

Authoring a data disk

You can also burn data CDs and DVDs using the Data CD/DVD window throughout WaveLab (if you wish to back up all your session files, for example). You can access this window via **Utilities** > **Data CD/DVD...**. You can add any file to a new data CD or dvd at anytime via the **File** > **Special** > **Add to DATA CD/DVD** function in both the Audio File and Audio Montage Workspaces. Note that all files referenced by the data disk must be saved before attempting to create a data CD/DVD. For more information see Data CD/DVD.

Burning your CD/DVD

Once you have compiled an audio CD using any of the tools mentioned you can burn it by choosing the button which opens the Write Audio CD dialog. Here you can choose either the device you wish to burn with or to save the disk as a DDP image. This dialog also allows you to choose standard options as to how the disk will be burnt. For more information see Write Audio CD.

As well as saving DDP images you can also burn them from within WaveLab. To burn a DDP image simply select **Utilities** > **Burn Audio CD form DDP Image...**. For more information see Burn Audio CD from DDP Image.

CD burning preferences

Some preferences for CD burning and DDP image creation are set in Global Preferences. Select **Preferences...** > **Global preferences** > **CD Burning tab** and use "What's this?" to get more detailed information on the options available.

Related topics

DDP files

Audio CD Report

Write Audio CD

CD Wizard

Data CD/DVD

CD Text Editor

CD

4.4 Markers

Markers are visible indicators that can be used to save and name specific time positions in an audio or audio montage file. In WaveLab, markers can be used for adding comments, editing, synchronizing, defining loops, and defining CD/DVD-A tracks. Markers can be used in pairs as well as individually to mark regions or places for future reference. Once you have defined some markers in your file you can quickly move the playhead to a marker during editing, or select the audio between two markers. Markers can also be used to identify an audio cue, or the beginning or end of a filter or effect.

About marker types

There are many types of markers, all of which, in addition to visually marking locations in a file, can be used to assist editing and playback. Some like the Loop and CD Track markers have a unique function. For information on each type and how to use them see Marker types.

Adding markers

The most common way to create markers is from the Marker window. If this is not visible make sure that "Markers" is ticked in the menu **Workspace** > **Specific tool windows** > **Markers**. Add markers by:

Placing a single marker

Play back the wave. When the cursor is at the position where you want a marker:

- Press [Insert] on the PC or 'M' on the Mac (there is no Insert key on the Mac) to insert a generic marker.
- Click a marker button on the Marker window toolbar. Tooltips for each type of marker are shown when hovering the cursor over its button.
- Select a marker of a specific type from the Marker Window Insert menu.

Placing a pair of markers

Some markers come in pairs. To create a pair of markers make a selection around the audio region where you wish to define a start and end:

- Click a marker pair button on the Marker window toolbar. The types of marker are shown when hovering the cursor over each button.
- Select "Create region type from selection" from the Marker Window Insert menu, choosing the type of marker pair you wish to use. You need to specify a region for these marker types to become available.

Some types of markers can be added during recording via the recording dialog. You can also create a marker by dragging a selection of text from a text application to a Time Ruler.

Moving markers

Once placed, markers can be dragged into a new position in the Waveform windows. If you hover the mouse cursor over a marker, the exact time position is displayed. Dragging the marker with the left mouse button held down causes the time display to be updated as you drag. You can also move a marker by editing its time entry in the Marker window.

Removing markers

In the Waveform window, click on the head of the marker you want to delete. Drag the marker vertically out of the wave window - a delete symbol is shown as confirmation. You can also use the "Delete selected markers" function from the Marker Window Functions menu.

Using markers for editing and playback

Once you have defined one or more markers in an audio montage or Audio File you can use them to:

- **Control playback** - you can access various commands to start at, stop at, loop between and skip markers during playback via the Transport menu. These can also be accessed from the Transport toolbar.

- **Define a selection** - you can use markers to help you make selections in a waveform. These can be accessed by various selection commands in the Edit > Select menu.You can also access it from the Audio Range dialog.

About importing and saving loop markers

There is an important point to consider when importing files with loops into WaveLab. Both Wave and AIFF file formats can store limited information about markers (e.g. loop points) in the Audio File, but WaveLab can store much more information (such as extended marker types and comments) in external ".mrk" files that are independent of file formats. Because loop markers may be stored in two ways, there is a potential risk for conflict:

- When you import a file that has never been used in WaveLab before, and which contains loops, the loops are "imported" and displayed as loop markers.

- When you then save the file in AIFF or Wave format, the loop points are stored, both as part of the actual file (so that other applications can read them) and in the ".mrk" file (for WaveLab).

- When you open a file that has been used in WaveLab before, loop tags that were originally in the file, (before you used it in WaveLab) are ignored. Only the loop information in the ".mrk" file is used.

Using markers in the Audio Montage Workspace

There are some additional marker functions that are specific to the Audio Montage Workspace which allow you to bind selected markers to individual Clips. These functions are useful as they allow you to freely move Clips about and perform editing functions without fear of loosing the exact position of a marker. These additional marker functions are located in the Marker Window when editing in the Audio Montage Workspace:

- **Bind selected marker to start of focused clip**
- **Bind selected marker to end of focused clip**
- **Detach selected marker from its associated clip**
- **Bind selected marker to audio samples of focused clip** -

Related topics

Marker types

Marker Window

4.4.1 Marker types

Markers have a variety of uses in editing and playback, such as indicating cue points, absolute time locations, highlighting "problem" sections and for visually separating tracks.WaveLab provides a set of specialized markers for creating CD/DVD-A tracks, muting or skipping sections and for use with the Error Correction tool.

Basic markers

- **Generic markers** - these are mainly used for locating certain important positions and for editing (for selecting all the audio between two points, for example). Generic markers can be created directly during recording.
- **Temporary markers** - these markers can be used for any purpose, but they only persist while a file is open, so are "self-clearing" when the file is closed.

CD/DVD-A authoring markers

- **CD Track start and end markers** - these denote where a CD track begins and ends. They also serve for DVD-A disks. CD/DVD markers must be used as a pair.
- **CD Track splice markers** - a CD track splice is used when a CD track starts just where another one ends. They also serve for DVD-A disks.
- **CD Track index markers** - these are used to create "index" points in CD tracks. They also serve for DVD-A disks.

Editing/Navigation markers

- **Region start and end markers** - these markers can be created directly during recording; they can be used to define start and end points for generic regions. Editing/Navigation markers must be used as a pair.
- **Playback start markers** - This marker is used to program a playback start point.
- **Exclusion start and end markers** - these let you temporarily silence a certain section. Sections between Exclusion regions are skipped if this option is activated on the Transport menu (**Skip options** > **Skip Exclusion regions**). In addition, Exclusion regions can be skipped during file rendering if this option is activated in the Render dialog. Exclusion markers must be used as a pair.

Loop markers

- **Loop start and end markers** - These are used for defining loop points and are required to access loop editing functions in the Process menu of the Audio File Workspace: they are directly connected to the playback Loop mode. These markers are often useful during editing and for creating loops before transferring a sound to a sampler. Loop markers must be used as a pair.

Error correction markers

These two marker pairs may be placed manually, but their main use is with the Error Detection and Correction tool - see Error detection and correction. You can access this tool by choosing **Workspace** > **Specific tool windows** > **Error Correction**.

- **Error start and end markers** - these markers are used to highlight errors such as clicks. They can also be saved in an Audio File Workspace file by choosing the corresponding preference in Audio file editing prefences.

- **Correction start and end markers** - these markers are used to highlight corrections done on regions previously marked as errors. They can also be saved in an Audio File Workspace file by choosing the corresponding preference in Audio file editing preferences.

Related topics

Markers

Marker Window

Error detection and correction

4.5 Metering

WaveLab contains a variety of audio meters to help in monitoring and analyzing audio.Meters can be used to monitor audio during playback, rendering, recording and also to analyze a specific selection of audio. There are seven different audio meters in WaveLab, each with its own separate window. The meters are accessed via the Meters menu, the Shared Tool windows menu, or from the Meters control bar.

Monitor Modes

You can choose the audio source and the mode for displaying information in the meters. You can access the following monitoring functions from the Meters menu or via the Meters command bar:

- **Monitor Playback** - the meters display the audio output of the Master Section **after the dithering section, unlike the Master Section's own meter**.

- **Monitor Audio Input** - the meters display the audio input you have chosen in the Audio Streaming Settings. This is useful for monitoring audio while recording.

- **Monitor File Rendering** - the meters display what is being written to disk during file rendering, taking into account Master Section settings, with average, minimum and maximum peak values computed. After rendering, the meter(s) freeze, and remain frozen until you refresh or change the monitor mode.

- **Monitor Edit cursor position** - the meters display static information about the audio beneath the edit cursor. Note that the Master Section settings are not taken into account in this mode.

- **Analyze audio selection** - this allows you to make a selection and have the meters display the average values over the selected range as a static display. The Master Section settings are not taken into account in this mode. When you change the selection, you need to update the meter displays by selecting "Refresh selection analysis" from the Meters menu (or by clicking the Refresh button on the Meters control bar).

If you have one of the continuous display modes chosen, you can also select to freeze the meters at any time using the "Freeze meters" command.

Using meter windows

There can only be one instance of each Audio Meter. For example, if you dock one meter in a Workspace, it will be automatically removed from the place where it was previously located.

Audio Meters can appear in the Audio File, Audio Montage Workspaces as well as the Control Window. They can be used:

- as a docked window in a Workspace
- as a tabbed window in the control window
- as an independent floating window. In this mode, it can be useful to make the window "frameless" from **Window** > **Hide frame**, to save screen space. In this case, the whole menu is accessed by right-clicking.

The axis of most Audio Meters may be rotated, to view the graphics horizontally or vertically.

Some meters can also have their visual style and display parameters customized via their corresponding settings dialog. This can be accessed through the **Functions** > **Settings...** menu in each meter window, or by using the icon.

Related topics

VU Meter

Spectroscope

Oscilloscope

Bit Meter

Phasescope

Spectrometer

Wavescope

4.5.1 VU Meter

The vu-meter is used to display the peak and average loudness/decibel level of your Audio File. It also displays the pan, or the balance between the left and right channels in a stereo file.

The upper part of the level/pan meter window shows the peak level and average loudness, in the following way:

- The Peak Level meters display the peak levels of each channel, graphically and numerically. By default, the meter segments and numerical peak values are displayed in green for low levels, yellow for levels between -6dB and -2dB, and red for levels above -2dB. You can change both the colors and the range boundaries if you wish via the Level/Pan Meter Settings dialog. This can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

- The VU (Volume Unit) meters measure the average loudness (RMS) of each channel. These meters have a built-in inertia, evening out loudness variations over a user-defined time span. If you are monitoring playback or audio input, you will also note two vertical lines following each VU meter bar, seemingly "trying to reach" the current RMS value. These lines indicate the average of the most recent minimum RMS values (left line) and the average of the most recent maximum RMS values (right line). To the left, the difference between the minimum and maximum average values is displayed (the level value in brackets) - this gives you an overview of the dynamic range of the audio material.

- If you are monitoring real-time audio (playback or input), the maximum peak and loudness values are displayed numerically to the right of the meter bars. Numbers in brackets to the right of the Maximum Peak values indicate the number of successive clips (0dB signal peaks). Recording levels should be set so that they only rarely clip. If the master level is set too high, the sound quality and frequency response will be compromised at high recording levels, with unwanted clipping effects. If the level is set too low, noise levels may be high relative to the main sound being recorded.

Pan meters

The lower part of the window shows the pan (the difference in level between the left and right channel, only applicable when monitoring stereo audio):

- The upper pan meters show the peak level difference between the channels, graphically and numerically. Note that the pan meters are "two-sided"; the level bars can go to the left or right, indicating which channel is the loudest. The two sides are shown in different colors (which can be changed via the Settings dialog as explained previously).

- The lower pan meters show the average difference in loudness between the channels, in a similar way. This gives you a visual indication of whether a stereo recording is properly centered, for example.

- If you are monitoring real-time audio (playback or input), the maximum balance difference value (peak and loudness) for each channel is displayed numerically to the left and right of the meter bars. The VU meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces.It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

4.5.2 Spectroscope

The Spectroscope shows a continuous graphical representation of the frequency spectrum, analyzed into 60 separate frequency bands, represented as vertical bars. Peak levels are

shown as a short horizontal lines above the corresponding band, indicating recent peak/-maximum values. The Spectroscope offers a quick spectrum overview. For a more detailed analysis of the audio spectrum, use the Spectrometer. The Spectroscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

Spectrometer

4.5.3 Oscilloscope

The Oscilloscope offers a highly magnified view of the waveform around the playback cursor position.

If you are analyzing a stereo file, the Oscilloscope normally shows the separate levels of the two channels. However, if you activate the option "Show Sum and Subtraction" on the Functions menu (or click the +/- icon), the upper half of the Oscilloscope shows the sum of the two channels and the lower half shows the subtraction.

Making settings

By opening the Settings dialog you can adjust the display colors, and choose to activate Auto-zoom or not. With Auto-zoom activated, the display is optimized so that the highest level reaches the top of the display at all times. The Settings dialog can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

The Oscilloscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

4.5.4 Bit Meter

The Bit Meter shows the resolution or the number of bits used in the digital audio being monitored. Normally the maximum number of bits in an Audio File is the same as the resolution of the Audio File (e.g. a 16 bit Audio File shows that up to 16 bits are used), but sometimes this is not the case.

As soon as you perform any kind of real-time processing on an Audio File, the audio data is treated at a much higher resolution (32 bit floating point), to allow for pristine audio quality. Examples of such processing include level adjustments, effects, the mixing of two or more files, etc. In fact, the only time when a 16 bit file is played back at 16 bit resolution is if you play it without any fades or effects, and with the Master Faders set to 0.00 (no level

adjustment). You can try this out for yourself by playing back a 16 bit Audio File and viewing the Bit Meter - as soon as you adjust the Master Faders, note that 24 bits are used and the "inter" indicator is lit (more on this below).

How to read the Bit Meter

- The innermost meters (closest to the bit scale) show the number of bits in use. You can adjust this display in the Settings dialog.
- The outer meters are "history" meters, showing how many bits were recently in use. You can adjust the hold time in the Settings dialog.
- The "over" segment indicates clipping, similar to a clip indicator.
- If the "below" segment is lit, there are more than 24 bits. The bit meter will show the 24 higher bits, and the "below" segment indicates the existence of extra, lower bits. Note that audio is always processed with more than 24 bits internally.
- If the "inter" segment is lit, this indicates that the audio data cannot really be expressed on a regular 24 bit scale (there are floating point values "in between" bits, thus the label "inter"). This is typically the case if you apply effects; thus the "inter" segment allows you to differentiate processed from non-processed 24 bit PCM files.

When to use the Bit Meter

- To check whether dithering is necessary or not. As a rule, if you're playing back or mixing down to 16 bits, and the Bit Meter shows that more than 16 bits are used, you should apply dithering.
- To see the "actual" resolution of an Audio File. For example, even though a file is in 24 bit format, only 16 bits may be used. Or, a 32 bit file may only use 24 bits (in this case, the "below" segment would not be lit). For this purpose, the Bit Meter is best used in "Analyze selection" mode.
- To see whether a "zeroed" plug-in still affects your signal, or whether a plug-in uses 16 bit internal processing, or more generally, to detect any signal modifier stage in the audio chain between reading and playing.

To adjust the Bit Meter settings, select "Settings..." from the Options pop-up menu or by using the icon.

The Bit meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Dithering

4.5.5 Phasescope

The Phasescope indicates the phase and amplitude relationship between two stereo channels. It is only really relevant when monitoring stereo audio material.

Reading the Phasescope

It can be interpreted as follows:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.
- A random but roughly elliptical shape indicates a well balanced stereo signal. If the shape "leans" to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the Phasescope will show a straight line, angled 45 degrees to the other side).
- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90 degrees on the other.
- Generally, the more you can see a "thread" shape, the more bass there is in the signal; the more "spray-like" the display, the more high frequencies are in the signal.

Phase Correlation meter

At the bottom of the display, you will find a Phase Correlation meter, which shows the same information but in a different way:

- The green line shows the current phase correlation, while the two red lines show the recent peak minimum and maximum values, respectively (these are the default colors, which you can change).
- With a mono signal, the meter would show +1, indicating that both channels are perfectly in phase.
- Similarly, -1 indicates that the two channels are the same, but one is inverted.
- Generally, for a "good" mix, the meter should show a value between 0 and +1.

Unlike the main Phasescope, the Phase Correlation meter is also available in "Analyze audio selection" mode, showing an average value for the selected range.

Changing settings

You can define display colors, peak hold time, and the resolution or number of samples to display from the **Functions** > **Settings...** menu.

The Phasescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

4.5.6 Spectrometer

The Spectrometer uses FFT (Fast Fourier Transform) techniques to display a continuous frequency graph, providing a very precise and detailed real-time frequency analysis.

- The current frequency spectrum is shown as a linear graph.
- Spectrum "peaks" are shown as a short horizontal lines, indicating recent peak/maximum values.

Snapshots

By using the "Add snapshot" and "Erase last snapshot" buttons, you can take and erase snapshots of the current spectrum. These will be superimposed over the current peak spectrum graph, in a color you can customize, until you click the icon again to take a new snapshot. One use for this is to check the effects of adding EQ, for example. Up to five snapshots can be superimposed on the display; the sixth snapshot replaces the earliest one in the display, and so on. (Don't confuse the order of snapshots with the numbered buttons - these are Spectrometer presets, see below.)

Zooming

You can adjust the frequency scale and range in the Settings dialog as described below, but it's also possible to temporarily zoom in on a frequency area of interest. This is done by clicking and dragging a rectangle in the Spectroscope display. When you release the mouse button, the display is zoomed in so that the enclosed frequency range fills the window. To return to full-scale display, select "Zoom-out fully" from the Functions menu, or double-click anywhere on the display.

Making settings

You can adjust the behavior and display of the meters as desired, and assign up to five sets of Spectrometer settings to the Preset buttons, for instant access. Open the Settings dialog by selecting "Settings" from the Functions menu or clicking the "tool" icon. Note that you can apply your settings without closing the dialog, by clicking the Apply button.

If you want to store your settings for later use (or assign them to a Preset button), select "Save as..." from the pop-up menu in the lower part of the dialog, and specify a name for the preset in the file dialog that appears. Now, you can choose to make the settings instantly available for selection in the FFT Meter window, by using the "Assign to preset button" submenu on the pop-up menu.

- When you are finished, click OK to close the dialog.

Selecting Spectrometer presets

If you have assigned your settings to the Preset buttons in the Settings dialog, you can quickly switch between different level scales and display modes, by clicking one of the Preset icons [1]-[5], or selecting the desired preset from the Options pop-up menu.

Exporting FFT data as ASCII text

When using the Spectrometer in off-line mode ("Monitor Edit cursor Position" or "Analyze audio election" mode) you can export the displayed FFT data as a text file, by selecting "Export FFT data as ASCII" from the Options pop-up menu. The resulting text file can then be imported into applications that allow graph plotting from text files (Microsoft Excel, for example).

The Spectrometer can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Spectroscope

4.5.7 Wavescope

The Wavescope meter displays a real-time waveform drawing of the audio signal being monitored.

It can be useful when recording or rendering a file if "Monitor File rendering" mode is active.

Making settings

You can adjust settings for the display via the Wavescope Settings dialog. This can be accessed via **Functions** > **Settings...** menu, or by using the icon. Here you can set various color options for the background, grid and waveform display, as well as setting the waveform rendering speed and vertical zoom. If "Clear waveform when reaching right of pane" is checked the waveform display is cleared each time the cursor reaches the right end of the display. If unchecked, the previous waveform is overwritten.

Tip: Wavescope is a useful meter for visualizing audio during recording. The Wavescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

4.6 Offline processing

Offline processes are used for a variety of editing purposes and creative effects. Offline processing refers to functions that are only performed when saving the file.

"Offline" means the process can't be monitored during playback, either because they require too much processing power or because they require an analysis pass prior to processing. Note that WaveLab only writes temporary files when applying an effect, so the original Audio File is safe. Only when saving the original is the original audio possibly modified.

These offline processes differ to realtime processes (such as listening to effects via the Master Section for example), which only temporarily render an effect while audio is played "through" them.

Applying processing

Processing can be applied to a selection or to a whole Audio File. For certain processing operations processing the entire file is necessary. This will be indicated by the effects respective dialog. If "Process whole file if there is no selection" is selected in the Audio File editing preferences editing tab, the whole file will automatically be processed if no selection exists.

To apply processing to some audio, proceed as follows:

1. Make a selection. If the file is in stereo, you can apply processing to either channel or both, by selecting one channel or both.
2. Select the desired offline process from the Process menu.
3. If a dialog appears, make the appropriate settings. If the dialog uses presets, you can load one of them to fill out the settings automatically.
4. When you have finished with the settings in the dialog, click the Apply/Paste/Process button to permanently render the effect to file.

In some cases a status bar dialog is displayed showing the progress of the operation. If you need to interrupt a long process, click the "Cancel" button on the status bar dialog.

Undo/Redo processing

Even though offline processes appear to alter the file permanently, you can always reinstate a previous version using WaveLab's "undo" feature. The number of Undo/Redo commands is limited only by your available hard drive space. Access the Undo and Redo commands via **Edit** > **Undo** or **Edit** > **Redo**.

Tips for using Offline Processing

Processing dialogs are non-modal - meaning that they stay open and on screen even after processing has been applied. This means that you can still playback audio and access undo/redo commands to audition the results of your effect and make any further adjustments to any settings. You can also use key commands and access any other menu commands while leaving the offline process dialog open. Remember that using and saving presets when you have settings you are happy with can also save you time.

WaveLab contains a powerful array of offline processing tools including the following:

Change Level

Level envelope

Normalize Level

Silence Generator

Loop Tone Uniformizer

Pitch correction

Time stretching

Loop Tweaker

Effect morphing

Loudness distribution

Loudness normalizer

Pan Normalizer

Pitch bend

Pitch quantize

Related topics

Presets

Rendering

4.6.1 Change Level

This dialog allows you to apply a gain to change the level of an Audio File.

You can also use "Find current peak level" to obtain a report on the peak level of the current audio selection (or the peak level in the whole file, if "Process whole file if there is no selection" is selected in the preferences). This can be useful if you wish to calculate how much you can increase the overall gain of a file without introducing clipping (exceeding 0dB), for example.

To change the level, enter the desired gain and click Apply.

You can access this dialog by choosing **Process** > **Change level...** in the Audio File Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Value editing

4.6.2 Level envelope

This dialog allows you to create a volume envelope which can be applied to a selected range or a whole Audio File. This could be useful if you need to even-out loud and quiet parts, for example.

You can edit the envelope by:

- Double clicking on the line to add a point: double click on a point to remove it.
- Selecting multiple points (using **[Shift]** click) allows you to move the selection as a group.

You can also use the controls to delete or reset points in the envelope. Use the envelope smoothing button to toggle the envelope points between a straight (polygonal) line or a curved path.

Click Apply to apply the envelope to the audio selection or file.

You can access this dialog in the Audio File Workspace via **Process** > **Level envelope...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Adjusting Envelopes

4.6.3 Normalize Level

This dialog allows you to change the peak level of your Audio File.

Enter the desired peak level (in dB) that you wish the audio selection to have. You can also use "Find current peak level" to obtain a report on the peak level of the current audio selection (or the peak level in the whole file, if "Process whole file if there is no selection" is selected in the preferences). You can choose to apply the same gain to both stereo channels (Stereo Link) or mix down to a mono file (Mix to Mono) with the assurance that no clipping will occur when both channels are mixed together.

To normalize the selected audio, enter the desired peak level and click Apply.

You can access this dialog by choosing **Process** > **Level Normalizer...** in the Audio File Workspace.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Value editing

4.6.4 Silence Generator

This dialog allows you to generate "silence", either by replacing an audio selection, or by specifying a duration.

You can also specify a cross fade at each end of the silent region, for a smoother transition. Radio buttons determine where the silent period is placed in the audio.

In WaveLab you can also specify a file to be inserted instead of "true" silence. This can be useful for example, if you wish to keep the ambience, or background noise of a particular environment in a recording.

You can access this dialog in the Audio File Workspace via **Edit** > **Silence (advanced)...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Silence

4.6.5 Loop Tone Uniformizer

This dialog allows you to create sounds that will loop, from audio that is seemingly "un-loopable". These are normally sounds that constantly decay in level or continuously change in timbre. The Loop Tone Uniformizer applies processing to the sound that evens out changes in level and timbral characteristics in order for a sound to loop properly. For example, this could be useful for creating looped samples for a softsynth or hardware sampler.

To use the Loop Tone Uniformizer you need to have a loop defined using a pair of loop markers . The original length of the loop is not changed.

Uniformizers

This tab allows you to specify the method/s used to even out the sound you wish to loop. Choose between "Slice mixing" and/or "Chorus smoothing". For a full explanation of how each of these methods operate use the "What's this?" help function.

For slice mixing, you may need to experiment to see how many slices are needed. Generally the more slices you use, the more natural the sound will be (up to a certain point).

Pre Cross fade

The Loop Tone Uniformizer also includes the option to cross fade the end of the loop with the start of the newly processed section so that transition into the newly looped section is smoother during playback. Use the envelope drag points or value sliders to adjust the cross fade.

A note about Post Cross fading

If you have used the Loop Tone Uniformizer within a region of an Audio File, the transition from the end of the loop into the original file in many cases won't be very natural. This can be fixed as follows:

1. Close the Loop Tone Uniformizer and open the Loop Tweaker.
2. Click on the Cross fade tab and deactivate the Cross fade option (the check box).
3. Click on the Post Cross fade tab and make sure Cross fade is activated.

4. Set up the post cross fade parameters and click Apply.

It can also be a useful tip to lock the loop markers after you use this command on a selection within an Audio File. Once the loop is processed in place, moving the loop markers will upset playback of the loop.

You can access this dialog in the Audio File Workspace via **Process** > **Loop Tone Uniformizer...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Markers

Adjusting Envelopes

4.6.6 Pitch correction

This dialog allows you to detect and change the pitch of a sound without affecting its length.

- **Amount of Shift** - use these controls to find out the current pitch of the audio and if desired, calculate the required shift to match a specific pitch.

- **Method** - use these controls to adjust the method and quality you wish to use when processing the pitch change. You can also choose how the length of the sound is affected by the operation. By default this process will not change the length of the sound.

This tool could be useful for fixing an off-key vocal note in a live recording, or tuning the pitch of a kick drum sample to fit a particular song, for example.

You can access this dialog from the Audio File Workspace via **Process** > **Pitch Correction...**.

Pitch correction is also available as a batch processing plug-in in the Batch Processor Workspace. In addition, Pitch correction for a Clip is available from the Focused Clip window (via **Edit** > **Pitch shift...**).

Note that for Batch processing and Clip processes, the envelope option is not available.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

4.6.7 Time stretching

This dialog allows you to change the duration of an audio selection (usually without changing its pitch).

You can view exact time information on the original audio selection you wish to stretch and choose the amount you wish to "stretch" it in seconds, beats per minute or as a percentage. You can also choose which method WaveLab uses to stretch the audio and the quality/speed of the processing.

If you select "Use modulation envelope" you can vary the time stretch over time using an envelope editor that becomes available. When choosing a method, it is useful to experiment with the different algorithms available to find which one suits your source material the best. Use "What's this?" to find out more information on each method and determine which one will produce the best time stretch without introducing unwanted artifacts.

This function is useful if you want to speed up or slow down a recording to match other material, such as matching the tempo of two drum beats, or fitting an audio track to a section of video, for example.

Time stretching is also available as a batch processing plug-in in the Batch Processor Workspace. In addition, Time stretching for a Clip is available from the Focused Clip window.

Note that for batch processing and Clip processing, the modulation envelope option is not available. You can access this dialog from the Audio File Workspace by choosing **Process > Time stretching...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

4.6.8 Loop Tweaker

This dialog allows you to refine a region of audio to loop seamlessly.

The Loop Tweaker can just be used to "tweak" an existing loop selection so that it loops perfectly or you can also use it to create a loop from material which does not naturally repeat. To use the Loop Tweaker you must first have a loop defined using a pair of loop markers .

The Loop Tweaker consists of the following tabs:

Loop points adjustment

This is a visualization of the beginning and end of the waveform between the loop markers. You use this dialog to manually refine a loop selection by dragging on the waveform left/right or using the automatic search buttons to find the nearest "good" loop point. The aim is to align the waveforms so they meet in the middle at a zero-crossing point with the waveforms matching as closely as possible, being "in phase". When you adjust your loop start and end points within the dialog the start and end loop markers in the main waveform window will adjust accordingly. Note that this movement may or may not be visible depending on the amount you move the markers and the zoom factor you have selected.

It is helpful to have the transport looped during playback so you can hear the difference when you adjust the loop markers within the dialog. Use the Display options to adjust how the Loop Tweaker waveform is visualized. You can also use the temporary memory buttons to capture a number of different loop "candidates" and audition them one by one. Note that if you are not using a cross fade or post cross fade you do not need to click Apply when just tweaking

loop points. You can also leave this dialog window open and manually adjust the position of the markers in the main waveform windows if you need to make large adjustments.

Cross fade

This tab allows you to apply a cross fade at the end of a loop by mixing a copy of the beginning of the loop with it. This can be useful to smooth the transition between the end of a loop and its beginning - especially when using material that doesn't naturally loop. Use the envelope drag points or value sliders to adjust the cross fade envelope. Click Apply to process the cross fade.

Post cross fade

This tab allows you to cross fade the loop back into the audio after the end of the loop. This is done by mixing a copy of the loop back into the audio. Use the envelope drag points or value sliders to adjust the cross fade envelope. Click Apply to process the post cross fade.

Using the "Copy" button you can insert multiple copies of the loop into the active Audio File, creating a smooth audio sequence, since copies will loop without an audible transition.

You can access this dialog by choosing **Process** > **Loop Tweaker...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Markers

Adjusting Envelopes

Zero crossing

4.6.9 Effect morphing

This dialog within the Audio File Workspace allows you to gradually mix two audio ranges that have different effects/processing applied to them.

Effect morphing allows you to smoothly "morph" or blend from one effect to another, or from an unprocessed audio segment to a processed audio segment. Effect Morphing always involves two audio ranges. Normally, these are two versions of the same audio range, of which one is unprocessed and the other processed.

Effect morphing works by using WaveLab's Undo buffer to mix a copy of the processed audio with the unprocessed version. You can also use any other bit of audio from the application's clipboard that has the same length as the one you wish to morph with.

To set up a basic effect morph, proceed as follows:

1. Start by making a selection range over the time you wish the effect morphing to take place.
2. Now process this range using any Master Section effect(s), or use some other off-line processing. You cannot use processing/effects that alter the length of the selection, e.g. time stretching.
3. Open the Effect Morphing dialog by choosing **Edit**>**Effect Morphing...**.

4. Adjust your envelope points over time between 0% and 100%. This will determine the level and direction of the morph. For example, starting at 100% and ending at 0% will fade out the effect you just applied.
5. Choose "Unprocessed selection" for this example. As mentioned previously you can also use any audio of the same duration from the clipboard.
6. Click Apply to apply the Effect morphing.

You can access this dialog in the Audio File Workspace via **Edit** > **Effect morphing...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

4.6.10 Loudness distribution

This dialog allows you to measure the most frequent loudness values found in an Audio File (this is not the same as average loudness). The graphic peaks found by the analysis represent these values.

The tool gives an answer to the question: "How often does a given loudness (vertical scale, in dB) appear in the whole file?". The percentage figure is relative to other peaks. Some simple examples:

- 1. If, for instance, you have a sine of 0 dB / 2 second followed by a sine of -6 dB / 2 seconds. This means the global audio material has as much 0 dB material as - 6dB material: you will see 2 peaks (0 / -6 dB), each at 100%.
- 2. If, for instance, uou have a sine of 0 dB / 1 second followed by a sine of -6 dB / 3 seconds. This means there are 3 times more -6 dB material than 0 dB material. You will see a peak of about 33% at 0 dB, and another peak of 100% at -6 dB.

This analysis is useful for learning about how the loudness of your music is distributed.

You can access this dialog in the Wave File Workspace via **Analysis** > **Loudness distribution...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

4.6.11 Loudness normalizer

This dialog allows you to adjust the loudness of a file. Because it relates to loudness as opposed to maximum peak level, this tool works quite differently from the Level Normalizer. A typical application is to specify a loudness, for example -12dB, and the Loudness Normalizer processes the audio to match this level of loudness.

Like Gain Change, increasing the loudness to a certain value can introduce clipping. To avoid this occurring, a peak limiter (the Peak Master plug-in) can be used as part of the process.

The Loudness Normalizer raises the loudness and limits peaks in the signal at the same time (if needed), to achieve the target loudness.

In the case of stereo files, both channels are processed independently. This process happens over several stages; analysis precedes the final rendering. The Loudness Normalizer dialog gives access to useful statistics about the file and the Loudness distribution tool, and you can choose to remove any DC offset in the file.

You can access this dialog from the Audio File Workspace via **Process** > **Loudness normalizer...**.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Loudness Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

DC Offset

Loudness distribution

4.6.12 Pan Normalizer

This dialog allows you to ensure that both channels of a stereo file have the same level or loudness. Since loudnesss is more important than peak volume in the perception of sound, this is a powerful tool to get the best possible stereo balance.

This process uses two passes to first analyze the audio and then render any level changes required.

You must have a stereo selection in a stereo file to apply this process.

You can access this dialog from the Audio File Workspace via **Process** > **Pan Normalizer...**.

This function is also available as a batch processing plug-in. You can access it from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **Pan Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

4.6.13 Pitch bend

This dialog allows you to change the pitch of a sound over time. Note that changing the pitch using Pitch bend will affect its duration unless length preservation is activated.

Using an envelope you can 'draw' the curve you wish the pitch to follow. The pitch offset is displayed along the vertical ruler of the envelope and the range the envelope effects can be adjusted using the spin control. Positive pitch values produce sounds with a higher pitch and of shorter length, negative values produce sounds with a lower pitch and of longer length.

If length preservation is activated, you can choose the algorithm that is used to perform the pitch bend operation. Depending on the type of audio material you are processing, choose the appropriate mode. For more information on the different modes, use the "What's This?" tool. You can also adjust the quality used when processing the pitch bend. The quality setting and the selected mode will affect the processing time for this effect.

This function could be used for creating the classic 'tape stop' effect, or for blending the tempo/pitch of one track into another, for example.

You can access this dialog from the Audio File Workspace via **Process** > **Pitch bend...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

4.6.14 Pitch quantize

This dialog allows you to correct the pitch of an Audio File.

The musical notes in the audio selection are quantized to the nearest semitone. Pitch quantize works best on recordings that have a single fundamental frequency, such as voice or single instruments. It can be useful for 're-tuning' the pitch of an imperfect recorded vocal performance, for example.

The dialog allows you to choose the reference frequency (normally 440 Hz) and the time taken for the pitch correction to reach the nearest semitone. An explanation of when and how to adjust the Slur time is given using the 'What's this?' tooltips.

You can also choose to preserve the formants of the original material. Choosing this option can sometimes help to produce a more realistic result when correcting the pitch of vocals or acoustic instrument sounds.

You can access this dialog from the Audio File Workspace via **Process** > **Pitch quantizing...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

Formant

4.6.15 Presets

WaveLab's powerful preset system allows you to create your own presets to store commonly used settings. They are available every time you open WaveLab. Presets can be saved for many parameters such as plug-in settings, Audio File Formats or File Attributes, for example. You can access the Presets pop-up menu wherever you see one of these icons: or . Presets are saved as local files on your hard drive, allowing you to conveniently back-up or

copy them for use on another computer.They are also useful when used in conjunction with batch conversions and Scripting.

Working with Presets

When working with Presets you save/edit and load them from their own pop-up menu. Depending on the context, the available functions sometimes vary but generally you can:

- **Save/Save As...** - creates a new Preset file by saving the currently selected settings to your hard disk.
- **Explore presets...** - opens the folder where any existing Presets are stored. This allows you to delete, duplicate and organize your settings into sub-folders using the flexibility of the operating system's file browser.
- **Restore factory settings** - restores the current settings to the factory default.
- **Preset list** - select a Preset from the list of those currently available.
- **Store/restore temporary presets** - when available, use the temporary slots provided to save your settings for the duration of the session. This is useful if you want to quickly test and compare different settings. Select the option "Store temporarily" and from the submenu select the number under which you want to save (#1 - #5). To quickly load the saved settings again, select the corresponding number from the "Restore" submenu.
- **Define shortcut for the current setting** - when available, this allows you to assign a shortcut to the current preset using either a key shortcut, keyword or via a MIDI trigger. For example, if you have a preset to normalize audio at -0.1 dB and you assign it a key shortcut then everytime you execute the shortcut, the preset is applied to the audio selection, without the need to open the dialog.

VST-2 Presets

VST plug-ins have their own preset handling. When you click the preset button for this type of effect the following options are available:

- **Load/Save Bank** - this allows you to load and save complete sets of presets. The file format is compatible with Cubase.
- **Load/Save Default Bank** - this loads the default bank of presets, or saves the current set of presets as the default bank.
- **Load/Save Effect** - this loads or saves one preset at a time. The file format is again compatible with Cubase.
- **Edit name of current program** - allows you to define or edit the name for the currently selected preset.
- **Preset list** - select a Preset from the list of those currently available.

VST-3 plugins have a simplified list of options.

About Preset files

Each preset is stored as an individual file inside WaveLab's main presets folder. This makes it easy to back them up or copy them to another computer. You can change the location of this main Presets folder via **Global Preferences** > **Setting location**. You have the choice of the following locations:

- a standardized location that can be accessed by all users of the computer. That is, all users will share the same settings.
- a standardized location accessed by the current user (the default). That is, each user has his/her own private settings.
- any specific location of your choice. This allows you to customize it for your own requirements.
- a location relative to the application. This allows you to use the application from an USB dongle.

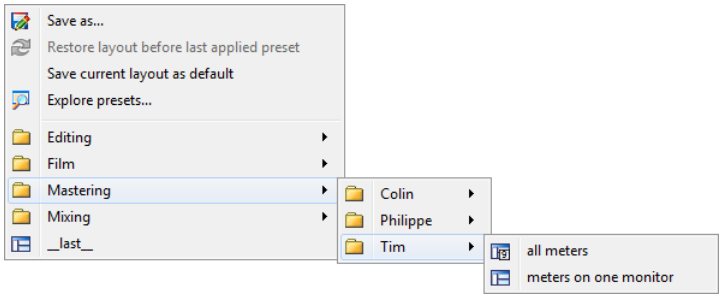
Default locations - By default the main Presets folder is found at:

- **Windows 7** - *C:\Users\[User Name]\AppData\Roaming\Steinberg\WaveLab 7\Presets*
- **Windows XP** - *C:\Documents and Settings\[User Name]\Application Data\Steinberg\WaveLab 7\Presets*
- **Mac OS X** - *root/[User Name]/Library/Prefences/WaveLab 7/Presets/*

You can also find this folder by using the "Open active settings folder" link in the preferences pane. This opens your file browser to wherever the main Presets folder is currently located.

Inside this Presets folder each type of preset file has its own folder. For example, all the presets for the Normalizer are stored within *C:\Users\[User Name]\AppData\Roaming\Steinberg\WaveLab 7\Presets\Normalizer* on Windows 7. When you open the Normalizer dialog and choose a preset via the pop-up menu, the files in this directory will appear as available presets. The reason each preset type has its own dedicated folder is so that WaveLab can locate them automatically when you open the preset pop-up menu.

You can organize your presets into sub-folders so they appear as sub-menus. This can be useful if you wish to organize a lot of presets into your own categories. You can continue to nest folders inside one another to build a standard tree structured menu system. Below is an example of how this looks for a set of window layout presets:



Related topics

Scripting

4.6.16 Rendering

To Render effects into a temporary or final file, you use the Render function (note that the "Save" function from the File menu does not Render!). The Render function is part of the Master Section and in other applications is sometimes referred to as "bouncing down" or "mixing down".

Rendering a file "prints" all audible settings and effects that the audio is being routed through in the Master section, to the resultant Audio File. This includes any plug-ins, the level of the faders and any dithering you have set in the Master Section. In WaveLab you normally use the Render function to:

- Mix down a file in a Wave window to a new Audio File, complete with Master Section effects, dithering and other settings.
- Mix down a complete Audio Montage to an Audio File.
- "Apply" all settings in the Master Section to a file in a Wave window.

When you render a file you can choose the format of the new Audio File and whether you wish to create a new file or render the file in place. When the file is rendering you will see a progress bar in the status bar.In WaveLab you can also view this Progress bar in the Background tasks window (a shared tool window that can be set to float independently).This allows you to monitor the progress of rendering, pause and if necessary cancel the rendering process.

Rendering tips

When you are new to the rendering process in WaveLab there are a few things to note:

- when rendering a file with the default option "Bypass Master Section on resulting Audio File" selected in the Render dialog, the Master Section will be bypassed completely when the new file has finished rendering. If this option is **not** selected you will hear the Master Section effects being played back on top of the newly rendered effects. You can check if the Master Section has been bypassed by checking the status of the "Play through Master Section" button, which is located at the bottom right hand corner of the status bar.
- the Master Section's "Bypass" switch affects playback only. So even if you can't hear effects when you are playing back your audio, any applied effects are still rendered.
- the "Effect on/off" switch for each individual plug-in affects both playback and Rendering.
- if you are working with an Audio File/montage that contains lots of effects and/or tracks and you experience playback problems, it can be useful to mix down a copy of your

montage/audio file to free up some processing power. You can then continue editing the resultant file with the effects rendered. Remember to save a copy of your session before doing this so you can go back and edit any levels or effects if you need to.

For more information about choosing settings when rendering see the Render Wave window and Render Montage window. For more information on applying effects see the Master Section.

Related topics

Render Wave window

Render Montage window

Master Section

Audio Plug-ins

4.7 Presets

WaveLab's powerful preset system allows you to create your own presets to store commonly used settings. They are available every time you open WaveLab. Presets can be saved for many parameters such as plug-in settings, Audio File Formats or File Attributes, for example. You can access the Presets pop-up menu wherever you see one of these icons: or . Presets are saved as local files on your hard drive, allowing you to conveniently back-up or copy them for use on another computer.They are also useful when used in conjunction with batch conversions and Scripting.

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- **Preset list** - select a Preset from the list of those currently available.
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- **Define shortcut for the current setting** - when available, this allows you to assign a shortcut to the current preset using either a key shortcut, keyword or via a MIDI trigger. For example, if you have a preset to normalize audio at -0.1 dB and you assign it a key shortcut then everytime you execute the shortcut, the preset is applied to the audio selection, without the need to open the dialog.

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VST-3 plugins have a simplified list of options.

About Preset files

Each preset is stored as an individual file inside WaveLab's main presets folder. This makes it easy to back them up or copy them to another computer. You can change the location of this main Presets folder via **Global Preferences** > **Setting location**. You have the choice of the following locations:

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- any specific location of your choice. This allows you to customize it for your own requirements.
- a location relative to the application. This allows you to use the application from an USB dongle.

Default locations - By default the main Presets folder is found at:

- **Windows 7** - *C:\Users\[User Name]\AppData\Roaming\Steinberg\WaveLab 7\Presets*

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- **Mac OS X** - *root/[User Name]/Library/Prefences/WaveLab 7/Presets/*

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Inside this Presets folder each type of preset file has its own folder. For example, all the presets for the Normalizer are stored within *C:\Users\[User Name]\AppData\Roaming\Steinberg\WaveLab 7\Presets\Normalizer* on Windows 7. When you open the Normalizer dialog and choose a preset via the pop-up menu, the files in this directory will appear as available presets. The reason each preset type has its own dedicated folder is so that WaveLab can locate them automatically when you open the preset pop-up menu.

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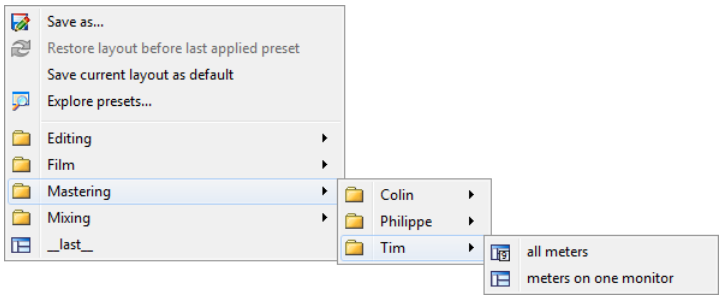


Related topics

Scripting

4.8 Rendering

To Render effects into a temporary or final file, you use the Render function (note that the "Save" function from the File menu does not Render!). The Render function is part of the Master Section and in other applications is sometimes referred to as "bouncing down" or "mixing down".

Rendering a file "prints" all audible settings and effects that the audio is being routed through in the Master section, to the resultant Audio File. This includes any plug-ins, the level of the faders and any dithering you have set in the Master Section. In WaveLab you normally use the Render function to:

- Mix down a file in a Wave window to a new Audio File, complete with Master Section effects, dithering and other settings.

- Mix down a complete Audio Montage to an Audio File.

- "Apply" all settings in the Master Section to a file in a Wave window.

When you render a file you can choose the format of the new Audio File and whether you wish to create a new file or render the file in place. When the file is rendering you will see a progress bar in the status bar.In WaveLab you can also view this Progress bar in the Background tasks window (a shared tool window that can be set to float independently).This allows you to monitor the progress of rendering, pause and if necessary cancel the rendering process.

Rendering tips

When you are new to the rendering process in WaveLab there are a few things to note:

- when rendering a file with the default option "Bypass Master Section on resulting Audio File" selected in the Render dialog, the Master Section will be bypassed completely when the new file has finished rendering. If this option is **not** selected you will hear the Master Section effects being played back on top of the newly rendered effects. You can check if the Master Section has been bypassed by checking the status of the "Play through Master Section" button, which is located at the bottom right hand corner of the status bar.

- the Master Section's "Bypass" switch affects playback only. So even if you can't hear effects when you are playing back your audio, any applied effects are still rendered.

- the "Effect on/off" switch for each individual plug-in affects both playback and Rendering.

- if you are working with an Audio File/montage that contains lots of effects and/or tracks and you experience playback problems, it can be useful to mix down a copy of your montage/audio file to free up some processing power. You can then continue editing the resultant file with the effects rendered. Remember to save a copy of your session before doing this so you can go back and edit any levels or effects if you need to.

For more information about choosing settings when rendering see the Render Wave window and Render Montage window. For more information on applying effects see the Master Section.

Related topics

Render Wave window

Render Montage window

Master Section

Audio Plug-ins

4.9 Scripting

WaveLab contains a powerful scripting language to help advanced users create their own scripts to automate tasks. Using basic scripts can be useful for automating repetitive editing

tasks such as trimming and cropping a file at specific times, for example. You can write scripts that perform other basic editing commands, apply offline processing, place markers, and display information about the active file. You can script commands to edit the active Audio File or the active Audio Montage. If you have some experience of programming with modern scripting languages you should have no problem writing utility scripts for WaveLab.

The WaveLab Scripting language is based on ECMAScript, with the addition of WaveLab specific commands. If you have had any experience with Javascript, Jscript or Actionscript the code syntax will be familiar to you as they are all based on ECMAScript too. If not, there is a large amount of teaching and reference material available online. A Javascript reference such as `www.w3schools.com` for example, or a book like `JavaScript: The Definitive Guide` are good places to start.

To begin exploring the WaveLab specific functions available see WaveLab Scripting Language. For a broader look at the complete subset of commands available see ECMAScript Reference.

Getting started

To write and execute a script:

1. Open the Script window in either the Audio File or Audio Montage Workspaces.
2. Type your script directly in the Script window or copy and paste it from an external text editor.
3. Run the script by clicking **Execute Script** or selecting it from the Functions menu.

The script will run if there are no syntax errors. Any errors will appear in a dialog box to help you debug them.

Tip: There are several free utility text editors that are context sensitive. This means that they can color and highlight parts of your code to make it more readable. If you use one of these for writing and editing your scripts, choose Javascript as the editing language and/or save the file with a .js (Javascript) extension.

Using the Log window

It is a good idea to begin by writing some simple scripts that output some text to the Log window. The Log window is part of the Control Window and can be accessed via **Utilities** > **Log**. To output a simple text message:

1. Open the Log window.
2. Open an Audio File in the Audio File Workspace.
3. Copy and paste the following script into the Script window and try executing it.

```
//output the number of samples in the active wave as text in the log window.
logWindow.printInfo("This file has " + activeWave.size() + " samples");
```

Note: Any lines of a script that begin with two forward slashes // are comments, and are ignored when the script is executed. For commenting multiple lines you can use the following syntax:

```
 /* This a long comment,
that wraps onto more than one line. */
```

Scripting window contexts

It is important to note that certain scripting commands are only available in the Audio File Workspace and others only in the Audio Montage Workspace. Others are "global", meaning you can use them in either Workspace. To begin exploring the WaveLab Scripting Language in more detail see the WaveLab Scripting Language.

A basic scripting example

Below is a basic scripting example which uses some WaveLab scripting functions to perform some simple operations on an Audio File in the Audio File Workspace. The script first displays information about the Audio File, fades in the start and fades out the end of the file, and then adds ten markers at 1 second intervals. Examine it line by line and read the comments for each operation to see how it works.

```
/* To run this script:
 - open an Audio File that is at least 10 seconds long
 - open the Log window via the Utilities menu
 - copy and paste this script into the Script window
 - choose Functions > Execute Script */

//clear the log window
logWindow.clear();

//show some information about the active wave file in the log window
logWindow.printInfo("This wave file has " + activeWave.size() + " samples");

logWindow.printInfo("Its sample rate is " + activeWave.sampleRate());

logWindow.printInfo("It has " +activeWave.numChannels() + " channels");

//Work out how long the file is in seconds and round to a whole number
var lengthSecs = activeWave.size() / activeWave.sampleRate();
logWindow.printInfo("This wave file is " + lengthSecs + " seconds long");

//Select the first 10 seconds of the file
activeWave.select(0, 10 * activeWave.sampleRate());

//Trim the file to 10 seconds
activeWave.trim();

//select the first two seconds of the file and fade it in
activeWave.select(0, 2 * activeWave.sampleRate()); //sample rate multiplied by tw
      o = 2 seconds
activeWave.fadeIn(linear);

//select the last two seconds of the file and fade it out
activeWave.select(activeWave.size() - (2 * activeWave.sampleRate()), activeWave.s
      ize());
activeWave.fadeOut(linear);

//loop through 10 times and add a marker each second
for (i = 1; i <= 10; i++)
{
  //work out next cursor time
  var nextCursorPosition = i * activeWave.sampleRate();

  //set cursor position forwards by a second
  activeWave.setCursorPosition(nextCursorPosition);

  //add a generic marker at the next cursor position and give it a name and comme
      nt
```

```
  activeWave.addMarker(generic, "Marker "+i, "A comment for marker "+i);

  //write some information about the new marker
  var cursorTimeSecs = nextCursorPosition/activeWave.sampleRate();
  logWindow.printInfo("created a new marker at " + cursorTimeSecs + " seconds");

}
```

Related topics

WaveLab Scripting Language

ECMAScript Reference

Scripting window (Audio File Workspace)

Scripting window (Audio Montage Workspace)

Control Window

4.9.1 WaveLab Scripting Language

WaveLab's scripting language varies slightly between Workspaces. This Help entry briefly introduces you to the commands that are available in each Workspace and to those that are global.

Global Commands

These are scripting commands that are available in all scripting contexts.

logWindow

Object representing the Log window, where you can output messages to. If the log window is not open, all functions are ignored (as they are not visible anyway).

Functions

printInfo(messageString) - Outputs an informal message to the Log window. The message argument must be typed as a String - i.e. between inverted commas.

Example:

```
logWindow.printInfo("start");
```

printWarning(messageString) - Outputs a warning message to the Log window.

printError(messageString) - Outputs an error message to the Log window.

clear() - Clears the log window.

Example:

```
logWindow.clear();
```

Audio File Scripting Commands

Objects

activeWave

Object representing the active Audio File.

Many functions make use of presets as an argument. For example, the normalize() function accepts a preset as an argument:

```
activeWave.normalize("myPreset");
```

The advantage is that you don't need to specify many parameters in your scripts, instead you can use the corresponding dialog to define the settings of a particular function, and then save them as a preset file. Since each type of preset is unique, you don't need to specify a full path name to the preset. Only specifying the preset name is enough, there is no need for its extension. Since presets can also be stored in a sub-folder, you can use a relative path name if necessary.

So if for example, you wanted to normalize a file using a preset you had defined in the Normalize level dialog (and saved in a sub-folder as a preset), it would look like this:

```
activeWave.normalize("mySubFolder/myPreset");
```

All audio processing functions operate on the selected audio range. If there is no selection the whole file range will be processed (if that editing option is selected in Audio File editing preferences). If the cursor or selection is in one channel only, only that channel is processed. In other words, it operates exactly the same as if you were applying a process from within a dialog.

All positions and sizes are measured in sample units. If you wish to specify a time range in another unit you will need to convert it from samples.

```
var twoSeconds = 2 * activeWave.sampleRate();
```

Functions

size() - Returns the number of samples in the Audio File.

sampleRate() - Returns the sample rate of the Audio File.

numChannels() - Returns the number of channels of the Audio File.

cursorPosition() - Returns the current cursor position (in samples).

setCursorPosition(pos) - Sets the current cursor position at a certain sample location.

selectionStart() - Returns the index of the first selected sample, or -1 if there is no selection.

selectionSize() - Returns the number of selected samples.

select(presetName) - Loads the audio range preset and applies its setting to the active Audio File.

select(start, size) - Selects a number of samples, starting from a given position.

setCursorChannel(channel) - Sets the cursor position to a new channel, Use leftCh, rightCh or allCh as arguments.

addMarker(type, name, comment) - Add a marker at the cursor position. Possible values for *type* are:

- `generic`
- `temporary`
- `cdTrackStart`
- `cdTrackEnd`
- `cdTrackFrontier`
- `cdTrackIndex`
- `loopStart`
- `loopEnd`
- `muteStart`
- `muteEnd`
- `playbackStarter`
- `regionStart`
- `regionEnd`
- `errorStart`
- `errorEnd`
- `correctionStart`
- `correctionEnd`

Example:

```
activeWave.addMarker(generic, "SomeName", "SomeComment");
```

findNextMarkerPosition(posStartSearch, type) - Searches for the next marker of *type*, from a given position. Returns the marker position, if any is found, or -1.

normalize(presetName) - Loads the normalizer preset and applies its setting to an audio range.

normalizeLoudness(presetName) - Loads the loudness normalizer preset and applies its setting to an audio range.

normalizePan(presetName) - Loads the pan normalizer preset and applies its setting to an audio range.

silence(presetName) - Loads the silence preset and applies its setting to an audio range.

timeStretch(presetName) - Loads the time stretch preset and applies its setting to an audio range.

pitchCorrection(presetName) - Loads the pitch correction preset and applies its setting to an audio range.

pitchQuantize(presetName) - Loads the pitch quantize preset and applies its setting to an audio range.

changeLevel(dbValue) - Changes the level of the selected audio range.

fadeIn(shape) and fadeOut(shape) - Applies a fade on the selected audio range. *shape* can be one of the following:

- `linear`
- `sinus`
- `squareRoot`
- `sinusoid`
- `log`
- `exp`
- `expp`

Example:

```
activeWave.fadeIn(squareRoot);
```

levelEnvelope(presetName) - Loads the envelope shape and applies its setting to an audio range.

morph(presetName) - Loads an effect morphing preset and applies it according to its settings.

invertPhase() - Inverts the phase of the samples in the audio range.

reverse() - Reverses the order of the samples in the audio range.

cut() - Cuts the selected audio range.

copy() - Copies the selected audio range.

paste() - Pastes audio from the clipboard to the current cursor position.

trim() - Trims the selected audio range.

remove() - Deletes the selected audio range.

removeSmooth() - Deletes the selected audio range and cross fades the resulting regions together.

mute() - Mutes the selected audio range.

swapChannels() - Swaps stereo channels.

undo() - Undoes the last command.

removeDcOffset() - Removes the DC offset in the audio range.

readSamples(channelIndex, from, numSamples) - Reads a number of samples from a given cursor position, on a given channel:

- Use 0 for the *left* channel
- Use 1 for the *right* channel

Returns the result in an array. **Example:**

```
buf = activeWave.readSamples(0, 20, 100); // read 100 samples on left channel, fr
      om sample index 20
for (i = 0; i < 100; i++)
{
 logWindow.printInfo(buf[i]);
}
```

Audio Montage Scripting Commands

Functions

Preliminary info: the Scripting API is an open project and will evolve according to user requests.

size() - Returns the number of samples in the Audio Montage.

sampleRate() - Returns the sample rate of the Audio Montage.

numChannels() - Returns the number of output channels of the Audio Montage.

numTracks() - Returns the number of tracks of the Audio Montage.

cursorPosition() - Returns the current cursor position (in samples).

setCursorPosition(pos) - Sets the current cursor position at a certain sample location.

selectionStart() - Returns the index of the first selected sample, or -1 if there is no selection.

selectionSize() - Returns the number of selected samples.

select(presetName) - Loads the audio range preset and applies its setting to the active Audio Montage.

select(start, size) - Selects a number of samples, starting from a given position.

setFocusedTrack(index) - Sets the focused track.

addMarker(type, name, comment) - Add a marker at the cursor position. Possible values for *type* are:

- `generic`
- `temporary`
- `cdTrackStart`
- `cdTrackEnd`
- `cdTrackFrontier`
- `cdTrackIndex`
- `loopStart`
- `loopEnd`
- `muteStart`
- `muteEnd`
- `playbackStarter`
- `regionStart`

- regionEnd
- errorStart
- errorEnd
- correctionStart
- correctionEnd

Example:

```
activeWave.addMarker(generic, "SomeName", "SomeComment");
```

findNextMarkerPosition(posStartSearch, type) - Searches for the next marker of *type*, from a given position. Returns the marker position, if any is found, or -1.

insertMonoTrack(where) - Adds a Mono Audio Track at index 'where'

insertStereoTrack(where) - Adds a Stereo Audio Track at index 'where'

insertClip(iTrack, timePosition, fileName, autoShift) - Creates a Clip from 'fileName', inserts it in track 'iTrack', on the time line at posision 'timePosition', add possibly shift other Clips to make room according to parameter 'autoShift' *autoShift* can be one of the following:

- autoShiftNo
- autoShiftTrack
- autoShiftGlobal

This function returns the ID of first created Clip, or 0.

clipWithName(name) - Returns the ID of first Clip with name 'name', or 0.

clipWithFile(name) - Returns the ID of first Clip that refers to 'fileName', or 0.

firstClip() - Returns the first Montage Clip ID, or 0, if the Montage is empty.

nextClip(clipId) - Returns the ID of the Clip stored after 'clipId', or 0. Clips are not sorted in any special order. Using both firstClip and nextClip allow to access all Montage Clips.

clipName(clipId) - Returns the name of the Clip identified by 'clipId'

clipPosition(clipId) - Returns the time line position of the Clip identified by 'clipId'

clipSize(clipId) - Returns the size of the Clip identified by 'clipId'

setClipName(clipId, name) - Rename the Clip identified by 'clipId'

setTrackName(index, name) - Rename the Track identified by 'index'

moveClip(clipId, newPos) - Move on the time line the Clip identified by 'clipId'

resizeClip(clipId, qlonglong newSize) - Resize the Clip identified by 'clipId'. The size will be limited by the Audio File referenced by the Clip.

setClipDefaultFadeIn(clipId) - Sets the default fade-in shape and time for the Clip identified by 'clipId'

setClipDefaultFadeOut(clipId) - Sets the default fade-out shape and time for the Clip identified by 'clipId'

undo() - Undoes the last command.

Related topics

ECMAScript Reference

4.9.2 ECMAScript Reference

WaveLab's scripting language is based on the ECMAScript scripting language, as defined in `standard ECMA-262` Microsoft's JScript, Netscape's JavaScript and Adobe's Actionscript are also based on the ECMAScript standard. If you are not familiar with the ECMAScript language, there is a large amount of teaching and reference material available online.

This reference page contains a list of all ECMAScript objects, functions and properties supported by the WaveLab scripting system. These are available in any scripting context but are not WaveLab specific. When scripting for the Audio File Workspace for example, (using the Scripting window (Audio File Workspace)) there are specific additional functions that allow you to perform editing commands. For a list of these WaveLab specific scripting commands and functions see WaveLab Scripting Language.

Tip: Some basic scripting examples are included below so you can see the scripting syntax in context. These scripts will work if you copy, paste and execute them in a script window.

Supported ECMAScript subset

The Global Object

Values

- NaN
- Infinity
- undefined
- Math

Functions

- eval(x)
- parseInt(string, radix)
- parseFloat(string)
- isNaN(number)
- isFinite(number)
- decodeURI(encodedURI)
- decodeURIComponent(encodedURIComponent)
- encodeURI(uri)
- encodeURIComponent(uriComponent)

Objects

- Object
- Function
- Array
- String
- Boolean
- Number
- Date
- RegExp
- Error

Examples:

```
//Test if a value is not a number
var isNumber = isNaN("this is a string");

//Print the result in the log window
logWindow.printInfo(isNumber); //Traces "true" because it is a String

//Convert a string to a number
var numStr = "2.345";
var num = parseFloat(numStr);
//Print the result in the log window
logWindow.printInfo(num);

//Encode a string into an internet valid ASCII String
var str = "a sentence with spaces";
var encodedStr = encodeURI(str);
logWindow.printInfo(encodedStr);
```

Function Object

Prototypes

- toString()
- toLocaleString()
- valueOf()
- hasOwnProperty(V)
- isPrototypeOf(V)
- propertyIsEnumerable(V)

Functions

- toString()
- apply(thisArg, argArray)
- call(thisArg [, arg1 [, arg2, ...]])

Example:

```
//Create a new custom marker Object
function customMarker(name, comment, timeSecs)
{
this.name=name;
this.comment=comment;
this.timeSecs=timeSecs;
}

//Create a new instance of the custom marker
var myMarker=new customMarker("A custom marker", "My custom marker comments",5);

//Use prototype function to add a new property to it
customMarker.prototype.samples = null;
myMarker.samples = activeWave.sampleRate() *  myMarker.timeSecs;

//Trace the results in the log window
logWindow.printInfo(myMarker.name);
logWindow.printInfo(myMarker.samples);
```

Array Objects

Functions

- toString()
- toLocaleString()
- concat([item1 [, item2 [, ...]]])
- join(separator)
- pop()
- push([item1 [, item2 [, ...]]])
- reverse()
- shift()
- slice(start, end)
- sort(comparefn)
- splice(start, deleteCount[, item1 [, item2 [, ...]]])
- unshift([item1 [, item2 [, ...]]])

Example:

```
//Create an empty array
var a = new Array();
//Add some items to it
a.push("first array item");
a.push("next array item", "last array item");
//Print them out in the Log window
logWindow.printInfo(a.toString());
//Call the Array's reverse function
a.reverse();
//View the results in the Log window
logWindow.printInfo(a.toString());
```

String Objects

Functions

- toString()
- valueOf()
- charAt(pos)
- charCodeAt(pos)
- concat([string1 [, string2 [, ...]]])
- indexOf(searchString ,position)
- lastIndexOf(searchString, position)
- localeCompare(that)
- match(regexp)
- replace(searchValue, replaceValue)
- search(regexp)
- slice(start, end)
- split(separator, limit)
- substring(start, end)
- toLowerCase()
- toLocaleLowerCase()
- toUpperCase()
- toLocaleUpperCase()

Example:

```
//Create a string variable
var str = new String("WaveLab is a powerful editing tool");
//Make it all upper case
var capsStr = str.toUpperCase();
//View the results in the Log window
logWindow.printInfo(capsStr);
```

Boolean objects

Functions

- toString()
- valueOf()

Example:

```
//Test if an equation is true or false
var isTrue = (1 + 1 == 3);
//Convert the Boolean to a String and trace in the Log window
logWindow.printInfo(isTrue.toString());
```

Number Objects

Functions

- toString(radix)
- toLocaleString()
- toFixed(fractionDigits)
- toExponential(fractionDigits)
- toPrecision(precision)

Example:

```
//Convert a number into exponential notation
var num = new Number(13.3714);
//Show the result in the Log window
logWindow.printInfo(num.toExponential());
```

Math Objects

Values

- E
- LN10
- LN2
- LOG2E
- LOG10E
- PI
- SQRT1_2
- SQRT2

Functions

- abs(x)
- acos(x)
- asin(x)
- atan(x)
- atan2(y, x)
- ceil(x)
- cos(x)
- exp(x)
- floor(x)
- log(x)
- max([value1 [, value2 [, ...]]])
- min([value1 [, value2 [, ...]]])
- pow(x, y)
- random()
- round(x)
- sin(x)
- sqrt(x)
- tan(x)

Examples:

```
//Get a random number from 0 to 1
var r = Math.random();

//Print it out in the log window
logWindow.printInfo(r);

//Trace out Pi in the log window
logWindow.printInfo(Math.PI);
```

Date Objects

Functions

- toString()
- toDateString()
- toTimeString()
- toLocaleString()

- toLocaleDateString()
- toLocaleTimeString()
- valueOf()
- getTime()
- getFullYear()
- getUTCFullYear()
- getMonth()
- getUTCMonth()
- getDate()
- getUTCDate()
- getDay()
- getUTCDay()
- getHours()
- getUTCHours()
- getMinutes()
- getUTCMinutes()
- getSeconds()
- getUTCSeconds()
- getMilliseconds()
- getUTCMilliseconds()
- getTimeZoneOffset()
- setTime(time)
- setMilliseconds(ms)
- setUTCMilliseconds(ms)
- setSeconds(sec [, ms])
- setUTCSeconds(sec [, ms])
- setMinutes(min [, sec [, ms]])
- setUTCMinutes(min [, sec [, ms]])
- setHours(hour [, min [, sec [, ms]]])
- setUTCHours(hour [, min [, sec [, ms]]])
- setDate(date)
- setUTCDate(date)
- setMonth(month [, date])

- setUTCMonth(month [, date])
- setFullYear(year [, month [, date]])
- setUTCFullYear(year [, month [, date]])
- toUTCString()

Example:

```
//Create a new date object
var d = new Date();

//Print it out in the log window
logWindow.printInfo(d);

//Get just the hours
logWindow.printInfo(d.getHours());
```

RegExp Objects

Functions

- exec(string)
- test(string)
- toString()

Example:

```
//Create a new regular expression defining a 5 digit number
var reg = new RegExp(/^\d{5}$/);

//Test a string with it to see if it contains a 5 digit number
var isFiveDigit = reg.test("12345");

//Trace the result to the log window
logWindow.printInfo(isFiveDigit);
```

Error Objects

Values

- name
- message

Functions

- toString()

Types of Native Errors available

- EvalError
- RangeError
- ReferenceError
- SyntaxError
- TypeError
- URIError

Related topics

WaveLab Scripting Language

Scripting window (Audio File Workspace)

Scripting window (Audio Montage Workspace)

4.10 The Spectrum display

In WaveLab's Audio File Workspace, the main wave editing and overview windows can be set to display your audio as a spectrogram (Spectrum).This spectrum mode allows a unique view of your audio and when used in conjunction with the Spectrum editor offers powerful audio restoration and processing functions. The Spectrum display is a "spectrogram", where each vertical line represents the frequency spectrum at the corresponding time location. The bottom of the display corresponds to the lower frequencies while the upper part of the display represents the higher frequencies. The intensity or level of the frequencies are either represented as a color spectrum, from red (maximum intensity) to purple/black (minimum intensity), or as a monochrome intensity (depending on the setting in the Spectrum options dialog). The vertical ruler on the left shows the frequency area (in Hz) for the spectrum.

You can view the Spectrum display in the Audio File Workspace by choosing the "Spectrum" tab below either of the waveform display windows.

Related topics

Spectrogram options

Spectrum editor

4.10.1 Spectrogram options

This dialog allows you to adjust how the Spectrogram is displayed.

You can adjust the visual style of the spectrogram and its resolution. You can also specify a cut-off point above which frequencies are displayed and to plot them using a linear or logarithmic scale. This can be useful for isolating a certain range of frequencies: for example, in sound restoration you may wish to focus on a high frequency band which is more accurately displayed using a linear scale.

You can access this dialog from both wave windows in the Audio File Workspace via the **Spectrum tab** > **Edit settings...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

The Spectrum display

Spectrum editor

4.11 Spectrum editor

This specific tool window allows you to select and perform operations on an Audio File via its audio spectrum.

It uses high quality linear-phase filters to process a spectrum selection both in the time domain and in the frequency domain. This can be useful for advanced audio restoration tasks like removing noise from a specific source in a recording, for example.

Making and editing selections

Spectrum editing can only be performed when a waveform is first displayed in Spectrum display mode and when a spectrum selection is defined. To activate the Spectrum editor, click on the Spectrum edit tool in the command bar (or via **Workspace** > **Specific tool windows** > **Spectrum Editor**). This displays the Spectrogram if this is not already the case (see The Spectrum display).

To select a region, point and drag to define a rectangle. The region selection defines a time and a specific frequency range. This allows you to edit and process audio both in the time domain and in a specific frequency domain, as opposed to standard wave editing which always operates in the full frequency domain.

Notes on spectrum selections:

- When defining a region in a stereo file, a "mirrored" region is automatically created in the other channel. To define a region in only one channel, press **[Shift]** while drawing the region.
- Once you have a region selected, if you place your mouse cursor inside the region, a pop-up appears showing the currently set frequency range (Hz) and the time range (seconds/milliseconds) for the selection.
- A selected region can be moved in any direction by clicking and dragging the region with the cursor.
- If you press **[Shift]** whilst dragging a region, it only moves horizontally thus retaining your selected frequency range. If you press **[Ctrl]/[Command]** + **[Shift]** the region only moves vertically thus retaining the selected time range.

- You can resize a region by placing the mouse cursor at the edges of the region (so that a double-arrow appears) and then clicking and dragging.
- To remove a region that you have drawn, click anywhere else on the waveform and it disappears.
- If you have defined a region as a Source or Target, it can be selected at any time to create a new selection using the same dimensions.

Spectrum editor functions

The Spectrum Editor has the following parts:

- **Selection** - use this part to refine and define your selection(s). You can expand selections, shift selections around and define a source and target region for performing copy operations.
- **Operations** - use this part to perform copying, filtering and processing operations. For copying operations, you require both a source and a target region to have been defined using the Selection tab. You can choose which method to copy audio between the Source and Target regions. You can also choose to apply processing to a single spectrum region using a variety of different modes. See Spectrum Processing Modes for information on each mode type.
- **Master Section** - use this part to process and route selected spectral audio regions through the Master Section and its effects plug-ins. This allows for frequency-selective processing. A selected region's frequency spectrum can be routed to the Master Section where you can choose to process it separately from the non-selected frequency spectrum. The signal is split so that one part (region spectrum or non-selected spectrum) is sent to the plug-ins, while the other part can be mixed with this processed signal, after the Master Section output.

Using the Spectrum editor

You can use the Spectrum editor in two main operational modes:

Region copying and filtering (Operations tab)

This is mainly intended for audio restoration applied over brief time ranges. Selections can be copied, pasted, and filtered. This type of processing is mainly used to reduce, remove or replace unwanted sound artifacts in the audio material, and can be carried out with great precision. This could be useful for example, to replace part of a live recording that contains an unwanted noise (such as a mobile phone ring tone), with a copy of a similar region of the spectrum that only contains a "clean" signal. In general, the spectral Copy/Paste combination gives the best results, given that the source and destination regions are properly choosen.

Master Section processing (Master Section tab)

This allows you to process a specific frequency range via the Master Section. The selected or non-selected regions of the spectrum can be processed differently. You can also employ a number of filters (Bandpass/Low-pass/High-pass) to further refine the particular range of frequencies to be affected by any Master Section effects.

A selected spectrum region can either be:

- Processed separately by the Master Section plug-ins. The non-selected spectrum can either be bypassed or sent to the Master Section.

- Bypassed. This removes the selected spectrum region from the Audio File. The non-selected spectrum can either be routed to the Master Section input or the Master Section output.

- Sent to Master Section output. The non-selected spectrum can either be bypassed or sent to the Master Section input. If the latter is selected, it will be mixed with the selected spectrum region at the Master Section output.

You can access this window by choosing **Workspace** > **Specific tool windows** > **Spectrum Editor**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

The Spectrum display

Spectrum Processing Modes

Spectrogram options

Spectrometer

4.11.0.1 Spectrum Processing Modes

When applying processing to a spectrum region in the Spectrum Editor you can use a variety of different modes:

Damp

This is used to attenuate the level of a region. The Gain parameter specifies the level of the attenuation (it is also possible to set positive gain values, i.e. to boost the frequencies in the region). Three filter types are available to perform the damping:

- Bandpass filter: All frequencies in the region are equally attenuated.
- Low-pass filter: Higher frequencies in the region will be more attenuated.
- High-pass filter: Lower frequencies in the region will be more attenuated.

For the Low- and High-pass filters, a low Filter steepness setting is normally used (around 6 to 18dB).

Blur peaks

This filter operation analyzes the region to find the frequencies with the highest level. The level of these frequencies will be attenuated (or boosted) according to the set Gain value. If the gain is negative, these frequencies will be "blurred" and disappear in the mix as a whole. The purpose of the filter is to mask the loudest frequencies (e.g. it could be used to eliminate a sudden burst of pitched but unwanted sound in the audio material, like acoustic feedback).

This mode works best with the Infinite steepness setting, which is automatically selected when Blur peaks is used. You can, however, freely set any Filter steepness setting. The filter type options are not available when this mode is selected.

Dispersion

This is a special filter that "smears" the dynamics and pitch of the region, without changing the actual frequency content. This works best on lower frequencies, to mask the identity of a signal without changing the frequency spectrum.

This mode works best with the Infinite steepness setting, which is automatically selected when Dispersion is used. You can, however, freely set any Filter steepness setting. The filter type options are not available when this mode is selected. Dispersion can also be used to create special effects as it warps the sound in an unique way.

Fade Out

This can be used with any of the three filter types (Bandpass/Low-pass/High-pass). It gradually filters the frequencies in the region along the time axis (from nothing at the left edge of the region, to maximum allowed by the gain parameter at the right edge), useful for gradually removing frequencies from a region.

Fade In

This works like Fade out but inversed. It can also be used in conjunction with any of the three filter types (Bandpass/Low-pass/High-pass).

Fade out then in

This is a mix of the two previous options; the filtering effect gradually fades out until the middle of the region, then fades back in. It can also be used in conjunction with any of the three filter types (Bandpass/Low-pass/High-pass).

Fade in then out

The inverse of the above. It can also be used in conjunction with any of the three filter types (Bandpass/Low-pass/High-pass).

Related topics

The Spectrum display

Spectrogram options

Spectrum editor

4.12 About Regular Expressions

At various places in WaveLab you can use Regular Expressions to build complex text matching capabilities into your conversion and renaming processes. A regular expression is a set of text symbols that describe a method to find a specific text string within a large body of text, and then apply a specific operation to this text string. Regular expressions are available for the advanced user to perform powerful string search/replace operations, for example, in batch renaming or batch processing.

Throughout WaveLab, wherever you see this bulb icon there is a field where you can create your own regular expressions. A fly-out menu containing shortcuts to build up the basic syntax of an expression is also available.

It is beyond the scope of this Help system to describe this subject thoroughly (there are good internet resources available), but please read on if you wish to check out the basic building blocks used in regular expressions.

Common expressions

A regular expression is a formula composed of characters that have special meanings (called **operators**). Other characters are just plain letters and numbers that are searched for.

The search engine browses the target text one character at a time and stops as soon as it finds a sequence of characters that matches the regular expression. Searches are not case-sensitive. Note that there are various versions of regular expressions: WaveLab uses a version that represents a good compromise between power and ease-of-use.

The term **expression** refers to a single character, a character class, or a sub-expression enclosed with () or {}.

The following items are available on the Regular Expression pop-up menu:

Menu item	Operator	Description
Any character	**.**	Symbolizes any character
Character in range	**[ ]**	A bracketed text is treated as a single character, for example: [AEW13] means A or E or W or 1 or 3. A hyphen within the brackets specifies a range of characters. For example, [F-I] means F or G or H or I, and [A-Z0-9] matches all letters and all digits.
Character not in range	**[^]**	A circumflex located at the first position in a bracket is a complement operator. In other words, all characters match except those included in the bracket. Example: [^E] means any character *except* E.
Slash	**/**	A slash before a character means that this character is to be treated literally, not as an operator.
0 or 1 match (1 if possible)	**?**	Matches 0 or 1 time the preceding expression. 1 repeat if possible is grabbed, then the rest of the regular expression continues to be evaluated.
0 or 1 match (0 if possible)	**??**	Matches 0 or 1 time the preceding expression. 0 repeat if possible (the NEXT step in the regular expression is also evaluated and has priority).
0 or more matches (as many as possible)	*	Matches 0 or more times the preceding expression. As many repeats as possible are grabbed, then the rest of the regular expression continues to be evaluated.
0 or more matches (as few as possible)	***?**	Matches 0 or more times the preceding expression. As few repeats as possible are grabbed (the NEXT step in the regular expression is also evaluated and has priority).
1 or more matches (as many as possible)	**+**	Matches 1 or more times the preceding expression. As many repeats as possible are grabbed, then the rest of the regular
WaveLab 7		expression continues to be evaluated.
1 or more matches (as few as possible)	**+?**	Matches 1 or more times the preceding expression. As few repeats as possible are grabbed (the next step in the regular expression is

The Shortcuts sub-menu on the Regular Expression pop-up menu contains the following options:

Menu item	Operator	Description
Any digit	**/d**	Special code to symbolize any digit, as [0-9].
Any non-digit	**/D**	Special code to symbolize any non-digit, as [^0-9].
Any letter	**/l**	Special code to symbolize any letter, as [a-z].
Any non-letter	**/L**	Special code to symbolize any non-letter, as [^a-z]. - Any alphabetic /w Special code to symbolize any alphabetic character, as [0-9a-z].
Any non-alphabetic	**/W**	Special code to symbolize any non-alphabetic character, as [^0-9a-z].
Any number	**/u**	Special code to symbolize a number (without a sign).
Number with possible +- before.	**/i**	Special code to symbolize a number which can be preceded by a + or - sign.
Quoted string	**/q**	Special code to symbolize quoted text.
Simple word	**/z**	Special code to symbolize a simple word (a sequence of letters surrounded by non-letters, e.g. spaces).

All special characters used for regular expressions are available from the "Special characters" submenu. There is also a Presets sub-menu with pre-defined operations, which include the following:

Menu item	Description
1st/2nd/3rd word	This searches for the first, second or third word, respectively (separated by a space).
Last word	This searches for the last word (separated by a space).
1st/2nd/3rd expression in parentheses	This searches for the first, second or third string enclosed in parentheses, respectively.
Last expression in parentheses	This searches for the last string enclosed in parentheses.
1st/2nd/3rd expression in brackets	This searches for the first, second or third string enclosed in brackets, respectively.
Last expression in brackets	This searches for the last string enclosed in brackets.

Related topics

Batch Renaming

Convert marker type

Regular Expressions

Chapter 5

Audio File editing

The Audio File Workspace's Wave window is the heart of WaveLab's audio editing capabilities. This is where you view, play back and edit individual Audio Files.

The Wave window gives you a graphical representation of an Audio File. It consists of two parts:

- The upper waveform (**Overview**)

  This is mainly used for navigating through long files. Since you can have different zoom factors in the two areas, the Overview can display the entire wave while the Main view only shows you a detailed portion for precise editing.

- The lower waveform area (**Main view**)

  This is where the audio editing action takes place. You can perform various operations on the wave such as copying, cutting, pasting, moving and deleting.

Waveform views

In the Audio File Workspace, the main wave editing window and the overview can independently be set to display one of three display modes:

- **Wave** - this is the default waveform display.

- **Loudness** - this displays the average loudness of an Audio File, which can be useful in many circumstances. For example, this display mode gives you an overview of the amount of compression applied to a song. The more dips in the curve, the more dynamics in the audio material. An even curve with few dips indicates that the material is compressed with a limited dynamic range (which is common for modern pop/dance music). Up to four different curves can be shown simultaneously in the Loudness envelope display. These curves have different colors and represent the average loudness of the signal in different areas of the frequency spectrum. This can be useful to visualize where in the frequency domain the loudness is "spread" at a given time, or to display the loudness of a specific part of the spectrum. The curves can be shown individually or in any combination.See Loudness envelope options for more details.

- **Spectrum** - This displays the audio as a "spectrogram" which allows you to view the level intensity of each area in the frequency spectrum across the time line. The Spectrum display mode in conjunction with the Spectrum editor provides you with a unique

editing/restoration tool. See The Spectrum display for more information on the unique Spectrum display mode.

Rulers

There are both time and level rulers. These can be hidden and displayed (see the Ruler and Wave Display context menus). You can also set which units of time and level they show. To access the settings for both rulers, right-click anywhere on the relevant ruler.

Editing Tools

Precise editing of your audio is achieved in the main (lower) waveform window. There are three main tools that are used for directly editing an Audio File:

Time Selection cursor - This is the default cursor and allows you to select and resize regions of audio you wish to edit. Click and drag in the main waveform window to make a selection. You can modify the selected region by click-dragging on the edges of the selection. You can also modify a selected audio region by holding down **[Shift]** and clicking on the waveform. The selection extends/shrinks to the clicked point. If you hold the cursor over one channel of a stereo file you can make selections in that individual channel. You can double-click inside a region to select it then drag the mouse to select the next or previous regions, that is, you snap to markers while dragging.

Spectrum Selection cursor - this cursor allows you to select and define regions when in the Spectrum view. See Spectrum editor for more information on making selections and using the Spectrum view for advanced editing purposes. **Pen tool** - this tool allows you to redraw the waveform directly in the wave window. It can be used to quickly repair waveform errors by hand. The Pen tool can only be used if the Zoom resolution is set to at least 1:8 (one pixel on the screen equals 8 samples) or higher. If you wish to redraw the waveform of both channels in a stereo file at once, press **[Shift]** while drawing.

Once you have defined an audio selection there are many editing tools available in the **Edit** menu of the Audio File Workspace for you to explore. These range from basic editing commands such as Copy and Paste to special functions which can perform tasks such as trimming, muting, crossfading, appending, prepending, silencing and repeating an audio selection, for example. It is well worth exploring the tools available in this menu.

There a number of options available that affect editing in the waveform window. See Audio File editing preferences for more information on the various preferences available.

Editing Shortcuts

You can navigate through your Audio File by dragging the Time Selection cursor. Alternatively you can use the following keyboard shortcuts:

- Use the [Left Arrow] and [Right Arrow] to move the cursor one pixel in either direction.
- Use **[Ctrl]/[Command]** + [Left Arrow] and [Right Arrow] to move the cursor twenty pixels in either direction. You can also use the [Page up] and [Page Down] keys.
- Use the [Home] and [End] keys to navigate to the beginning and end of the wave.

You can also scroll and zoom within the Audio File Workspace using several other mouse and keyboard shortcuts. See Zooming and scrolling for more detailed information.

To audition your Audio File hold down **[Alt]/[Option]** and click anywhere in the waveform to begin playback.

Dragging can also be used with several editing functions in the Audio File Workspace. See Audio File Workspace in Dragging operations for more information.

Related topics

Offline processing

Analysis

Metering

Transport controls

Specific tool windows

Shared tool windows

Mix dialog

The Spectrum display

Spectrum editor

Tools

Zoom factor

Peak Files

5.1 Offline processing

Offline processes are used for a variety of editing purposes and creative effects. Offline processing refers to functions that are only performed when saving the file.

"Offline" means the process can't be monitored during playback, either because they require too much processing power or because they require an analysis pass prior to processing. Note that WaveLab only writes temporary files when applying an effect, so the original Audio File is safe. Only when saving the original is the original audio possibly modified.

These offline processes differ to realtime processes (such as listening to effects via the Master Section for example), which only temporarily render an effect while audio is played "through" them.

Applying processing

Processing can be applied to a selection or to a whole Audio File. For certain processing operations processing the entire file is necessary. This will be indicated by the effects respective dialog. If "Process whole file if there is no selection" is selected in the Audio File editing preferences editing tab, the whole file will automatically be processed if no selection exists.

To apply processing to some audio, proceed as follows:

1. Make a selection. If the file is in stereo, you can apply processing to either channel or both, by selecting one channel or both.

2. Select the desired offline process from the Process menu.
3. If a dialog appears, make the appropriate settings. If the dialog uses presets, you can load one of them to fill out the settings automatically.
4. When you have finished with the settings in the dialog, click the Apply/Paste/Process button to permanently render the effect to file.

In some cases a status bar dialog is displayed showing the progress of the operation. If you need to interrupt a long process, click the "Cancel" button on the status bar dialog.

Undo/Redo processing

Even though offline processes appear to alter the file permanently, you can always reinstate a previous version using WaveLab's "undo" feature. The number of Undo/Redo commands is limited only by your available hard drive space. Access the Undo and Redo commands via **Edit** > **Undo** or **Edit** > **Redo**.

Tips for using Offline Processing

Processing dialogs are non-modal - meaning that they stay open and on screen even after processing has been applied. This means that you can still playback audio and access undo/redo commands to audition the results of your effect and make any further adjustments to any settings. You can also use key commands and access any other menu commands while leaving the offline process dialog open. Remember that using and saving presets when you have settings you are happy with can also save you time.

WaveLab contains a powerful array of offline processing tools including the following:

Change Level

Level envelope

Normalize Level

Silence Generator

Loop Tone Uniformizer

Pitch correction

Time stretching

Loop Tweaker

Effect morphing

Loudness distribution

Loudness normalizer

Pan Normalizer

Pitch bend

Pitch quantize

Related topics

Presets

Rendering

5.1.1 Change Level

This dialog allows you to apply a gain to change the level of an Audio File.

You can also use "Find current peak level" to obtain a report on the peak level of the current audio selection (or the peak level in the whole file, if "Process whole file if there is no selection" is selected in the preferences). This can be useful if you wish to calculate how much you can increase the overall gain of a file without introducing clipping (exceeding 0dB), for example.

To change the level, enter the desired gain and click Apply.

You can access this dialog by choosing **Process** > **Change level...** in the Audio File Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Value editing

5.1.2 Level envelope

This dialog allows you to create a volume envelope which can be applied to a selected range or a whole Audio File. This could be useful if you need to even-out loud and quiet parts, for example.

You can edit the envelope by:

- Double clicking on the line to add a point: double click on a point to remove it.
- Selecting multiple points (using **[Shift]** click) allows you to move the selection as a group.

You can also use the controls to delete or reset points in the envelope. Use the envelope smoothing button to toggle the envelope points between a straight (polygonal) line or a curved path.

Click Apply to apply the envelope to the audio selection or file.

You can access this dialog in the Audio File Workspace via **Process** > **Level envelope...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Adjusting Envelopes

5.1.3 Normalize Level

This dialog allows you to change the peak level of your Audio File.

Enter the desired peak level (in dB) that you wish the audio selection to have. You can also use "Find current peak level" to obtain a report on the peak level of the current audio selection (or the peak level in the whole file, if "Process whole file if there is no selection" is selected in the preferences). You can choose to apply the same gain to both stereo channels (Stereo Link) or mix down to a mono file (Mix to Mono) with the assurance that no clipping will occur when both channels are mixed together.

To normalize the selected audio, enter the desired peak level and click Apply.

You can access this dialog by choosing **Process** > **Level Normalizer...** in the Audio File Workspace.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Value editing

5.1.4 Silence Generator

This dialog allows you to generate "silence", either by replacing an audio selection, or by specifying a duration.

You can also specify a cross fade at each end of the silent region, for a smoother transition. Radio buttons determine where the silent period is placed in the audio.

In WaveLab you can also specify a file to be inserted instead of "true" silence. This can be useful for example, if you wish to keep the ambience, or background noise of a particular environment in a recording.

You can access this dialog in the Audio File Workspace via **Edit** > **Silence (advanced)...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Silence

5.1.5 Loop Tone Uniformizer

This dialog allows you to create sounds that will loop, from audio that is seemingly "unloopable". These are normally sounds that constantly decay in level or continuously change in timbre. The Loop Tone Uniformizer applies processing to the sound that evens out changes

in level and timbral characteristics in order for a sound to loop properly. For example, this could be useful for creating looped samples for a softsynth or hardware sampler.

To use the Loop Tone Uniformizer you need to have a loop defined using a pair of loop markers . The original length of the loop is not changed.

Uniformizers

This tab allows you to specify the method/s used to even out the sound you wish to loop. Choose between "Slice mixing" and/or "Chorus smoothing". For a full explanation of how each of these methods operate use the "What's this?" help function.

For slice mixing, you may need to experiment to see how many slices are needed. Generally the more slices you use, the more natural the sound will be (up to a certain point).

Pre Cross fade

The Loop Tone Uniformizer also includes the option to cross fade the end of the loop with the start of the newly processed section so that transition into the newly looped section is smoother during playback. Use the envelope drag points or value sliders to adjust the cross fade.

A note about Post Cross fading

If you have used the Loop Tone Uniformizer within a region of an Audio File, the transition from the end of the loop into the original file in many cases won't be very natural. This can be fixed as follows:

1. Close the Loop Tone Uniformizer and open the Loop Tweaker.
2. Click on the Cross fade tab and deactivate the Cross fade option (the check box).
3. Click on the Post Cross fade tab and make sure Cross fade is activated.
4. Set up the post cross fade parameters and click Apply.

It can also be a useful tip to lock the loop markers after you use this command on a selection within an Audio File. Once the loop is processed in place, moving the loop markers will upset playback of the loop.

You can access this dialog in the Audio File Workspace via **Process** > **Loop Tone Uniformizer...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Markers

Adjusting Envelopes

5.1.6 Pitch correction

This dialog allows you to detect and change the pitch of a sound without affecting its length.

- **Amount of Shift** - use these controls to find out the current pitch of the audio and if desired, calculate the required shift to match a specific pitch.

- **Method** - use these controls to adjust the method and quality you wish to use when processing the pitch change. You can also choose how the length of the sound is affected by the operation. By default this process will not change the length of the sound.

This tool could be useful for fixing an off-key vocal note in a live recording, or tuning the pitch of a kick drum sample to fit a particular song, for example.

You can access this dialog from the Audio File Workspace via **Process** > **Pitch Correction...**.

Pitch correction is also available as a batch processing plug-in in the Batch Processor Workspace. In addition, Pitch correction for a Clip is available from the Focused Clip window (via **Edit** > **Pitch shift...**).

Note that for Batch processing and Clip processes, the envelope option is not available.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

5.1.7 Time stretching

This dialog allows you to change the duration of an audio selection (usually without changing its pitch).

You can view exact time information on the original audio selection you wish to stretch and choose the amount you wish to "stretch" it in seconds, beats per minute or as a percentage. You can also choose which method WaveLab uses to stretch the audio and the quality/speed of the processing.

If you select "Use modulation envelope" you can vary the time stretch over time using an envelope editor that becomes available. When choosing a method, it is useful to experiment with the different algorithms available to find which one suits your source material the best. Use "What's this?" to find out more information on each method and determine which one will produce the best time stretch without introducing unwanted artifacts.

This function is useful if you want to speed up or slow down a recording to match other material, such as matching the tempo of two drum beats, or fitting an audio track to a section of video, for example.

Time stretching is also available as a batch processing plug-in in the Batch Processor Workspace. In addition, Time stretching for a Clip is available from the Focused Clip window.

Note that for batch processing and Clip processing, the modulation envelope option is not available. You can access this dialog from the Audio File Workspace by choosing **Process** > **Time stretching...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

5.1.8 Loop Tweaker

This dialog allows you to refine a region of audio to loop seamlessly.

The Loop Tweaker can just be used to "tweak" an existing loop selection so that it loops perfectly or you can also use it to create a loop from material which does not naturally repeat. To use the Loop Tweaker you must first have a loop defined using a pair of loop markers .

The Loop Tweaker consists of the following tabs:

Loop points adjustment

This is a visualization of the beginning and end of the waveform between the loop markers. You use this dialog to manually refine a loop selection by dragging on the waveform left/right or using the automatic search buttons to find the nearest "good" loop point. The aim is to align the waveforms so they meet in the middle at a zero-crossing point with the waveforms matching as closely as possible, being "in phase". When you adjust your loop start and end points within the dialog the start and end loop markers in the main waveform window will adjust accordingly. Note that this movement may or may not be visible depending on the amount you move the markers and the zoom factor you have selected.

It is helpful to have the transport looped during playback so you can hear the difference when you adjust the loop markers within the dialog. Use the Display options to adjust how the Loop Tweaker waveform is visualized. You can also use the temporary memory buttons to capture a number of different loop "candidates" and audition them one by one. Note that if you are not using a cross fade or post cross fade you do not need to click Apply when just tweaking loop points. You can also leave this dialog window open and manually adjust the position of the markers in the main waveform windows if you need to make large adjustments.

Cross fade

This tab allows you to apply a cross fade at the end of a loop by mixing a copy of the beginning of the loop with it. This can be useful to smooth the transition between the end of a loop and its beginning - especially when using material that doesn't naturally loop. Use the envelope drag points or value sliders to adjust the cross fade envelope. Click Apply to process the cross fade.

Post cross fade

This tab allows you to cross fade the loop back into the audio after the end of the loop. This is done by mixing a copy of the loop back into the audio. Use the envelope drag points or value sliders to adjust the cross fade envelope. Click Apply to process the post cross fade.

Using the "Copy" button you can insert multiple copies of the loop into the active Audio File, creating a smooth audio sequence, since copies will loop without an audible transition.

You can access this dialog by choosing **Process** > **Loop Tweaker...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Markers

Adjusting Envelopes

Zero crossing

5.1.9 Effect morphing

This dialog within the Audio File Workspace allows you to gradually mix two audio ranges that have different effects/processing applied to them.

Effect morphing allows you to smoothly "morph" or blend from one effect to another, or from an unprocessed audio segment to a processed audio segment. Effect Morphing always involves two audio ranges. Normally, these are two versions of the same audio range, of which one is unprocessed and the other processed.

Effect morphing works by using WaveLab's Undo buffer to mix a copy of the processed audio with the unprocessed version. You can also use any other bit of audio from the application's clipboard that has the same length as the one you wish to morph with.

To set up a basic effect morph, proceed as follows:

1. Start by making a selection range over the time you wish the effect morphing to take place.
2. Now process this range using any Master Section effect(s), or use some other off-line processing. You cannot use processing/effects that alter the length of the selection, e.g. time stretching.
3. Open the Effect Morphing dialog by choosing **Edit**>**Effect Morphing...**.
4. Adjust your envelope points over time between 0% and 100%. This will determine the level and direction of the morph. For example, starting at 100% and ending at 0% will fade out the effect you just applied.
5. Choose "Unprocessed selection" for this example. As mentioned previously you can also use any audio of the same duration from the clipboard.
6. Click Apply to apply the Effect morphing.

You can access this dialog in the Audio File Workspace via **Edit** > **Effect morphing...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

5.1.10 Loudness distribution

This dialog allows you to measure the most frequent loudness values found in an Audio File (this is not the same as average loudness). The graphic peaks found by the analysis represent these values.

The tool gives an answer to the question: "How often does a given loudness (vertical scale, in dB) appear in the whole file?". The percentage figure is relative to other peaks. Some simple examples:

- 1. If, for instance, you have a sine of 0 dB / 2 second followed by a sine of -6 dB / 2 seconds. This means the global audio material has as much 0 dB material as - 6dB material: you will see 2 peaks (0 / -6 dB), each at 100%.

- 2. If, for instance, uou have a sine of 0 dB / 1 second followed by a sine of -6 dB / 3 seconds. This means there are 3 times more -6 dB material than 0 dB material. You will see a peak of about 33% at 0 dB, and another peak of 100% at -6 dB.

This analysis is useful for learning about how the loudness of your music is distributed.

You can access this dialog in the Wave File Workspace via **Analysis** > **Loudness distribution...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

5.1.11 Loudness normalizer

This dialog allows you to adjust the loudness of a file. Because it relates to loudness as opposed to maximum peak level, this tool works quite differently from the Level Normalizer. A typical application is to specify a loudness, for example -12dB, and the Loudness Normalizer processes the audio to match this level of loudness.

Like Gain Change, increasing the loudness to a certain value can introduce clipping. To avoid this occurring, a peak limiter (the Peak Master plug-in) can be used as part of the process. The Loudness Normalizer raises the loudness and limits peaks in the signal at the same time (if needed), to achieve the target loudness.

In the case of stereo files, both channels are processed independently. This process happens over several stages; analysis precedes the final rendering. The Loudness Normalizer dialog gives access to useful statistics about the file and the Loudness distribution tool, and you can choose to remove any DC offset in the file.

You can access this dialog from the Audio File Workspace via **Process** > **Loudness normalizer...**.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Loudness Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

DC Offset

Loudness distribution

5.1.12 Pan Normalizer

This dialog allows you to ensure that both channels of a stereo file have the same level or loudness. Since loudnesss is more important than peak volume in the perception of sound, this is a powerful tool to get the best possible stereo balance.

This process uses two passes to first analyze the audio and then render any level changes required.

You must have a stereo selection in a stereo file to apply this process.

You can access this dialog from the Audio File Workspace via **Process** > **Pan Normalizer...**.

This function is also available as a batch processing plug-in. You can access it from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **Pan Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

5.1.13 Pitch bend

This dialog allows you to change the pitch of a sound over time. Note that changing the pitch using Pitch bend will affect its duration unless length preservation is activated.

Using an envelope you can 'draw' the curve you wish the pitch to follow. The pitch offset is displayed along the vertical ruler of the envelope and the range the envelope effects can be adjusted using the spin control. Positive pitch values produce sounds with a higher pitch and of shorter length, negative values produce sounds with a lower pitch and of longer length.

If length preservation is activated, you can choose the algorithm that is used to perform the pitch bend operation. Depending on the type of audio material you are processing, choose the appropriate mode. For more information on the different modes, use the "What's This?" tool. You can also adjust the quality used when processing the pitch bend. The quality setting and the selected mode will affect the processing time for this effect.

This function could be used for creating the classic 'tape stop' effect, or for blending the tempo/pitch of one track into another, for example.

You can access this dialog from the Audio File Workspace via **Process** > **Pitch bend...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

5.1.14 Pitch quantize

This dialog allows you to correct the pitch of an Audio File.

The musical notes in the audio selection are quantized to the nearest semitone. Pitch quantize works best on recordings that have a single fundamental frequency, such as voice or

single instruments. It can be useful for 're-tuning' the pitch of an imperfect recorded vocal performance, for example.

The dialog allows you to choose the reference frequency (normally 440 Hz) and the time taken for the pitch correction to reach the nearest semitone. An explanation of when and how to adjust the Slur time is given using the 'What's this?' tooltips.

You can also choose to preserve the formants of the original material. Choosing this option can sometimes help to produce a more realistic result when correcting the pitch of vocals or acoustic instrument sounds.

You can access this dialog from the Audio File Workspace via **Process** > **Pitch quantizing...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

Formant

5.1.15 Presets

WaveLab's powerful preset system allows you to create your own presets to store commonly used settings. They are available every time you open WaveLab. Presets can be saved for many parameters such as plug-in settings, Audio File Formats or File Attributes, for example. You can access the Presets pop-up menu wherever you see one of these icons: or . Presets are saved as local files on your hard drive, allowing you to conveniently back-up or copy them for use on another computer.They are also useful when used in conjunction with batch conversions and Scripting.

Working with Presets

When working with Presets you save/edit and load them from their own pop-up menu. Depending on the context, the available functions sometimes vary but generally you can:

- **Save/Save As...** - creates a new Preset file by saving the currently selected settings to your hard disk.
- **Explore presets...** - opens the folder where any existing Presets are stored. This allows you to delete, duplicate and organize your settings into sub-folders using the flexibility of the operating system's file browser.
- **Restore factory settings** - restores the current settings to the factory default.
- **Preset list** - select a Preset from the list of those currently available.
- **Store/restore temporary presets** - when available, use the temporary slots provided to save your settings for the duration of the session. This is useful if you want to quickly test and compare different settings. Select the option "Store temporarily" and from the

submenu select the number under which you want to save (#1 - #5). To quickly load the saved settings again, select the corresponding number from the "Restore" submenu.

- **Define shortcut for the current setting** - when available, this allows you to assign a shortcut to the current preset using either a key shortcut, keyword or via a MIDI trigger. For example, if you have a preset to normalize audio at -0.1 dB and you assign it a key shortcut then everytime you execute the shortcut, the preset is applied to the audio selection, without the need to open the dialog.

VST-2 Presets

VST plug-ins have their own preset handling. When you click the preset button for this type of effect the following options are available:

- **Load/Save Bank** - this allows you to load and save complete sets of presets. The file format is compatible with Cubase.
- **Load/Save Default Bank** - this loads the default bank of presets, or saves the current set of presets as the default bank.
- **Load/Save Effect** - this loads or saves one preset at a time. The file format is again compatible with Cubase.
- **Edit name of current program** - allows you to define or edit the name for the currently selected preset.
- **Preset list** - select a Preset from the list of those currently available.

VST-3 plugins have a simplified list of options.

About Preset files

Each preset is stored as an individual file inside WaveLab's main presets folder. This makes it easy to back them up or copy them to another computer. You can change the location of this main Presets folder via **Global Preferences** > **Setting location**. You have the choice of the following locations:

- a standardized location that can be accessed by all users of the computer. That is, all users will share the same settings.
- a standardized location accessed by the current user (the default). That is, each user has his/her own private settings.
- any specific location of your choice. This allows you to customize it for your own requirements.
- a location relative to the application. This allows you to use the application from an USB dongle.

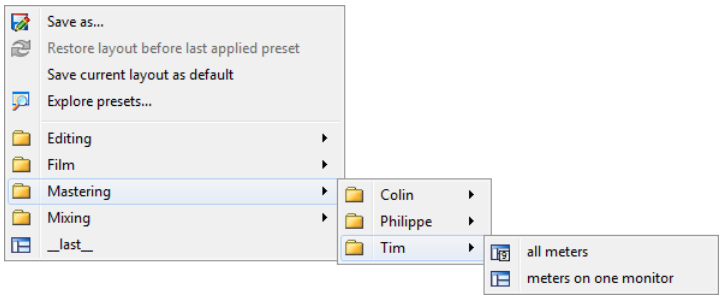
Default locations - By default the main Presets folder is found at:

- **Windows 7** - *C:\Users\[User Name]\AppData\Roaming\Steinberg\WaveLab 7\Presets*
- **Windows XP** - *C:\Documents and Settings\[User Name]\Application Data\Steinberg\WaveLab 7\Presets*
- **Mac OS X** - *root/[User Name]/Library/Prefences/WaveLab 7/Presets/*

You can also find this folder by using the "Open active settings folder" link in the preferences pane. This opens your file browser to wherever the main Presets folder is currently located.

Inside this Presets folder each type of preset file has its own folder. For example, all the presets for the Normalizer are stored within *C:\Users\[User Name]\AppData\Roaming\Steinberg\WaveLab 7\Presets\Normalizer* on Windows 7. When you open the Normalizer dialog and choose a preset via the pop-up menu, the files in this directory will appear as available presets. The reason each preset type has its own dedicated folder is so that WaveLab can locate them automatically when you open the preset pop-up menu.

You can organize your presets into sub-folders so they appear as sub-menus. This can be useful if you wish to organize a lot of presets into your own categories. You can continue to nest folders inside one another to build a standard tree structured menu system. Below is an example of how this looks for a set of window layout presets:



Related topics

Scripting

5.1.16 Rendering

To Render effects into a temporary or final file, you use the Render function (note that the "Save" function from the File menu does not Render!). The Render function is part of the Master Section and in other applications is sometimes referred to as "bouncing down" or "mixing down".

Rendering a file "prints" all audible settings and effects that the audio is being routed through in the Master section, to the resultant Audio File. This includes any plug-ins, the level of the faders and any dithering you have set in the Master Section. In WaveLab you normally use the Render function to:

- Mix down a file in a Wave window to a new Audio File, complete with Master Section effects, dithering and other settings.

- Mix down a complete Audio Montage to an Audio File.
- "Apply" all settings in the Master Section to a file in a Wave window.

When you render a file you can choose the format of the new Audio File and whether you wish to create a new file or render the file in place. When the file is rendering you will see a progress bar in the status bar.In WaveLab you can also view this Progress bar in the Background tasks window (a shared tool window that can be set to float independently).This allows you to monitor the progress of rendering, pause and if necessary cancel the rendering process.

Rendering tips

When you are new to the rendering process in WaveLab there are a few things to note:

- when rendering a file with the default option "Bypass Master Section on resulting Audio File" selected in the Render dialog, the Master Section will be bypassed completely when the new file has finished rendering. If this option is **not** selected you will hear the Master Section effects being played back on top of the newly rendered effects. You can check if the Master Section has been bypassed by checking the status of the "Play through Master Section" button, which is located at the bottom right hand corner of the status bar.
- the Master Section's "Bypass" switch affects playback only. So even if you can't hear effects when you are playing back your audio, any applied effects are still rendered.
- the "Effect on/off" switch for each individual plug-in affects both playback and Rendering.
- if you are working with an Audio File/montage that contains lots of effects and/or tracks and you experience playback problems, it can be useful to mix down a copy of your montage/audio file to free up some processing power. You can then continue editing the resultant file with the effects rendered. Remember to save a copy of your session before doing this so you can go back and edit any levels or effects if you need to.

For more information about choosing settings when rendering see the Render Wave window and Render Montage window. For more information on applying effects see the Master Section.

Related topics

Render Wave window

Render Montage window

Master Section

Audio Plug-ins

5.2 Analysis

WaveLab provides you with a comprehensive set of tools for analyzing your audio and diagnosing any errors. You can view your Audio File in many different ways using the suite of audio meters, via its frequency spectrum, or even in three dimensions. There are also several tools to interrogate any sample of your audio and find any errors or anomalies.In WaveLab you can even compare two Audio Files with the Audio file compare tool and view audio in a Spectrum or Loudness view for greater precision when editing. Below you can find links to some of these tools used in WaveLab. Most of them can be found in the **Analysis** menu. The meters can be found in the **Meters** menu or via the **Workspace** > **Shared tool windows**. More specialized tools also exist as part of some offline processing dialogs, such as the "Find current peak level" command in the Change level dialog, or the "Eliminate DC Offset" dialog, for example.

Related topics

Global analysis

Audio file comparer

3D Frequency Analysis

Loudness distribution

Metering

The Spectrum display

Audio File editing

Offline processing

5.2.1 Global analysis

This dialog allows you to perform advanced analysis on your audio to identify areas with specified properties. It can be used to find problem areas such as glitches or clipped samples, or just to check general information such as the pitch of a sound.

How it works When you analyze a section of an Audio File, WaveLab scans it and extracts information which it displays in the dialog. It also "pin-points" sections of the file that meet specific characteristics, for example, sections being very loud or almost silent. You can then browse between these points, set markers or zoom in on them.

Types of analysis On most of the tabs, you will find settings determining exactly how the analysis will be performed. Each tab focuses on a particular analysis area:

- **Peaks** - this tab is for finding individual samples with very high decibel values.
- **Loudness** - this tab is for finding sections that are perceived by the human ear as louder or weaker in volume. WaveLab uses an accurate method (RMS - Root Mean Square), to measure a consecutive section of samples and then average their value.
- **Pitch** - this tab is used for finding the exact average pitch of a section of audio. The method works best on monophonic material (single notes, without chords or harmonies) and assumes the analyzed section has a relatively stable pitch. As a general rule try to analyze the sustain portion of a sound, rather than the attack.

- **Extras** - this tab is for informing you about DC offsets and the significant bit resolution of the file. This can be useful, for example, if you wanted to check whether a 16 bit file really uses 16 bits (or if it was actually recorded with only 8 bit resolution and then upscaled to 16 bits).

- **Errors** - this tab is to help you find glitches and sections where the audio has clipped (gone beyond 0 dB).For more control over the analysis, use Error detection and correction. **Checking and browsing the results** Checking the results on the Pitch and Extra tabs is simple, since only one value is returned for the whole section of analyzed audio. For all the other tabs, the analysis methods returns a series of exact points ("hot points") in the file or section of audio. Use the relevant buttons in each tab to select the "hot points" you wish to look at in more detail. Using the slider you can then browse or skip between these points, adding markers or focusing the waveform display at that particular point in the file.

You can access this dialog in the Audio File Workspace via **Analysis** > **Global Analysis...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Error detection and correction

3D Frequency Analysis

5.2.2 Audio file comparer

This analysis tool allows you to compare two Audio Files for differences between them. It can create a "delta file" that contains these differences. It can also drop markers at positions where differences are found.

This can be useful for:

- Judging the effect of using an equalizer by comparing the file before and after. The delta file will show what was added or removed.
- Check the noise added by a processor.
- Compare two digitally recorded files to check for dropouts.

Note - Make sure that the two documents you want to compare are open.

You can access this dialog via **Analysis** > **File comparator...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

5.2.3 3D Frequency Analysis

This function in WaveLab allows you to view a wave file in both the frequency and time domains. Although a wave display (time domain) tells you a lot about where one sound

starts or ends in a file, for example, it doesn't tell you anything about the timbral contents of the file. A frequency graph (frequency domain) allows you to investigate the individual frequency components of an Audio File. With the addition of the time dimension you can follow these frequencies in your Audio File over time and pick out individual snare hits or vocal notes in a song, for example. The graph used in WaveLab is sometimes called a 3D-Spectrogram.

Displaying and viewing the graph

You can choose to analyze a selection of audio or an entire file. If you select a stereo recording, a mix of the two channels will be analyzed. Once you have your region selected or no region (in the case you wish to analyze the whole file) choose **Analysis** > **3D Frequency Analysis...**. The graph window will appear.

There is a scrub wheel that allows you to rotate the view around and a settings button that displays the range of frequencies currently displayed. Click on this button to open the settings dialog and edit the frequency range displayed, as well as to access other options.

Tip: The length of the selection affects the accuracy of the analysis. For short selections the result will be very detailed. For longer selections (over a minute or so) the results are generally less detailed as the harmonic content may vary between "measuring points", and as such are not included in the graph. You might for example make a separate analysis of the attack (beginning) of a sound, since the most drastic variations usually occur there.

Some example uses for 3D Frequency Analysis

The 3d Frequency Analysis graph in WaveLab can be used for many purposes, for example:

- to see how the frequency spectrum is distributed in a mix.
- as a basis for EQ-ing, so that you know which frequencies to reduce or boost.
- to see which parts of the audio spectrum a certain background noise occupies (for removing by filtering).
- for educational purposes - these graphs tell you a lot about how different sounds are "built".

For detailed spectrum analysis, the 2D Spectrometer remains the best choice, as it provides more accurate viewing and more options.

Related topics

Frequency range

Metering

5.2.4 Loudness distribution

This dialog allows you to measure the most frequent loudness values found in an Audio File (this is not the same as average loudness). The graphic peaks found by the analysis represent these values.

The tool gives an answer to the question: "How often does a given loudness (vertical scale, in dB) appear in the whole file?". The percentage figure is relative to other peaks. Some simple examples:

- 1. If, for instance, you have a sine of 0 dB / 2 second followed by a sine of -6 dB / 2 seconds. This means the global audio material has as much 0 dB material as - 6dB material: you will see 2 peaks (0 / -6 dB), each at 100%.
- 2. If, for instance, uou have a sine of 0 dB / 1 second followed by a sine of -6 dB / 3 seconds. This means there are 3 times more -6 dB material than 0 dB material. You will see a peak of about 33% at 0 dB, and another peak of 100% at -6 dB.

This analysis is useful for learning about how the loudness of your music is distributed.

You can access this dialog in the Wave File Workspace via **Analysis** > **Loudness distribution...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

5.3 Metering

WaveLab contains a variety of audio meters to help in monitoring and analyzing audio.Meters can be used to monitor audio during playback, rendering, recording and also to analyze a specific selection of audio. There are seven different audio meters in WaveLab, each with its own separate window. The meters are accessed via the Meters menu, the Shared Tool windows menu, or from the Meters control bar.

Monitor Modes

You can choose the audio source and the mode for displaying information in the meters. You can access the following monitoring functions from the Meters menu or via the Meters command bar:

- **Monitor Playback** - the meters display the audio output of the Master Section **after the dithering section, unlike the Master Section's own meter**.
- **Monitor Audio Input** - the meters display the audio input you have chosen in the Audio Streaming Settings. This is useful for monitoring audio while recording.
- **Monitor File Rendering** - the meters display what is being written to disk during file rendering, taking into account Master Section settings, with average, minimum and maximum peak values computed. After rendering, the meter(s) freeze, and remain frozen until you refresh or change the monitor mode.

- **Monitor Edit cursor position** - the meters display static information about the audio beneath the edit cursor. Note that the Master Section settings are not taken into account in this mode.

- **Analyze audio selection** - this allows you to make a selection and have the meters display the average values over the selected range as a static display. The Master Section settings are not taken into account in this mode. When you change the selection, you need to update the meter displays by selecting "Refresh selection analysis" from the Meters menu (or by clicking the Refresh button on the Meters control bar).

If you have one of the continuous display modes chosen, you can also select to freeze the meters at any time using the "Freeze meters" command.

Using meter windows

There can only be one instance of each Audio Meter. For example, if you dock one meter in a Workspace, it will be automatically removed from the place where it was previously located.

Audio Meters can appear in the Audio File, Audio Montage Workspaces as well as the Control Window. They can be used:

- as a docked window in a Workspace
- as a tabbed window in the control window
- as an independent floating window. In this mode, it can be useful to make the window "frameless" from **Window** > **Hide frame**, to save screen space. In this case, the whole menu is accessed by right-clicking.

The axis of most Audio Meters may be rotated, to view the graphics horizontally or vertically.

Some meters can also have their visual style and display parameters customized via their corresponding settings dialog. This can be accessed through the **Functions** > **Settings...** menu in each meter window, or by using the icon.

Related topics

VU Meter

Spectroscope

Oscilloscope

Bit Meter

Phasescope

Spectrometer

Wavescope

5.3.1 VU Meter

The vu-meter is used to display the peak and average loudness/decibel level of your Audio File. It also displays the pan, or the balance between the left and right channels in a stereo

file.

The upper part of the level/pan meter window shows the peak level and average loudness, in the following way:

- The Peak Level meters display the peak levels of each channel, graphically and numerically. By default, the meter segments and numerical peak values are displayed in green for low levels, yellow for levels between -6dB and -2dB, and red for levels above -2dB. You can change both the colors and the range boundaries if you wish via the Level/Pan Meter Settings dialog. This can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

- The VU (Volume Unit) meters measure the average loudness (RMS) of each channel. These meters have a built-in inertia, evening out loudness variations over a user-defined time span. If you are monitoring playback or audio input, you will also note two vertical lines following each VU meter bar, seemingly "trying to reach" the current RMS value. These lines indicate the average of the most recent minimum RMS values (left line) and the average of the most recent maximum RMS values (right line). To the left, the difference between the minimum and maximum average values is displayed (the level value in brackets) - this gives you an overview of the dynamic range of the audio material.

- If you are monitoring real-time audio (playback or input), the maximum peak and loudness values are displayed numerically to the right of the meter bars. Numbers in brackets to the right of the Maximum Peak values indicate the number of successive clips (0dB signal peaks). Recording levels should be set so that they only rarely clip. If the master level is set too high, the sound quality and frequency response will be compromised at high recording levels, with unwanted clipping effects. If the level is set too low, noise levels may be high relative to the main sound being recorded.

Pan meters

The lower part of the window shows the pan (the difference in level between the left and right channel, only applicable when monitoring stereo audio):

- The upper pan meters show the peak level difference between the channels, graphically and numerically. Note that the pan meters are "two-sided"; the level bars can go to the left or right, indicating which channel is the loudest. The two sides are shown in different colors (which can be changed via the Settings dialog as explained previously).

- The lower pan meters show the average difference in loudness between the channels, in a similar way. This gives you a visual indication of whether a stereo recording is properly centered, for example.

- If you are monitoring real-time audio (playback or input), the maximum balance difference value (peak and loudness) for each channel is displayed numerically to the left and right of the meter bars. The VU meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces.It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

5.3.2 Spectroscope

The Spectroscope shows a continuous graphical representation of the frequency spectrum, analyzed into 60 separate frequency bands, represented as vertical bars. Peak levels are shown as a short horizontal lines above the corresponding band, indicating recent peak/-maximum values. The Spectroscope offers a quick spectrum overview. For a more detailed analysis of the audio spectrum, use the Spectrometer. The Spectroscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

Spectrometer

5.3.3 Oscilloscope

The Oscilloscope offers a highly magnified view of the waveform around the playback cursor position.

If you are analyzing a stereo file, the Oscilloscope normally shows the separate levels of the two channels. However, if you activate the option "Show Sum and Subtraction" on the Functions menu (or click the +/- icon), the upper half of the Oscilloscope shows the sum of the two channels and the lower half shows the subtraction.

Making settings

By opening the Settings dialog you can adjust the display colors, and choose to activate Auto-zoom or not. With Auto-zoom activated, the display is optimized so that the highest level reaches the top of the display at all times. The Settings dialog can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

The Oscilloscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

5.3.4 Bit Meter

The Bit Meter shows the resolution or the number of bits used in the digital audio being monitored. Normally the maximum number of bits in an Audio File is the same as the resolution of the Audio File (e.g. a 16 bit Audio File shows that up to 16 bits are used), but sometimes this is not the case.

As soon as you perform any kind of real-time processing on an Audio File, the audio data is treated at a much higher resolution (32 bit floating point), to allow for pristine audio quality. Examples of such processing include level adjustments, effects, the mixing of two or more files, etc. In fact, the only time when a 16 bit file is played back at 16 bit resolution is if you play it without any fades or effects, and with the Master Faders set to 0.00 (no level adjustment). You can try this out for yourself by playing back a 16 bit Audio File and viewing the Bit Meter - as soon as you adjust the Master Faders, note that 24 bits are used and the "inter" indicator is lit (more on this below).

How to read the Bit Meter

- The innermost meters (closest to the bit scale) show the number of bits in use. You can adjust this display in the Settings dialog.
- The outer meters are "history" meters, showing how many bits were recently in use. You can adjust the hold time in the Settings dialog.
- The "over" segment indicates clipping, similar to a clip indicator.
- If the "below" segment is lit, there are more than 24 bits. The bit meter will show the 24 higher bits, and the "below" segment indicates the existence of extra, lower bits. Note that audio is always processed with more than 24 bits internally.
- If the "inter" segment is lit, this indicates that the audio data cannot really be expressed on a regular 24 bit scale (there are floating point values "in between" bits, thus the label "inter"). This is typically the case if you apply effects; thus the "inter" segment allows you to differentiate processed from non-processed 24 bit PCM files.

When to use the Bit Meter

- To check whether dithering is necessary or not. As a rule, if you're playing back or mixing down to 16 bits, and the Bit Meter shows that more than 16 bits are used, you should apply dithering.
- To see the "actual" resolution of an Audio File. For example, even though a file is in 24 bit format, only 16 bits may be used. Or, a 32 bit file may only use 24 bits (in this case, the "below" segment would not be lit). For this purpose, the Bit Meter is best used in "Analyze selection" mode.
- To see whether a "zeroed" plug-in still affects your signal, or whether a plug-in uses 16 bit internal processing, or more generally, to detect any signal modifier stage in the audio chain between reading and playing.

To adjust the Bit Meter settings, select "Settings..." from the Options pop-up menu or by using the icon.

The Bit meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Dithering

5.3.5 Phasescope

The Phasescope indicates the phase and amplitude relationship between two stereo channels. It is only really relevant when monitoring stereo audio material.

Reading the Phasescope

It can be interpreted as follows:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.
- A random but roughly elliptical shape indicates a well balanced stereo signal. If the shape "leans" to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the Phasescope will show a straight line, angled 45 degrees to the other side).
- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90 degrees on the other.
- Generally, the more you can see a "thread" shape, the more bass there is in the signal; the more "spray-like" the display, the more high frequencies are in the signal.

Phase Correlation meter

At the bottom of the display, you will find a Phase Correlation meter, which shows the same information but in a different way:

- The green line shows the current phase correlation, while the two red lines show the recent peak minimum and maximum values, respectively (these are the default colors, which you can change).
- With a mono signal, the meter would show +1, indicating that both channels are perfectly in phase.
- Similarly, -1 indicates that the two channels are the same, but one is inverted.
- Generally, for a "good" mix, the meter should show a value between 0 and +1.

Unlike the main Phasescope, the Phase Correlation meter is also available in "Analyze audio selection" mode, showing an average value for the selected range.

Changing settings

You can define display colors, peak hold time, and the resolution or number of samples to display from the **Functions** > **Settings...** menu.

The Phasescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

5.3.6 Spectrometer

The Spectrometer uses FFT (Fast Fourier Transform) techniques to display a continuous frequency graph, providing a very precise and detailed real-time frequency analysis.

- The current frequency spectrum is shown as a linear graph.
- Spectrum "peaks" are shown as a short horizontal lines, indicating recent peak/maximum values.

Snapshots

By using the "Add snapshot" and "Erase last snapshot" buttons, you can take and erase snapshots of the current spectrum. These will be superimposed over the current peak spectrum graph, in a color you can customize, until you click the icon again to take a new snapshot. One use for this is to check the effects of adding EQ, for example. Up to five snapshots can be superimposed on the display; the sixth snapshot replaces the earliest one in the display, and so on. (Don't confuse the order of snapshots with the numbered buttons - these are Spectrometer presets, see below.)

Zooming

You can adjust the frequency scale and range in the Settings dialog as described below, but it's also possible to temporarily zoom in on a frequency area of interest. This is done by clicking and dragging a rectangle in the Spectroscope display. When you release the mouse button, the display is zoomed in so that the enclosed frequency range fills the window. To return to full-scale display, select "Zoom-out fully" from the Functions menu, or double-click anywhere on the display.

Making settings

You can adjust the behavior and display of the meters as desired, and assign up to five sets of Spectrometer settings to the Preset buttons, for instant access. Open the Settings dialog by selecting "Settings" from the Functions menu or clicking the "tool" icon. Note that you can apply your settings without closing the dialog, by clicking the Apply button.

If you want to store your settings for later use (or assign them to a Preset button), select "Save as..." from the pop-up menu in the lower part of the dialog, and specify a name for the preset in the file dialog that appears. Now, you can choose to make the settings instantly available for selection in the FFT Meter window, by using the "Assign to preset button" submenu on the pop-up menu.

- When you are finished, click OK to close the dialog.

Selecting Spectrometer presets

If you have assigned your settings to the Preset buttons in the Settings dialog, you can quickly switch between different level scales and display modes, by clicking one of the Preset icons [1]-[5], or selecting the desired preset from the Options pop-up menu.

Exporting FFT data as ASCII text

When using the Spectrometer in off-line mode ("Monitor Edit cursor Position" or "Analyze audio election" mode) you can export the displayed FFT data as a text file, by selecting "Export FFT data as ASCII" from the Options pop-up menu. The resulting text file can then be imported into applications that allow graph plotting from text files (Microsoft Excel, for example).

The Spectrometer can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Spectroscope

5.3.7 Wavescope

The Wavescope meter displays a real-time waveform drawing of the audio signal being monitored.

It can be useful when recording or rendering a file if "Monitor File rendering" mode is active.

Making settings

You can adjust settings for the display via the Wavescope Settings dialog. This can be accessed via **Functions** > **Settings...** menu, or by using the icon. Here you can set various color options for the background, grid and waveform display, as well as setting the waveform rendering speed and vertical zoom. If "Clear waveform when reaching right of pane" is checked the waveform display is cleared each time the cursor reaches the right end of the display. If unchecked, the previous waveform is overwritten.

Tip: Wavescope is a useful meter for visualizing audio during recording. The Wavescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces.

It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

5.4 Transport controls

The transport controls allow you to control playback within an Audio File or montage, and to open the record window. It can be accessed via the Transport menu or the Transport toolbar. See Command bars for more information.

The Transport menu gives access to (and shows keyboard shortcuts to) the same functions that are available in the transport toolbar.

The Transport toolbar gives quick access to most of these functions. If the toolbar is hidden, select **Workspace** > **Command bars** > **Transport commands**. Transport functions operate on the current Audio File.

The basic transport functions are:

- **Loop On/Off**
  - Toggles looping on/off for the currently selected Audio File or montage.
- **Start of file / End of File**
  - Move the cursor to the beginning or end of the currently selected Audio File.
- **Skip forward / Skip backward**
  - Skip the cursor forwards or backwards within the currently selected Audio File.
- **Play**
  - Begin playback of the current Audio File.
- **Stop**
  - Stop playback of the current Audio File. A second click moves the cursor to the beginning of the last start position. A further click returns the cursor to the start of the file.
- **Record**
  - Opens the recording window from which you can start recording.

Additional transport functions such as start, skip and stopping playback options are also available from the transport command bar. In WaveLab additional playback speed and jog/shuttle shortcuts are also provided.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Playback speed

Recording Dialog

Command bars

5.5 Specific tool windows

Specific tool windows are windows that are specific to the current Workspace. They can perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Audio File Workspace specific tool windows

The following specific tool windows are available within the Audio File Workspace:

Basic Audio CD

Error detection and correction

File Browser

Marker Window

Master Project

Metadata window

Sample attributes

Scripting window (Audio File Workspace)

Spectrum editor

5.5.1 Basic Audio CD

This specific tool window allows you to burn basic "Red Book compatible" audio CDs. You create your audio CD by adding Audio Files to create a list of tracks. Each track contains a reference to the external Audio File. This means that you can save your Basic Audio CD layout as its own session and continue editing individual tracks, for example. Once you are happy with your CD layout you can choose to check the cd for conformity to the Red Book standard, burn the CD or export it to the Audio Montage Workspace to edit further. You can also choose to consolidate the Audio Files in the CD into a single Audio File with track markers.

Adding tracks to your audio CD

To create tracks in your audio CD:

- **Drag Audio Files directly** - drag Audio Files directly from your computer's file browser into the window. You can also drag Audio Files from the Audio File Workspace by their

tab or document icon. This creates CD track markers at the file edges, quantized on CD frame boundaries.

- **Drag an audio selection** - dragging a selection (rather than a whole file) into the basic audio CD creates a CD track with markers at the edges, but quantized on CD frame boundaries, not necessarily exactly on the exact position of the selection edges.

- **Browse for Audio Files** - select **File** > **Add track(s)...** from the Basic Audio CD menu. You can then select Audio Files to insert in your CD.

Once you have added your tracks, information about each track is displayed. Each track also has its marker information nested within it. To access this information click on the left most arrow icon to expand the track view.

Using the track list

Once you have added a number of tracks to your audio CD you can use the track list to:

- **Audition playback** - you can audition each track, with or without pre-roll, by clicking on the leftmost icons. Click on in the "audition" column to play the track from that track's start marker with a pre-roll. Click on the right hand button to playback the track exactly from the start marker position. You will notice a progress display appears at the bottom of the window to show you the playback position/time and track number. You can also start playback with or without a pre-roll from the tracks start or end markers using the same controls. If the markers are not visible click on the track expand arrow to reveal the track's markers.

- **Edit a track's start pause time** - for all tracks except the first one you can adjust the pause duration before the track begins playing. If the pause checkbox is not visible click on the track expand arrow to reveal the track's markers and pause time setting. Double click in the cell to edit the value.

- **Add ISRC code** - add an ISRC code for a track by Select-clicking on the field under the ISRC column.

- **Edit a track name** - by Select-clicking on the name in the list. This also renames the track's start marker labels.

- **Add a comment** - add a comment to each track by Select-clicking to edit its value.

- **Editing track times** - you cannot edit track start and end times directly in the track listing. Instead, open the tracks accompanying Audio File via **Edit** > **Edit Audio** and adjust the track start and track end marker locations in the main waveform display. The start and end times will update accordingly.

Completing your Audio CD

In addition to editing via the track window you can also set overall CD settings via the Basic Audio CD Settings dialog. You can use this dialog to add UPC/EAN codes and adjust

pauses and silences in your CD. The Master Section is never used to process the signal to burn, unlike in the montage. This is why playback through the Master Section is an option that is off by default.

When you are happy with your CD layout or you wish to further refine it further you can:

- **Check CD conformity** - before burning your CD you can first check its Red Book conformity by choosing **Edit** > **Check CD conformity**.
- **Write audio CD** - choose to burn the CD using the Write Audio CD dialog.
- **Convert to Audio Montage** - convert the CD layout into an audio montage for further editing by choosing **Edit** > **Convert to Audio Montage**

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Authoring CDs and DVDs

Basic Audio CD Settings

Write Audio CD

Select-clicking

5.5.2 Error detection and correction

This specific tool window allows you to search for unwanted clicks and digital artifacts in an Audio File. Several detection and restoration methods are available. You can detect, mark and name, jump to, play back and remove individual audio errors.

The settings for error detection and correction are organized within the following groups of controls:

- **Detection tab** - choose the method used to detect any errors. Each different method has its own settings to determine when it will detect an error. Click Detection 1 and 2 will search for clicks in certain frequency ranges while Digital Click Detection looks for clicks that are caused by clicks typical to digital errors.
- **Correction tab** - specify the method used to correct any errors.
- **Search Area tab** - specify the range of audio you wish to search for errors.
- **Options tab** - this provides a range of preferences for playing back, viewing, and marking any errors found.
- **Browse & Correct panel** - step through your defined search region detecting each error in turn. You can choose to correct any errors or mark them for later correction. There are controls to skip between error markers and make fine adjustments to the selection. You can also automatically detect and correct all marked errors in the search region.

Strategies to detect and correct errors

There are several strategies that you can employ when detecting and correcting errors:

- Define an audio selection where you have identified an error, then click "Correct", or "Mark for subsequent correction".
- Use the function "Detect next error" to let WaveLab automatically find the next error, and then click "Correct", or "Mark for subsequent correction".
- Use the function "Detect all errors" to let WaveLab automatically find all errors in the predefined range. Then browse the marked errors, removing or adjusting each audio range in turn, and click "Correct" to correct any specific errors, or you can use the option "Correct all".

You can access this dialog by choosing **Workspace** > **Specific tool windows** > **Error Correction**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Marker types

5.5.3 File Browser

This dialog allows you to browse files directly from within WaveLab rather than using your operating system's file browser.

It provides you with all the standard browsing functions (such as List and Icon views)as well as additional controls to audition Audio Files and any marker defined regions.You can use it to open or insert an entire file or a specific region of a file by dragging and releasing the file in the location you wish to insert the file. In the Audio Montage Workspace you can also choose to only view certain types of WaveLab specific files. The File Browser can be very useful in speeding up the process of auditioning a long list of sound files. It can also be used to drag to windows such as Data CD/DVD, including dragging folders.

The playback functions

Play file Once you have selected a file in the File Browser list, you can audition it by clicking Play in the File Browser's toolbar. Click again to stop.

Auto-play You can also have files play back automatically as soon as they are selected. To do this, activate Auto-Play mode.

Tip: To audition a long list of Audio Files, enable Auto-play and use the UP and DOWN cursor keys to step through each Audio File in quick succession.

Opening and inserting a file or region

Once you have chosen a file you wish to open, double-click it to open the file in the current Workspace. You can also open it by dragging it onto an empty tabbed group, or onto the tab bar of a tabbed group.

You can also drag the file over a wave view to **insert** it at a given point. When you drag a file over a wave view, a beam is displayed so as to clearly display the insertion point.

When browsing Audio Files, the file browser also has a region panel. When you select an Audio File, its related regions are listed in this panel. You can then drag the region from the list, to insert only that part of the audio. Double-clicking on a region has the effect of opening the file in the Workspace and selecting that region. You can access this dialog via **Workspace** > **Specific tool windows** > **File Browser**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

5.5.4 Marker Window

The marker window is a specific tool window that allows you to create, edit and use markers while working on an audio waveform.

From the Functions menu you can also access useful functions to convert marker types, rename multiple markers and export the marker list as text.

The Marker list

The marker window contains a list of any markers placed in the currently active file along with their corresponding details. Click on a column header to sort the list in ascending or descending order based on the values in that column. You can also use the Filter menu to toggle which types of markers are displayed in the list.

From this list of markers you can use the fields, controls and menus to:

- **Create a new marker at the playback head position** - click on a marker icon (at the top of the window) or choose a marker type from the Insert menu to drop a marker at the current playhead location.

- **Create a pair of markers from a selection** - make a selection in the waveform and then click on a pair of marker icons (at the top of the window) or choose a "Create region" command from the Insert menu to drop a pair of markers at either side of the current selection.

- **Start playback** - It is possible to start playback at the selected marker position, with or without pre-roll, by clicking on the leftmost icons. Click on in the "audition" column to play the wave from that marker position with a pre-roll. Click on the right hand button to play the wave exactly from the marker position.

- **Change marker types** - You can change a marker type by (left) clicking on its icon and choosing another marker type from the pop-up list that appears.

- **Edit a marker's name/time/comments** - double click in a cell to edit a value (to align the marker to an exact time, for instance). You can edit the name of the marker, its time position and any comments.

- **Lock a marker's position** - select the checkbox to lock the marker. This prevents it from being accidentally dragged to a new position in the Waveform window.

For more information on using markers see Markers. For details on the different marker types and their uses see Marker types. Note that there are more marker features available in the Montage version compared to the Audio File version. See "Using markers in the Audio Montage Workspace" in Markers for more details.

You can access this specific tool window within the Audio Files and Audio Montage Workspaces via **Workspace** > **Specific tool windows** > **Markers**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Markers

Marker types

Convert marker type

Text format

5.5.5 Metadata window

This specific tool window displays any Meta-data tags belonging to the currently active file.

This usually consists of a set of tags that describe the audio contents. For example it might include the title of the track, the author, and the date it was created. This data will vary depending on the file type, and not all file types store this information so it may be blank. To edit the file's meta-data tags click anywhere in the window. This will open the corresponding file attributes dialog.

You can access this dialog in the Audio File Workspace via **Workspace** > **Specific tool windows** > **Meta-Data**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

5.5.6 Sample attributes

This dialog allows you to define settings for an audio sample before loading it into a hardware or software sampler.

The settings do not process the sample in any way, they just give the file properties that the receiving sampler can use. This includes information about the pitch of the sample (which can be detected automatically by WaveLab), the key range that the sample should span, and the velocity range to occupy. For WAV and AIFF files, this information is stored in the file's header.

By default, there are no sample attributes in an Audio File. Therefore you must explicitly create this information by clicking on the "Create" button first. This dialog is useful if your sampler can take advantage of these additional tags. If supported by your sampler, it can save you time by allowing you to both edit and define a samples properties from within WaveLab.

You can access this dialog via **Workspace** > **Specific tool windows** > **Sample Attributes...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

5.5.7 Scripting window (Audio File Workspace)

This window allows you to write and execute scripts in the Audio File Workspace.

The inbuilt text editor helps you when writing scripts by highlighting the different parts of the script with colors, making it more readable. A script can also be written in another text editor, and loaded via the File menu. To run a script, from within the script window choose **Functions** > **Execute script**.

For a general description of scripting, see Scripting. For an overview of the scripting language, see ECMAScript Reference.

You can access this dialog from the Audio File Workspace via **Workspace** > **Specific tool windows** > **Script**.

Note: To view trace or log messages when executing scripts that contain the logWindow() function, ensure that the Log window is visible and that it's warning filter buttons - ! ⚠ ⓘ are selected.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Scripting

ECMAScript Reference

Log window

5.5.8 Spectrum editor

This specific tool window allows you to select and perform operations on an Audio File via its audio spectrum.

It uses high quality linear-phase filters to process a spectrum selection both in the time domain and in the frequency domain. This can be useful for advanced audio restoration tasks like removing noise from a specific source in a recording, for example.

Making and editing selections

Spectrum editing can only be performed when a waveform is first displayed in Spectrum display mode and when a spectrum selection is defined. To activate the Spectrum editor, click on the Spectrum edit tool in the command bar (or via **Workspace** > **Specific tool windows** > **Spectrum Editor**). This displays the Spectrogram if this is not already the case (see The Spectrum display).

To select a region, point and drag to define a rectangle. The region selection defines a time and a specific frequency range. This allows you to edit and process audio both in the time domain and in a specific frequency domain, as opposed to standard wave editing which always operates in the full frequency domain.

Notes on spectrum selections:

- When defining a region in a stereo file, a "mirrored" region is automatically created in the other channel. To define a region in only one channel, press **[Shift]** while drawing the region.
- Once you have a region selected, if you place your mouse cursor inside the region, a pop-up appears showing the currently set frequency range (Hz) and the time range (seconds/milliseconds) for the selection.
- A selected region can be moved in any direction by clicking and dragging the region with the cursor.
- If you press **[Shift]** whilst dragging a region, it only moves horizontally thus retaining your selected frequency range. If you press **[Ctrl]/[Command]** + **[Shift]** the region only moves vertically thus retaining the selected time range.
- You can resize a region by placing the mouse cursor at the edges of the region (so that a double-arrow appears) and then clicking and dragging.
- To remove a region that you have drawn, click anywhere else on the waveform and it disappears.
- If you have defined a region as a Source or Target, it can be selected at any time to create a new selection using the same dimensions.

Spectrum editor functions

The Spectrum Editor has the following parts:

- **Selection** - use this part to refine and define your selection(s). You can expand selections, shift selections around and define a source and target region for performing copy operations.

- **Operations** - use this part to perform copying, filtering and processing operations. For copying operations, you require both a source and a target region to have been defined using the Selection tab. You can choose which method to copy audio between the Source and Target regions. You can also choose to apply processing to a single spectrum region using a variety of different modes. See Spectrum Processing Modes for information on each mode type.

- **Master Section** - use this part to process and route selected spectral audio regions through the Master Section and its effects plug-ins. This allows for frequency-selective processing. A selected region's frequency spectrum can be routed to the Master Section where you can choose to process it separately from the non-selected frequency spectrum. The signal is split so that one part (region spectrum or non-selected spectrum) is sent to the plug-ins, while the other part can be mixed with this processed signal, after the Master Section output.

Using the Spectrum editor

You can use the Spectrum editor in two main operational modes:

Region copying and filtering (Operations tab)

This is mainly intended for audio restoration applied over brief time ranges. Selections can be copied, pasted, and filtered. This type of processing is mainly used to reduce, remove or replace unwanted sound artifacts in the audio material, and can be carried out with great precision. This could be useful for example, to replace part of a live recording that contains an unwanted noise (such as a mobile phone ring tone), with a copy of a similar region of the spectrum that only contains a "clean" signal. In general, the spectral Copy/Paste combination gives the best results, given that the source and destination regions are properly choosen.

Master Section processing (Master Section tab)

This allows you to process a specific frequency range via the Master Section. The selected or non-selected regions of the spectrum can be processed differently. You can also employ a number of filters (Bandpass/Low-pass/High-pass) to further refine the particular range of frequencies to be affected by any Master Section effects.

A selected spectrum region can either be:

- Processed separately by the Master Section plug-ins. The non-selected spectrum can either be bypassed or sent to the Master Section.

- Bypassed. This removes the selected spectrum region from the Audio File. The non-selected spectrum can either be routed to the Master Section input or the Master Section output.

- Sent to Master Section output. The non-selected spectrum can either be bypassed or sent to the Master Section input. If the latter is selected, it will be mixed with the selected spectrum region at the Master Section output.

You can access this window by choosing **Workspace** > **Specific tool windows** > **Spectrum Editor**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

The Spectrum display

Spectrum Processing Modes

Spectrogram options

Spectrometer

5.5.8.1 Spectrum Processing Modes

When applying processing to a spectrum region in the Spectrum Editor you can use a variety of different modes:

Damp

This is used to attenuate the level of a region. The Gain parameter specifies the level of the attenuation (it is also possible to set positive gain values, i.e. to boost the frequencies in the region). Three filter types are available to perform the damping:

- Bandpass filter: All frequencies in the region are equally attenuated.
- Low-pass filter: Higher frequencies in the region will be more attenuated.
- High-pass filter: Lower frequencies in the region will be more attenuated.

For the Low- and High-pass filters, a low Filter steepness setting is normally used (around 6 to 18dB).

Blur peaks

This filter operation analyzes the region to find the frequencies with the highest level. The level of these frequencies will be attenuated (or boosted) according to the set Gain value. If the gain is negative, these frequencies will be "blurred" and disappear in the mix as a whole. The purpose of the filter is to mask the loudest frequencies (e.g. it could be used to eliminate a sudden burst of pitched but unwanted sound in the audio material, like acoustic feedback).

This mode works best with the Infinite steepness setting, which is automatically selected when Blur peaks is used. You can, however, freely set any Filter steepness setting. The filter type options are not available when this mode is selected.

Dispersion

This is a special filter that "smears" the dynamics and pitch of the region, without changing the actual frequency content. This works best on lower frequencies, to mask the identity of a signal without changing the frequency spectrum.

This mode works best with the Infinite steepness setting, which is automatically selected when Dispersion is used. You can, however, freely set any Filter steepness setting. The filter type options are not available when this mode is selected. Dispersion can also be used to create special effects as it warps the sound in an unique way.

Fade Out

This can be used with any of the three filter types (Bandpass/Low-pass/High-pass). It gradually filters the frequencies in the region along the time axis (from nothing at the left edge of

the region, to maximum allowed by the gain parameter at the right edge), useful for gradually removing frequencies from a region.

Fade In

This works like Fade out but inversed. It can also be used in conjunction with any of the three filter types (Bandpass/Low-pass/High-pass).

Fade out then in

This is a mix of the two previous options; the filtering effect gradually fades out until the middle of the region, then fades back in. It can also be used in conjunction with any of the three filter types (Bandpass/Low-pass/High-pass).

Fade in then out

The inverse of the above. It can also be used in conjunction with any of the three filter types (Bandpass/Low-pass/High-pass).

Related topics

The Spectrum display

Spectrogram options

Spectrum editor

5.6 Shared tool windows

Shared tool windows are windows that can only exist in one place at a time and are shared between Workspaces. They perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Shared tool windows:

Phasescope

VU Meter

Oscilloscope

Bit Meter

Spectroscope

Spectrometer

Wavescope

Timecode

Background task monitor

Log window

Master Section

Related topics

About Tool Windows

Metering

Docking windows

5.6.1 Phasescope

The Phasescope indicates the phase and amplitude relationship between two stereo channels. It is only really relevant when monitoring stereo audio material.

Reading the Phasescope

It can be interpreted as follows:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.
- A random but roughly elliptical shape indicates a well balanced stereo signal. If the shape "leans" to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the Phasescope will show a straight line, angled 45 degrees to the other side).
- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90 degrees on the other.
- Generally, the more you can see a "thread" shape, the more bass there is in the signal; the more "spray-like" the display, the more high frequencies are in the signal.

Phase Correlation meter

At the bottom of the display, you will find a Phase Correlation meter, which shows the same information but in a different way:

- The green line shows the current phase correlation, while the two red lines show the recent peak minimum and maximum values, respectively (these are the default colors, which you can change).
- With a mono signal, the meter would show +1, indicating that both channels are perfectly in phase.
- Similarly, -1 indicates that the two channels are the same, but one is inverted.
- Generally, for a "good" mix, the meter should show a value between 0 and +1.

Unlike the main Phasescope, the Phase Correlation meter is also available in "Analyze audio selection" mode, showing an average value for the selected range.

Changing settings

You can define display colors, peak hold time, and the resolution or number of samples to display from the **Functions** > **Settings...** menu.

The Phasescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

5.6.2 VU Meter

The vu-meter is used to display the peak and average loudness/decibel level of your Audio File. It also displays the pan, or the balance between the left and right channels in a stereo file.

The upper part of the level/pan meter window shows the peak level and average loudness, in the following way:

- The Peak Level meters display the peak levels of each channel, graphically and numerically. By default, the meter segments and numerical peak values are displayed in green for low levels, yellow for levels between -6dB and -2dB, and red for levels above -2dB. You can change both the colors and the range boundaries if you wish via the Level/Pan Meter Settings dialog. This can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

- The VU (Volume Unit) meters measure the average loudness (RMS) of each channel. These meters have a built-in inertia, evening out loudness variations over a user-defined time span. If you are monitoring playback or audio input, you will also note two vertical lines following each VU meter bar, seemingly "trying to reach" the current RMS value. These lines indicate the average of the most recent minimum RMS values (left line) and the average of the most recent maximum RMS values (right line). To the left, the difference between the minimum and maximum average values is displayed (the level value in brackets) - this gives you an overview of the dynamic range of the audio material.

- If you are monitoring real-time audio (playback or input), the maximum peak and loudness values are displayed numerically to the right of the meter bars. Numbers in brackets to the right of the Maximum Peak values indicate the number of successive clips (0dB signal peaks). Recording levels should be set so that they only rarely clip. If the master level is set too high, the sound quality and frequency response will be compromised at high recording levels, with unwanted clipping effects. If the level is set too low, noise levels may be high relative to the main sound being recorded.

Pan meters

The lower part of the window shows the pan (the difference in level between the left and right channel, only applicable when monitoring stereo audio):

- The upper pan meters show the peak level difference between the channels, graphically and numerically. Note that the pan meters are "two-sided"; the level bars can go to the left or right, indicating which channel is the loudest. The two sides are shown in different colors (which can be changed via the Settings dialog as explained previously).

- The lower pan meters show the average difference in loudness between the channels, in a similar way. This gives you a visual indication of whether a stereo recording is properly centered, for example.

- If you are monitoring real-time audio (playback or input), the maximum balance difference value (peak and loudness) for each channel is displayed numerically to the left and right of the meter bars. The VU meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces.It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

5.6.3 Oscilloscope

The Oscilloscope offers a highly magnified view of the waveform around the playback cursor position.

If you are analyzing a stereo file, the Oscilloscope normally shows the separate levels of the two channels. However, if you activate the option "Show Sum and Subtraction" on the Functions menu (or click the +/- icon), the upper half of the Oscilloscope shows the sum of the two channels and the lower half shows the subtraction.

Making settings

By opening the Settings dialog you can adjust the display colors, and choose to activate Auto-zoom or not. With Auto-zoom activated, the display is optimized so that the highest level reaches the top of the display at all times. The Settings dialog can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

The Oscilloscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

5.6.4 Bit Meter

The Bit Meter shows the resolution or the number of bits used in the digital audio being monitored. Normally the maximum number of bits in an Audio File is the same as the resolution

of the Audio File (e.g. a 16 bit Audio File shows that up to 16 bits are used), but sometimes this is not the case.

As soon as you perform any kind of real-time processing on an Audio File, the audio data is treated at a much higher resolution (32 bit floating point), to allow for pristine audio quality. Examples of such processing include level adjustments, effects, the mixing of two or more files, etc. In fact, the only time when a 16 bit file is played back at 16 bit resolution is if you play it without any fades or effects, and with the Master Faders set to 0.00 (no level adjustment). You can try this out for yourself by playing back a 16 bit Audio File and viewing the Bit Meter - as soon as you adjust the Master Faders, note that 24 bits are used and the "inter" indicator is lit (more on this below).

How to read the Bit Meter

- The innermost meters (closest to the bit scale) show the number of bits in use. You can adjust this display in the Settings dialog.
- The outer meters are "history" meters, showing how many bits were recently in use. You can adjust the hold time in the Settings dialog.
- The "over" segment indicates clipping, similar to a clip indicator.
- If the "below" segment is lit, there are more than 24 bits. The bit meter will show the 24 higher bits, and the "below" segment indicates the existence of extra, lower bits. Note that audio is always processed with more than 24 bits internally.
- If the "inter" segment is lit, this indicates that the audio data cannot really be expressed on a regular 24 bit scale (there are floating point values "in between" bits, thus the label "inter"). This is typically the case if you apply effects; thus the "inter" segment allows you to differentiate processed from non-processed 24 bit PCM files.

When to use the Bit Meter

- To check whether dithering is necessary or not. As a rule, if you're playing back or mixing down to 16 bits, and the Bit Meter shows that more than 16 bits are used, you should apply dithering.
- To see the "actual" resolution of an Audio File. For example, even though a file is in 24 bit format, only 16 bits may be used. Or, a 32 bit file may only use 24 bits (in this case, the "below" segment would not be lit). For this purpose, the Bit Meter is best used in "Analyze selection" mode.
- To see whether a "zeroed" plug-in still affects your signal, or whether a plug-in uses 16 bit internal processing, or more generally, to detect any signal modifier stage in the audio chain between reading and playing.

To adjust the Bit Meter settings, select "Settings..." from the Options pop-up menu or by using the icon.

The Bit meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Dithering

5.6.5 Spectroscope

The Spectroscope shows a continuous graphical representation of the frequency spectrum, analyzed into 60 separate frequency bands, represented as vertical bars. Peak levels are shown as a short horizontal lines above the corresponding band, indicating recent peak/-maximum values. The Spectroscope offers a quick spectrum overview. For a more detailed analysis of the audio spectrum, use the Spectrometer. The Spectroscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

Spectrometer

5.6.6 Spectrometer

The Spectrometer uses FFT (Fast Fourier Transform) techniques to display a continuous frequency graph, providing a very precise and detailed real-time frequency analysis.

- The current frequency spectrum is shown as a linear graph.
- Spectrum "peaks" are shown as a short horizontal lines, indicating recent peak/maximum values.

Snapshots

By using the "Add snapshot" and "Erase last snapshot" buttons, you can take and erase snapshots of the current spectrum. These will be superimposed over the current peak spectrum graph, in a color you can customize, until you click the icon again to take a new snapshot. One use for this is to check the effects of adding EQ, for example. Up to five snapshots can be superimposed on the display; the sixth snapshot replaces the earliest one in the display, and so on. (Don't confuse the order of snapshots with the numbered buttons - these are Spectrometer presets, see below.)

Zooming

You can adjust the frequency scale and range in the Settings dialog as described below, but it's also possible to temporarily zoom in on a frequency area of interest. This is done by clicking and dragging a rectangle in the Spectroscope display. When you release the mouse button, the display is zoomed in so that the enclosed frequency range fills the window. To

return to full-scale display, select "Zoom-out fully" from the Functions menu, or double-click anywhere on the display.

Making settings

You can adjust the behavior and display of the meters as desired, and assign up to five sets of Spectrometer settings to the Preset buttons, for instant access. Open the Settings dialog by selecting "Settings" from the Functions menu or clicking the "tool" icon. Note that you can apply your settings without closing the dialog, by clicking the Apply button.

If you want to store your settings for later use (or assign them to a Preset button), select "Save as..." from the pop-up menu in the lower part of the dialog, and specify a name for the preset in the file dialog that appears. Now, you can choose to make the settings instantly available for selection in the FFT Meter window, by using the "Assign to preset button" submenu on the pop-up menu.

- When you are finished, click OK to close the dialog.

Selecting Spectrometer presets

If you have assigned your settings to the Preset buttons in the Settings dialog, you can quickly switch between different level scales and display modes, by clicking one of the Preset icons [1]-[5], or selecting the desired preset from the Options pop-up menu.

Exporting FFT data as ASCII text

When using the Spectrometer in off-line mode ("Monitor Edit cursor Position" or "Analyze audio election" mode) you can export the displayed FFT data as a text file, by selecting "Export FFT data as ASCII" from the Options pop-up menu. The resulting text file can then be imported into applications that allow graph plotting from text files (Microsoft Excel, for example).

The Spectrometer can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Spectroscope

5.6.7 Wavescope

The Wavescope meter displays a real-time waveform drawing of the audio signal being monitored.

It can be useful when recording or rendering a file if "Monitor File rendering" mode is active.

Making settings

You can adjust settings for the display via the Wavescope Settings dialog. This can be accessed via **Functions** > **Settings...** menu, or by using the icon. Here you can set various color options for the background, grid and waveform display, as well as setting the waveform rendering speed and vertical zoom. If "Clear waveform when reaching right of pane" is checked the waveform display is cleared each time the cursor reaches the right end of the display. If unchecked, the previous waveform is overwritten.

Tip: Wavescope is a useful meter for visualizing audio during recording. The Wavescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

5.6.8 Timecode

This shared tool window displays the current time of the playback head (or the cursor position if not playing back), in the timecode format currently selected in the Time Format dialog.

The current time can be in a range of standard SMPTE formats, CD or DVD specific formats, or a custom one - see Time Format dialog.

The most common use for this window is for working with video or film. You can use it to display the playback head position of the current Audio File in a SMPTE format to frame level resolution (i.e. hours:minutes:seconds:frames). You can also use it to display CD frames (minutes:seconds:frames) when you are authoring CD or DVDs.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Timecode**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Time Format dialog

CD Frame

SMPTE timecode

Authoring CDs and DVDs

5.6.9 Background task monitor

This specific tool window allows you to view all background rendering processes that are in progress.

You can adjust the priority with which they are processed, pause or cancel them. It is useful if you have a number of lengthy processes underway and want to free up some processing power to focus on editing. You can either lower the priority of a task so it does not use as much of the computer processor's capacity, or pause the task temporarily.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Background task monitor**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window by choosing **Utilities** > **Background task monitor**.

You can choose to automatically open the Background task monitor when a rendering process begins. To enable this option, select the checkbox via **Options** > **Global Preferences** > **Options** > **Make the Background Task Monitor visible when a task starts** . Since this window is a Shared_tool_window, the location where it opens may vary.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Control Window

Rendering

5.6.10 Log window

This shared tool window allows you to view log messages that have been issued by WaveLab.

When using WaveLab's scripting language for example, the *logWindow( )* function outputs messages to this window. There are a number of toggle buttons and these allow you to filter the types of messages displayed.

Note: When using the logWindow( ) method you need to have the "Show informal notes" button selected for the messages to be visible. If no log messages are displayed, make sure all the toggle buttons are selected - ! .

You can access this window via **Workspace** > **Shared tool windows** > **Log**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window via **Utilities** > **Log**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Control Window

Specific tool windows

Scripting

5.7 Tools

In the Audio File Workspace, WaveLab provides you with powerful tools for generating audio, splitting files, importing audio from CD and working with external applications. This page contains links to these tools which can be accessed from the **Tools** menu.

Related topics

Error detection and correction

Auto Split

Signal generator

DTMF Generator

5.7.1 Error detection and correction

This specific tool window allows you to search for unwanted clicks and digital artifacts in an Audio File. Several detection and restoration methods are available. You can detect, mark and name, jump to, play back and remove individual audio errors.

The settings for error detection and correction are organized within the following groups of controls:

- **Detection tab** - choose the method used to detect any errors. Each different method has its own settings to determine when it will detect an error. Click Detection 1 and 2 will search for clicks in certain frequency ranges while Digital Click Detection looks for clicks that are caused by clicks typical to digital errors.
- **Correction tab** - specify the method used to correct any errors.
- **Search Area tab** - specify the range of audio you wish to search for errors.
- **Options tab** - this provides a range of preferences for playing back, viewing, and marking any errors found.
- **Browse & Correct panel** - step through your defined search region detecting each error in turn. You can choose to correct any errors or mark them for later correction. There are controls to skip between error markers and make fine adjustments to the selection. You can also automatically detect and correct all marked errors in the search region.

Strategies to detect and correct errors

There are several strategies that you can employ when detecting and correcting errors:

- Define an audio selection where you have identified an error, then click "Correct", or "Mark for subsequent correction".

- Use the function "Detect next error" to let WaveLab automatically find the next error, and then click "Correct", or "Mark for subsequent correction".
- Use the function "Detect all errors" to let WaveLab automatically find all errors in the predefined range. Then browse the marked errors, removing or adjusting each audio range in turn, and click "Correct" to correct any specific errors, or you can use the option "Correct all".

You can access this dialog by choosing **Workspace** > **Specific tool windows** > **Error Correction**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Marker types

5.7.2 Auto Split

This dialog allows you to automatically split an Audio File into separate Audio Files according to certain specified rules.

You can split files:

- ...at marker points
- ...at specific time intervals
- ...at regions containing silences
- ...at beat points using beat detection
- ...according to a description of regions stored in a text file

The Auto Split function can be used in many situations such as:

- cutting up a single Audio File of a recording session into its individual takes (using the silence threshold)
- cutting up a drum loop into its individual drum hit samples (using beat detection)
- outputting individual tracks from an album master file (using marker detection or "Learn regions from text file")
- converting marker types from one type to another (by using the option "Do not split, only create markers at splitting points")
- silencing the regions between audio information in an instrumental take (using "Mute audio outside regions")

The Auto Split tool is made up of several pages of dialogs which take you through the process of splitting your audio. Note that the sequence of pages is not always the same and depends on the choices you make.

You can access this dialog in the Audio File Workspace via **Tools** > **Auto Split...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Split Clip at silences

5.7.3 Signal generator

This dialog allows you to generate complex synthesized sounds in mono or stereo.

You can layer different waveform generators together and if outputting a stereo file, adjust different settings for both the left and right channels. There are a multitude of settings to adjust the character (Source tab), frequency (Frequency tab) and amplitude (Level tab) of the generated signals.

To generate a sound follow these basic steps:

- choose whether the file is to be mono or stereo, and define the number of samples, and bit depth using the Audio properties dialog.
- choose how many layers of signal generators you wish to use (up to 64).
- for each layer, use the source tab to choose the type of signal generator you wish to use, switch to the frequency tab to define a frequency and its envelope, and use the level tab to define a level envelope. If you have chosen a stereo file you can also choose to adjust left and right channels independently.
- once all settings are made, choose Generate and WaveLab calculates an Audio File according to your specifications. The file is generated and opens up in a new window.

Generating signals can be useful for a variety of testing and measurement purposes. For example, you could use the Signal Generator to create a sine wave sweep from say 20Hz to 20,000kHz to test your monitoring environment for resonant frequencies.

You can access this dialog in the Audio File Workspace via **Tools** > **Signal Generator...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio properties dialog

DTMF Generator

5.7.4 DTMF Generator

This dialog allows you to generate DTMF (Dual-tone multi-frequency) or MF tones as used by some analog telephone systems.

These signals are created by combining two sine waves with variable frequencies together. Push button telephones generate these two sine waves at different frequencies depending on the number you press. These dial pulses are then decoded by the telephone exchange to identify which letters or numbers you pressed.

The DTMF system uses eight different frequency signals transmitted in pairs to represent sixteen different numbers, symbols and letters. The characters you can use are: 0123456789ABCD∗#,)

The MF system is limited to only fifteen characters: 0123456789ABC∗#,

The DTMF Generator allows you to generate these DTMF/MF tones to create an Audio File. You can enter the string of numbers and letters, specify the durations of the tones and set the amplitude and output type.

You can access this dialog from the Audio File Workspace via **Tools** > **DTMF Generator...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Signal generator

5.8 Peak Files

A peak file is a small file with the extension ".gpk" that is automatically created by WaveLab each time a file is modified or opened in WaveLab for the first time.

The peak file contains information about the waveform and determines how it is drawn in the wave window. The purpose of the peak file created by WaveLab is to speed up the time it takes to draw a file's waveform.

By default, the peak file is stored in the same location as the Audio File whose waveform it represents. You can choose to store a peak file in another location and/or delete it when you close an Audio File. These settings are found in the Audio File editing preferences.

Chapter 6

Audio Montages

The WaveLab Audio Montage Workspace is a non-destructive editing environment that allows you to arrange, edit, play back audio Clips. Features include both track and Clip-based effects, volume and pan automation, and wide-ranging fade and cross fade functions. The Audio Montage is a great tool for creating music CDs, mastering, multimedia work, radio spot production, etc. You can create intricate multi-track compositions and author professional audio CDs. Support for Multichannel and DVD-Audio disks allows you to create surround sound mixes too.

Basic terminology

The Audio Montage window is where you assemble audio "Clips" into a montage. Once imported, you can arrange, edit and play back Clips on an unlimited number of both stereo or mono tracks. These provide a way for the user to structure the work graphically, but they should not be viewed as "virtual tape tracks".

On an audio track, you can place any number of Clips. These are "containers" for the audio, and include a number of settings and functions such as volume and pan curves, fades, etc. A Clip contains a reference to a source Audio File on your hard disk, as well as start and end positions in the file (allowing Clips to play back smaller sections of their source Audio Files). Any number of Clips can reference the same source file.

Creating a Montage

To create a blank new montage file choose **File** > **New...** to open the Audio Montage properties dialog. From this dialog you can define the mode, number of channels and sample rate to use for your new montage. The default setup is a stereo 44.1kHz montage. If you wish to change the number of channels at a later stage you can by opening this dialog again via **Edit** > **Audio Montage properties...**. See "Multi-channel surround setup" below for more information on setting up for multiple channels.

There are also ways to generate a new montage automatically. If you are working in the Audio File Workspace you can choose **Edit** > **Create Montage from Audio File...** to create a new Audio Montage with a selection or an entire Audio File. You can also create a montage by importing different file formats directly into the Audio Montage Workspace. Several of these are found in the **File** > **Import** > menu. You can create a montage `file:`

- by importing an audio DDP file as an Audio Montage.

- importing an audio CD Cue file/CD image as an Audio Montage. This function allows you to open a cue file – a text file containing information about how CD tracks correspond to the audio in an associated CD image file. The result will be a montage with Clips corresponding to the tracks in the cue sheet. Cue sheets and CD images can be created in many programs, including WaveLab.

- by importing audio CD tracks directly to a montage using the Import Audio CD dialog from the Utilities menu.

- by converting a Basic Audio CD into an audio montage from the Audio File Workspace.

- by creating a montage from an Audio File using the Auto Split function.

- by opening an Audio File in the Audio File Workspace and selecting "Create Audio Montage from Audio File" from the Edit menu.

- by importing an AES-31 project file. You can for example, export a complete Nuendo project as an AES-31 file, and import this into WaveLab, where a Montage containing all the tracks and files of the original Nuendo project is created.

- by importing an XML text file previously exported by WaveLab.

Multi-channel surround setup

In addition to the default stereo mode, WaveLab supports up to 8 channel operation and up to 6 channel surround operation. To enable multichannel operation mode, a proper audio driver must first have been selected in the Audio Streaming Settings. Once you have setup your sound card to receive multiple channels you can choose the number of output channels you wish in your Montage via the Audio Montage properties dialog. You can then route different tracks to different channels using the Track dispatching dialog and adjust the pan between different surround channels using the Surround Panner dialog or the mini surround pan window in the track's control window.

Tracks and Clips

The main montage window is divided into a number of horizontal "tracks" which you can add and remove. Tracks can be mono/stereo audio or picture tracks. You layout your Clips on audio tracks to create your montage.

Adding new tracks

You can add new tracks of any type:

- by using the **Track** > **Add [track type]** menu, or

- by clicking on a track's number button to bring up the Track context menu and choosing "Add [track type] track".

See "Managing tracks" below for more information on working with tracks.

Adding new Clips

You then import audio to your audio tracks to begin assembling them as Clips in your montage. You can create Clips by copying audio selections from the Audio File Workspace into the Audio Montage as well as importing external files directly. Add an Audio File/s to a track:

- by using drag and drop from the Audio File Workspace (via a tab or the document icon).
- by dragging an Audio File directly from your operating system's file browser.
- by copying and pasting audio from an Audio File in the Audio File Workspace and pasting it into a track.
- by dragging from the File Browser window.
- by dragging a track from the Basic Audio CD window.
- by dragging regions from the files view.

Editing a Montage

Editing/arranging Clips

Once you have imported your audio Clips you can begin laying them out and editing them in a multitude of ways. Most of the functions for editing Clips are accessible via the Focused Clip window. You can cut, copy, paste, delete, move, repeat, split, pitch shift, lock, time stretch and change the referenced Audio File for all Clips in your montage. By dragging the edges of two Clips on the same track up against other you can automatically add a cross fade. The cross fade type can be adjusted and you can also add envelope points for volume and pan. See the Focused Clip window for more information.

Shortcut: You can also access the Edit pane commands from the Focused Clip editor by right-clicking on a Clip to reveal a context menu.

Modes

The Modes menu provides you with an array of options to help you when moving and cross-fading Clips. You can choose how you wish Clips to be handled when moving and adjust whether automatic fades are created or not and how they react when moving Clips.

Editing context menus

There are many context menus in the Audio Montage Workspace to help you while editing your montages. They exist for managing tracks (Track) , Clips (Clip) and inserting Audio Files into your montage (Insert). You can also save selections of Clips as Groups so that you can easily re-select a number of Clips via the Groups window. See Groups for more information on using Groups. Dragging can be used with several editing functions in the Montage Workspace. See Dragging operations for more information.

Tip: Editing multiple envelopes

When editing a cross fade length with the mouse, or the envelope level, you can press **[Alt]/[Option]** at the same time, and this will adjust the fade of all selected Clips.

Getting around the Audio Montage window In addition to some of the standard zooming and playback controls present in the Audio File Workspace there are some that are unique to the Audio Montage Workspace. There are two magnifying glass icons near the scroll wheel which allow you to zoom in and out in the track window. Clicking each icon will display more or less tracks. The active track is zoomed in while the remaining tracks shrink down to allow you to concentrate on one track. See Zooming and scrolling for more information. There is also the handy Navigator for quickly moving around the visible Workspace in the montage main window. To make it easy to quickly move between different views and zoom levels,

you can also capture view settings as "snapshots", and apply these at any time using the Snapshots window.

The info line

The info line is the grey area below the main montage track view. It indicates what will happen when you click the mouse buttons (with or without modifier keys), depending on the location of your mouse (much like tool tips for buttons). This is useful since editing in the audio montage is based on mouse zones – you click in different areas of the Clips to perform different actions. The info line will also display positions and lengths when you are moving, copying or resizing Clips. For more information see Audio Montage contextual information.

Working with Clips

When you add a single Clip to the montage (by pasting, importing from disk or using drag and drop), the Paste context menu appears. By selecting one of the menu items, you can specify how the Clip is inserted, and whether existing Clips should be affected or not, etc. You can choose from:

- **Add/Mix** - The Clip is inserted without affecting any Clips that already exist on the destination track. The exception is if an inserted audio Clip partially overlaps another audio Clip, in which case a cross fade is created in the overlapping zone (this is only the case if the Auto crossfade mode is activated).

- **Add/Mix & bind right Clips (track)** - When the Clip is inserted, all Clips to the right of it (on the same track) are moved to the right, to make room for the inserted Clip.

- **Add/Mix & bind right Clips (global)** - When the Clip is inserted, all Clips to the right of it (on all tracks) are moved to the right, to make room for the inserted Clip.

- **Split/Insert** - This option is only available if the insertion point is within an existing Clip (audio tracks only). When the Clip is inserted, the existing Clip is split and the right section is moved to the right to make room for the inserted Clip. Other Clips are not affected.

- **Split/Insert & bind right Clips (track)** - As Split/Insert, but other Clips to the right on the same track are also moved accordingly (audio tracks only).

- **Split/Insert & bind right Clips (global)** - As Split/Insert, but other Clips to the right on all tracks are also moved accordingly (audio tracks only).

- **Replace selected range** - Only available if there is a selection range on the destination track. The Clip with the selection range will be split at the selection range edges, the inserted Clip will replace the range and the section to the right of the range is moved to the left or right (depending on the length of the inserted Clip vs. the length of the selection range), so that there is no gap after the inserted Clip.

- **Replace selected range (trim source accordingly)** - This is only possible if there is a selection range on the destination track, and if that selection range is shorter than the range of the Clip to be inserted. The inserted Clip will overwrite the selected range only, that is, the range to be inserted will be trimmed according to the current selection.

- **Replace selected range & bind right Clips (track)** - As Replace selected range, but other Clips to the right on the same track are also moved accordingly.
- **Replace selected range & bind right Clips (global)** - As Replace selected range, but other Clips to the right on all tracks are also moved accordingly.
- **Overwrite from cursor** - If the inserted Clip overlaps any other Clips, the overlapped regions are removed from the existing Clips.
- **Cancel** - No Clip is added (you can also cancel the insertion by clicking outside the pop-up menu).

Focused Clip versus Select Clips

It is important to note that there is a difference between a "focused" Clip and a "selected" Clip. Some editing functions can only be executed on an individual Clip or "focused Clip", while some others can be executed on multiple Clips or "selected Clips".

- **Focused Clip** - there can only ever be one or no focused Clip at a time. The focused Clip's name is displayed in red. It's necessary to identify the Clip as focused because there are certain functions that can only be executed on a single Clip. You use the Focused Clip window to perform these functions.
- **Selected Clip** - there can be many selected Clips. This allows you to edit multiple Clips at the same time. You can perform functions such as copy, delete, move, etc. Selected Clips have a different background color.

Managing tracks

Managing tracks is done mainly in the area to the left of each track in the main window. In the middle of this area is a button with the number of the track. Clicking this button brings up the track pop-up menu, which contains most of the track functions. There are also controls for:

- **Minimizing/folding a track** - use the small drop-down arrow to save screen space by toggling the track between a zoomed to closed "folded" view.
- **Soloing a track** - the small diamond-shaped button allows you to solo a track so that all other tracks are muted but it is heard.
- **Muting a track** - the small round shaped button allows you to mute/unmute a track. Gray means the mute button is engaged and the track is not audible. Green means that mute is off and the track is audible.
- **Choose a track's output channel** - the small "Lf:Rf" button allows you to dispatch audio to different channels.
- **Adjust track's volume** - the two small channel faders allow you to adjust that track's volume on both left and right channels (for a stereo track). Holding down **[Ctrl]/[Command]** will reset the sliders to 0dB. In a stereo track holding down **[Shift]** or right-clicking and dragging on either the left or right fader will allow you to adjust the balance of one channel at a time. Use **[Shift]** or right-click and **[Ctrl]/[Command]** to reset a single fader to 0dB.

Multiple Clips and Groups

It is possible to group multiple Clips together by holding down **[Shift]** or **[Ctrl]/[Command]** while selecting them. You can then save them as a "selection group" so that later you can easily select all Clips in that Group with a single click. You can even nest Groups inside each other. See Groups for more information.

Playing back and using effects

Playback in the Audio Montage works in much the same way as in the Audio File Workspace. There are however some additional points to note:

Audio tracks are normally routed through the Master Section

This allows you to add global effects to the Audio Montage or use the Render function to create a mixdown Audio File, separate Audio Files in the case of a multichannel Montage or an intermediary Basic Audio CD. You can also choose to route the audio from a track to the nearest track above for use with Clip effects that allow modulating audio inputs (such as the Ducker plug-in). The routing is done using the "Route to" options on the Track context menu.

Effects can be added to Clips or Tracks

Each independent Clip and/or audio track in the Montage can be independently processed by up totenVST effect plug-ins. Effects are configured either as Inserts, when all sound is processed by the effect, or as Send effects. See Effects for more information.

Using markers in the montage

Markers work in much the same way as in the Audio File Workspace however there are some additional functions available that are useful for working with Clips. You can for example, bind a marker to a Clip so that it moves with the Clip. For more information use "What's this?" or see "Using markers in the Audio Montage Workspace" in Markers for more details.

Mixing down – The Render function

The Render function in the Master Section allows you to "mix down" the whole Montage (or sections of it) to a single Audio File, or to several files in case of a multichannel Montage. It also allows you to render to a basic audio CD or CD image and cue sheet, or even a new Audio Montage. This can be useful for example when you wish to "print" any effects to file. The Render function can be used in many ways:

- It's useful if you want to burn a CD from a CPU-intensive Montage, as it allows you to first render all track and Clip effect processing to recreate a new montage and then burn the CD in a second pass.
- Surround channels can be rendered as multiple files, retaining the stereo/mono status of the individual surround channels.
- You can create a CD image and cue-sheet
- You can create a Basic Audio CD

- It allows you to render a whole Montage to a single file, or various parts to multiple Audio Files in one operation (for example: rendering regions, Groups or Clips)

For more information see Rendering and Render Montage window.

CD Preparation

The Audio Montage Workspace is an especially powerful tool when it comes to authoring professional Audio CDs. You can easily author a CD using the CD window. This window contains a variety of tools for authoring and burning CDs, including a CD Wizard to get you started, full CD playback previewing, the ability to add and edit CD-Text, the generation of CD reports and the ability to check the conformity of your disk to common standards. In conjunction with the Marker Window you can edit, create and delete different Track marker types with precision, giving you the ability to create gap-less CDs and control the pauses between track etc. Any changes you make to Track markers will be reflected in the CD window too. Once you are happy with your CD layout you can burn a CD directly from the CD window or save a DDP image using the Write Audio CD window.

For more information on burning CDs see Authoring CDs and DVDs and the CD window.

DVD-Audio burning

You can also author and burn DVD-Audio disks in the Audio Montage Workspace. This is done using the dedicated DVD-Audio window. The process is similar to creating CDs but there are some differences. Montages are always rendered prior to the DVD burning process. A DVD-Audio project can contain up to 9 Montages with up to 99 tracks in each Montage. Thus, for DVD-Audio you will probably make preparations for several separate Montages that are to be included on the DVD first, then proceed to the DVD-Audio window to configure the DVD-Audio project where you add the separate Montages as groups. Lastly you render the files before the actual burning process takes place. Each Montage can have a different channel configuration, bit resolution and sample rate in the DVD-Audio project.

For more information on authoring DVD-Audio disks see DVD-Audio.

DVD-A picture tracks

DVD-A picture tracks can be added to Montages that are to be burned to DVD. These tracks have no functionality in WaveLab itself, other than offering the ability to place pictures along the time line which will be shown together with the audio when the final DVD-A is played back in a compatible player connected to a video display. To add a picture first add a new picture track via the track menu. Place your cursor where you would like to insert your image and right-click on the track to display the Insert context menu where you can browse for an image file to include. You can also adjust picture settings or add default images via the DVD-Audio options dialog.

Exporting and importing AES-31 files

You can import and export AES-31 files into and out of WaveLab. You can for example, import a project created in Steinberg's Nuendo into WaveLab by importing an AES-31 file. It is possible to add specific codes to the names of markers in Nuendo to facilitate their conversion into WaveLab specific markers so that when the AES-31 file exported from Nuendo

is imported into WaveLab, the markers it contains will be interpreted as WaveLab markers upon import.

For the various CD track markers, the codes to use are as follows:

Marker type	Code	Example marker name
CD track start	[t-start]	So it begins [t-start]
CD track end	[t-end]	The end [t-end] of the road
CD track splice	t-splice]	Intermission [t-splice]
CD track index	[t-index]	[t-index] Hello

- You must use Nuendo 2.0 or later if you want to create specially named markers that will be interpreted as WaveLab markers.
- In Nuendo, a marker track has to be created for the specially named markers.
- When importing AES-31 projects containing specially named markers, the marker codes (e.g. [t-end]) will not be displayed in WaveLab.

For more information on AES-31 files see AES31.

XML Export/Import of Audio Montages

This is available on the Save special (Montage) submenu, and is solely intended for users familiar with XML. XML stands for "E Xtensible Markup Language" and is a markup language much like HTML.

This feature can be useful for example, if you wish to change many file names used by the Audio Montage (export then import). It could also be used to generate Audio Montages from scratch (or from a template), or to convert another DAW file format to WaveLab's Audio Montage. Another application could be to compare two Audio Montages using a text file comparison tool.

There are many freeware XML editors available, but a simple text editor can also be used.

It is beyond the scope of this manual to provide details about how to edit XML files, however here's some basic information:

- All strings are included in a CDATA section, in the UTF-8 format.
- All integer values are in simple textual form: 127 = "127"
- All float and double values are stored in Base64 format.
- Any binary data is stored in Base64 format.

Related topics

Metering

Tools
Transport controls
Specific tool windows
Shared tool windows
Utilities
Audio Montage Colors
Clip
Audio Montage contextual information

6.1 Metering

WaveLab contains a variety of audio meters to help in monitoring and analyzing audio.Meters can be used to monitor audio during playback, rendering, recording and also to analyze a specific selection of audio. There are seven different audio meters in WaveLab, each with its own separate window. The meters are accessed via the Meters menu, the Shared Tool windows menu, or from the Meters control bar.

Monitor Modes

You can choose the audio source and the mode for displaying information in the meters. You can access the following monitoring functions from the Meters menu or via the Meters command bar:

- **Monitor Playback** - the meters display the audio output of the Master Section **after the dithering section, unlike the Master Section's own meter**.
- **Monitor Audio Input** - the meters display the audio input you have chosen in the Audio Streaming Settings. This is useful for monitoring audio while recording.
- **Monitor File Rendering** - the meters display what is being written to disk during file rendering, taking into account Master Section settings, with average, minimum and maximum peak values computed. After rendering, the meter(s) freeze, and remain frozen until you refresh or change the monitor mode.
- **Monitor Edit cursor position** - the meters display static information about the audio beneath the edit cursor. Note that the Master Section settings are not taken into account in this mode.
- **Analyze audio selection** - this allows you to make a selection and have the meters display the average values over the selected range as a static display. The Master Section settings are not taken into account in this mode. When you change the selection, you need to update the meter displays by selecting "Refresh selection analysis" from the Meters menu (or by clicking the Refresh button on the Meters control bar).

If you have one of the continuous display modes chosen, you can also select to freeze the meters at any time using the "Freeze meters" command.

Using meter windows

There can only be one instance of each Audio Meter. For example, if you dock one meter in a Workspace, it will be automatically removed from the place where it was previously located.

Audio Meters can appear in the Audio File, Audio Montage Workspaces as well as the Control Window. They can be used:

- as a docked window in a Workspace
- as a tabbed window in the control window
- as an independent floating window. In this mode, it can be useful to make the window "frameless" from **Window** > **Hide frame**, to save screen space. In this case, the whole menu is accessed by right-clicking.

The axis of most Audio Meters may be rotated, to view the graphics horizontally or vertically.

Some meters can also have their visual style and display parameters customized via their corresponding settings dialog. This can be accessed through the **Functions** > **Settings...** menu in each meter window, or by using the icon.

Related topics

VU Meter

Spectroscope

Oscilloscope

Bit Meter

Phasescope

Spectrometer

Wavescope

6.1.1 VU Meter

The vu-meter is used to display the peak and average loudness/decibel level of your Audio File. It also displays the pan, or the balance between the left and right channels in a stereo file.

The upper part of the level/pan meter window shows the peak level and average loudness, in the following way:

- The Peak Level meters display the peak levels of each channel, graphically and numerically. By default, the meter segments and numerical peak values are displayed in green for low levels, yellow for levels between -6dB and -2dB, and red for levels above -2dB. You can change both the colors and the range boundaries if you wish via the Level/Pan Meter Settings dialog. This can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

- The VU (Volume Unit) meters measure the average loudness (RMS) of each channel. These meters have a built-in inertia, evening out loudness variations over a user-defined

time span. If you are monitoring playback or audio input, you will also note two vertical lines following each VU meter bar, seemingly "trying to reach" the current RMS value. These lines indicate the average of the most recent minimum RMS values (left line) and the average of the most recent maximum RMS values (right line). To the left, the difference between the minimum and maximum average values is displayed (the level value in brackets) - this gives you an overview of the dynamic range of the audio material.

- If you are monitoring real-time audio (playback or input), the maximum peak and loudness values are displayed numerically to the right of the meter bars. Numbers in brackets to the right of the Maximum Peak values indicate the number of successive clips (0dB signal peaks). Recording levels should be set so that they only rarely clip. If the master level is set too high, the sound quality and frequency response will be compromised at high recording levels, with unwanted clipping effects. If the level is set too low, noise levels may be high relative to the main sound being recorded.

Pan meters

The lower part of the window shows the pan (the difference in level between the left and right channel, only applicable when monitoring stereo audio):

- The upper pan meters show the peak level difference between the channels, graphically and numerically. Note that the pan meters are "two-sided"; the level bars can go to the left or right, indicating which channel is the loudest. The two sides are shown in different colors (which can be changed via the Settings dialog as explained previously).

- The lower pan meters show the average difference in loudness between the channels, in a similar way. This gives you a visual indication of whether a stereo recording is properly centered, for example.

- If you are monitoring real-time audio (playback or input), the maximum balance difference value (peak and loudness) for each channel is displayed numerically to the left and right of the meter bars. The VU meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces.It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

6.1.2 Spectroscope

The Spectroscope shows a continuous graphical representation of the frequency spectrum, analyzed into 60 separate frequency bands, represented as vertical bars. Peak levels are shown as a short horizontal lines above the corresponding band, indicating recent peak/-maximum values. The Spectroscope offers a quick spectrum overview. For a more detailed analysis of the audio spectrum, use the Spectrometer. The Spectroscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

Spectrometer

6.1.3 Oscilloscope

The Oscilloscope offers a highly magnified view of the waveform around the playback cursor position.

If you are analyzing a stereo file, the Oscilloscope normally shows the separate levels of the two channels. However, if you activate the option "Show Sum and Subtraction" on the Functions menu (or click the +/- icon), the upper half of the Oscilloscope shows the sum of the two channels and the lower half shows the subtraction.

Making settings

By opening the Settings dialog you can adjust the display colors, and choose to activate Auto-zoom or not. With Auto-zoom activated, the display is optimized so that the highest level reaches the top of the display at all times. The Settings dialog can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

The Oscilloscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

6.1.4 Bit Meter

The Bit Meter shows the resolution or the number of bits used in the digital audio being monitored. Normally the maximum number of bits in an Audio File is the same as the resolution of the Audio File (e.g. a 16 bit Audio File shows that up to 16 bits are used), but sometimes this is not the case.

As soon as you perform any kind of real-time processing on an Audio File, the audio data is treated at a much higher resolution (32 bit floating point), to allow for pristine audio quality. Examples of such processing include level adjustments, effects, the mixing of two or more files, etc. In fact, the only time when a 16 bit file is played back at 16 bit resolution is if you play it without any fades or effects, and with the Master Faders set to 0.00 (no level adjustment). You can try this out for yourself by playing back a 16 bit Audio File and viewing the Bit Meter - as soon as you adjust the Master Faders, note that 24 bits are used and the "inter" indicator is lit (more on this below).

How to read the Bit Meter

- The innermost meters (closest to the bit scale) show the number of bits in use. You can adjust this display in the Settings dialog.

- The outer meters are "history" meters, showing how many bits were recently in use. You can adjust the hold time in the Settings dialog.

- The "over" segment indicates clipping, similar to a clip indicator.

- If the "below" segment is lit, there are more than 24 bits. The bit meter will show the 24 higher bits, and the "below" segment indicates the existence of extra, lower bits. Note that audio is always processed with more than 24 bits internally.

- If the "inter" segment is lit, this indicates that the audio data cannot really be expressed on a regular 24 bit scale (there are floating point values "in between" bits, thus the label "inter"). This is typically the case if you apply effects; thus the "inter" segment allows you to differentiate processed from non-processed 24 bit PCM files.

When to use the Bit Meter

- To check whether dithering is necessary or not. As a rule, if you're playing back or mixing down to 16 bits, and the Bit Meter shows that more than 16 bits are used, you should apply dithering.

- To see the "actual" resolution of an Audio File. For example, even though a file is in 24 bit format, only 16 bits may be used. Or, a 32 bit file may only use 24 bits (in this case, the "below" segment would not be lit). For this purpose, the Bit Meter is best used in "Analyze selection" mode.

- To see whether a "zeroed" plug-in still affects your signal, or whether a plug-in uses 16 bit internal processing, or more generally, to detect any signal modifier stage in the audio chain between reading and playing.

To adjust the Bit Meter settings, select "Settings..." from the Options pop-up menu or by using the icon.

The Bit meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Dithering

6.1.5 Phasescope

The Phasescope indicates the phase and amplitude relationship between two stereo channels. It is only really relevant when monitoring stereo audio material.

Reading the Phasescope

It can be interpreted as follows:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.
- A random but roughly elliptical shape indicates a well balanced stereo signal. If the shape "leans" to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the Phasescope will show a straight line, angled 45 degrees to the other side).
- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90 degrees on the other.
- Generally, the more you can see a "thread" shape, the more bass there is in the signal; the more "spray-like" the display, the more high frequencies are in the signal.

Phase Correlation meter

At the bottom of the display, you will find a Phase Correlation meter, which shows the same information but in a different way:

- The green line shows the current phase correlation, while the two red lines show the recent peak minimum and maximum values, respectively (these are the default colors, which you can change).
- With a mono signal, the meter would show +1, indicating that both channels are perfectly in phase.
- Similarly, -1 indicates that the two channels are the same, but one is inverted.
- Generally, for a "good" mix, the meter should show a value between 0 and +1.

Unlike the main Phasescope, the Phase Correlation meter is also available in "Analyze audio selection" mode, showing an average value for the selected range.

Changing settings

You can define display colors, peak hold time, and the resolution or number of samples to display from the **Functions** > **Settings...** menu.

The Phasescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

6.1.6 Spectrometer

The Spectrometer uses FFT (Fast Fourier Transform) techniques to display a continuous frequency graph, providing a very precise and detailed real-time frequency analysis.

- The current frequency spectrum is shown as a linear graph.
- Spectrum "peaks" are shown as a short horizontal lines, indicating recent peak/maximum values.

Snapshots

By using the "Add snapshot" and "Erase last snapshot" buttons, you can take and erase snapshots of the current spectrum. These will be superimposed over the current peak spectrum graph, in a color you can customize, until you click the icon again to take a new snapshot. One use for this is to check the effects of adding EQ, for example. Up to five snapshots can be superimposed on the display; the sixth snapshot replaces the earliest one in the display, and so on. (Don't confuse the order of snapshots with the numbered buttons - these are Spectrometer presets, see below.)

Zooming

You can adjust the frequency scale and range in the Settings dialog as described below, but it's also possible to temporarily zoom in on a frequency area of interest. This is done by clicking and dragging a rectangle in the Spectroscope display. When you release the mouse button, the display is zoomed in so that the enclosed frequency range fills the window. To return to full-scale display, select "Zoom-out fully" from the Functions menu, or double-click anywhere on the display.

Making settings

You can adjust the behavior and display of the meters as desired, and assign up to five sets of Spectrometer settings to the Preset buttons, for instant access. Open the Settings dialog by selecting "Settings" from the Functions menu or clicking the "tool" icon. Note that you can apply your settings without closing the dialog, by clicking the Apply button.

If you want to store your settings for later use (or assign them to a Preset button), select "Save as..." from the pop-up menu in the lower part of the dialog, and specify a name for the preset in the file dialog that appears. Now, you can choose to make the settings instantly available for selection in the FFT Meter window, by using the "Assign to preset button" submenu on the pop-up menu.

- When you are finished, click OK to close the dialog.

Selecting Spectrometer presets

If you have assigned your settings to the Preset buttons in the Settings dialog, you can quickly switch between different level scales and display modes, by clicking one of the Preset icons [1]-[5], or selecting the desired preset from the Options pop-up menu.

Exporting FFT data as ASCII text

When using the Spectrometer in off-line mode ("Monitor Edit cursor Position" or "Analyze audio election" mode) you can export the displayed FFT data as a text file, by selecting "Export FFT data as ASCII" from the Options pop-up menu. The resulting text file can then be imported into applications that allow graph plotting from text files (Microsoft Excel, for example).

The Spectrometer can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Spectroscope

6.1.7 Wavescope

The Wavescope meter displays a real-time waveform drawing of the audio signal being monitored.

It can be useful when recording or rendering a file if "Monitor File rendering" mode is active.

Making settings

You can adjust settings for the display via the Wavescope Settings dialog. This can be accessed via **Functions** > **Settings...** menu, or by using the icon. Here you can set various color options for the background, grid and waveform display, as well as setting the waveform rendering speed and vertical zoom. If "Clear waveform when reaching right of pane" is checked the waveform display is cleared each time the cursor reaches the right end of the display. If unchecked, the previous waveform is overwritten.

Tip: Wavescope is a useful meter for visualizing audio during recording. The Wavescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

6.2 Tools

In the Audio Montage Workspace, WaveLab provides you with some tools for assisting you in editing your montages. This page contains links to these tools which can be accessed from the **Tools** menu.

Related topics

Meta-normalizer

Configure external tools

6.2.1 Meta-normalizer

This dialog allows you to adjust the loudness of each Clip so that they get the same loudness.

You can use it to equalize the volume of all the Clips within a montage, either by peak level or loudness. This can be an easy way to ensure that all Clips playback at a similar volume. You can choose to only equalize the volume of selected Clips, and effects can be excluded if you wish.

This tool is to achieve the **same** level of loudness in all Clips (the highest loudness found, if possible), while ensuring that no Clip will cause sample clipping.

For each Clip, a specific gain is computed by the algorithm after **all** Clips have been analyzed (and prior to changing the Clip gain to achieve the common loudness). If it is not possible to match the highest found loudness, the level of the Clip with the highest loudness is reduced, so that other Clips can achieve the same loudness.

To avoid clipping the Master Section you can also normalize the mixdown output of the montage (before it goes into the Master Section), and/or the Master Section output.

You can access this dialog from the Audio Montage Workspace by choosing **Tools** > **Meta-normalizer...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Master Section

6.2.2 Configure external tools

This dialog allows you to configure WaveLab to work with external applications. You can pass command line arguments to the external applications so that they can process the current file/folder you are working on, or even the WaveLab's settings folder. This function can be useful if you wish to edit an Audio File in another application, or wish to compress all your settings files into a backup zip file, for example.

Note that an external tool only works within the Workspace in which it is defined. This is by design, so that each type of Workspace can have its own external "toolkit".

To define an external tool you need to:

- Give the command a title
- Specify the path to the application's executable file
- Choose which arguments you wish to pass to the external application
- Additionally you can specify an initial folder to direct the external application to

- Select how you wish WaveLab to behave before the external tool is launched.

Once you have defined an external tool in this way, you can run it by selecting its title from the Tools menu from within its Workspace.

You can access this dialog by choosing **Tools** > **Configure external tools...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

6.3 Transport controls

The transport controls allow you to control playback within an Audio File or montage, and to open the record window. It can be accessed via the Transport menu or the Transport toolbar. See Command bars for more information.

The Transport menu gives access to (and shows keyboard shortcuts to) the same functions that are available in the transport toolbar.

The Transport toolbar gives quick access to most of these functions. If the toolbar is hidden, select **Workspace** > **Command bars** > **Transport commands**. Transport functions operate on the current Audio File.

The basic transport functions are:

- **Loop On/Off**
  - Toggles looping on/off for the currently selected Audio File or montage.

- **Start of file / End of File**
  - Move the cursor to the beginning or end of the currently selected Audio File.

- **Skip forward / Skip backward**
  - Skip the cursor forwards or backwards within the currently selected Audio File.

- **Play**
  - Begin playback of the current Audio File.

- **Stop**
  - Stop playback of the current Audio File. A second click moves the cursor to the beginning of the last start position. A further click returns the cursor to the start of the file.

- **Record**
  - Opens the recording window from which you can start recording.

Additional transport functions such as start, skip and stopping playback options are also available from the transport command bar. In WaveLab additional playback speed and jog/shuttle shortcuts are also provided.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Playback speed

Recording Dialog

Command bars

6.4 Specific tool windows

Specific tool windows are windows that are specific to the current Workspace. They perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Audio Montage Workspace specific tool windows

The following specific tool windows are available within the Audio Montage Workspace: File

CD

File Browser

Marker Window

History

Clips

Focused Clip window

Notes

Master Project

Scripting window (Audio Montage Workspace)

Groups

DVD-Audio

Navigator

Zoom

Effects

Snapshots

6.4.1 File

This specific tool window helps you to manage files that are used in the current Audio Montage.

It displays all current files that are being used by Clips in the current montage, along with their location, size and last modified date. In addition a number of file operations are available:

- add one or more Audio Files to a montage, this will open the Insert Audio Files window when one than one file is selected

- replace individual files in a montage (this will update any Clip references)
- rename a file (this updates all internal Clip references)
- select Clips that are referenced by a specific file
- open a file in the Audio File Workspace, and view the file in your computer's file browser.

This window is a useful tool to use when constructing and editing your montage.

You can access this dialog in the Audio Montage Workspace by choosing **Workspace** > **Specific tool windows** > **Files**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Insert Audio Files

6.4.2 CD

This specific tool window brings together all the functions for creating an audio CD or DVD-A within WaveLab.

It displays a list of CD tracks along with information about each track. You can edit each track and the CD's playback properties as well as access functions to check its conformity to Red Book standards, add and edit CD-text, add UPC/EAN codes, generate a CD report and burn the CD. There are also controls to preview skipping between each track and a number of other options including the ability to choose the "Audio in Pause" mode to determine whether gaps between track markers are replaced with silence (the normal default) or if they contain sound.

Adding tracks

To create an audio CD in WaveLab you must at least have a CD start and end marker pair. You can generate these automatically from audio Clips in the montage using the CD Wizard or by adding track markers manually.For more information on creating a CD or DVD-A see Authoring CDs and DVDs.

The Track list

Once you have created one or more tracks, the CD window displays a list of all tracks in the currently active montage along with their corresponding properties. From this list of CD tracks you can use the fields, controls and menus to:

- **Audition playback** - It is possible to audition playback at the selected track's start time, with or without pre-roll, by clicking on the leftmost icons. Click on the left hand button in the "audition" column to play the track from its start time with a pre-roll. Click on the right hand button to play the track exactly from its start time.

- **Edit a tracks's name** - You can add/edit a name for each track. Double click in a cell to edit. Note that to change the track name is to indirectly edit the marker name.

- **Edit a track's start/end time** - You can edit the start and end time for each track. This is the same as moving the track markers physically but can be done with more precision. Double-click in a cell to edit.

- **Pre-gap time** - This displays the gap between the CD track start marker and where the first track's audio actually begins. It can only be edited by physically moving the CD track start marker.

- **Post-gap time** - This displays the gap between the end of the audio of the CD's last track and the CD track end marker. It can only be edited by physically moving the CD track end marker.

- **Copy protection** - Select the checkbox to include a copy protection flag with the track. This may prevent it from later being imported by software that recognises this flag.

- **Emphasis** - This setting is used to indicate if the track was recorded with emphasis or not. Please note that this does not apply/remove emphasis from the audio, it is just an indicator as to how the file was created. Normally it is left off.

- **ISRC Code** - This allows you to enter an optional International Standard Recording Code. Double-click in a cell to add/edit a code.

- **CD-Text** - You can optionally enter CD-Text information for each track and the entire CD/DVD-A. Double clicking a cell will open the CD Text Editor.

- **Add/Edit a tracks's comments** - You can add/edit text comments for a track. Double click in a cell to edit. Note that this text comment is for reference only: nothing is burnt on CD.

Note that CD tracks can be reorderd by dragging them into new positons: this means Clips and markers are moved accordingly.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Authoring CDs and DVDs

Write Audio CD

CD Wizard

CD Text Editor

Audio CD Report

CD Pre-emphasis

Red Book CD-DA

CD Text

ISRC

UPC/EAN

6.4.3 File Browser

This dialog allows you to browse files directly from within WaveLab rather than using your operating system's file browser.

It provides you with all the standard browsing functions (such as List and Icon views)as well as additional controls to audition Audio Files and any marker defined regions.You can use it to open or insert an entire file or a specific region of a file by dragging and releasing the file in the location you wish to insert the file. In the Audio Montage Workspace you can also choose to only view certain types of WaveLab specific files. The File Browser can be very useful in speeding up the process of auditioning a long list of sound files. It can also be used to drag to windows such as Data CD/DVD, including dragging folders.

The playback functions

Play file Once you have selected a file in the File Browser list, you can audition it by clicking Play in the File Browser's toolbar. Click again to stop.

Auto-play You can also have files play back automatically as soon as they are selected. To do this, activate Auto-Play mode.

Tip: To audition a long list of Audio Files, enable Auto-play and use the UP and DOWN cursor keys to step through each Audio File in quick succession.

Opening and inserting a file or region

Once you have chosen a file you wish to open, double-click it to open the file in the current Workspace. You can also open it by dragging it onto an empty tabbed group, or onto the tab bar of a tabbed group.

You can also drag the file over a wave view to **insert** it at a given point. When you drag a file over a wave view, a beam is displayed so as to clearly display the insertion point.

When browsing Audio Files, the file browser also has a region panel. When you select an Audio File, its related regions are listed in this panel. You can then drag the region from the list, to insert only that part of the audio. Double-clicking on a region has the effect of opening the file in the Workspace and selecting that region. You can access this dialog via **Workspace** > **Specific tool windows** > **File Browser**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

6.4.4 Marker Window

The marker window is a specific tool window that allows you to create, edit and use markers while working on an audio waveform.

From the Functions menu you can also access useful functions to convert marker types, rename multiple markers and export the marker list as text.

The Marker list

The marker window contains a list of any markers placed in the currently active file along with their corresponding details. Click on a column header to sort the list in ascending or descending order based on the values in that column. You can also use the Filter menu to toggle which types of markers are displayed in the list.

From this list of markers you can use the fields, controls and menus to:

- **Create a new marker at the playback head position** - click on a marker icon (at the top of the window) or choose a marker type from the Insert menu to drop a marker at the current playhead location.

- **Create a pair of markers from a selection** - make a selection in the waveform and then click on a pair of marker icons (at the top of the window) or choose a "Create region" command from the Insert menu to drop a pair of markers at either side of the current selection.

- **Start playback** - It is possible to start playback at the selected marker position, with or without pre-roll, by clicking on the leftmost icons. Click on in the "audition" column to play the wave from that marker position with a pre-roll. Click on the right hand button to play the wave exactly from the marker position.

- **Change marker types** - You can change a marker type by (left) clicking on its icon and choosing another marker type from the pop-up list that appears.

- **Edit a marker's name/time/comments** - double click in a cell to edit a value (to align the marker to an exact time, for instance). You can edit the name of the marker, its time position and any comments.

- **Lock a marker's position** - select the checkbox to lock the marker. This prevents it from being accidentally dragged to a new position in the Waveform window.

For more information on using markers see Markers. For details on the different marker types and their uses see Marker types. Note that there are more marker features available in the Montage version compared to the Audio File version. See "Using markers in the Audio Montage Workspace" in Markers for more details.

You can access this specific tool window within the Audio Files and Audio Montage Workspaces via **Workspace** > **Specific tool windows** > **Markers**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Markers

Marker types

Convert marker type

Text format

6.4.5 History

This specific tool window allows you to view a history of all your recent editing operations.

You can revert the current montage to a previous state by selecting the operation you want to return to and double clicking on it. You can then use the redo command (**Edit** > **Redo**) to return to any state. Use the history menu options (**Edit** > **History** >) to undo all operations, redo all operations, or clear the history.

There are a couple of history options available in the Audio Montage preferences window. You can choose to empty the history each time you save a montage file and also to group similar types of sequential operations into a single operation. Both of these options enable you to use less memory when storing the operation history.

You can access this window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **History**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montage Preferences

6.4.6 Clips

This specific tool window contains a list of any Clips placed in the currently active montage file together with their corresponding details.

Click on a column header to sort the list in ascending or descending order based on the values in that column. Clicking on any row of the list will move the main window's view to show the selected Clip. To change a value, double click in a cell to begin editing (to align the Clip to an exact time, for instance). Clips may be dragged into new positions in the list.

From this list of Clips you can use the input fields and controls to:

- **Start playback** - It is possible to start playback at the selected Clip position, with or without pre-roll, by clicking on the leftmost icons. Click on in the "audition" column to play the montage from that Clip's position with a pre-roll. Click on the right hand button to playback the montage exactly from the start of the Clip.

- **Edit a Clips's name/start/end/length/comments** - edit the name of the Clip, its start/end time position and any comments.

- **Lock a Clips's position** - select the checkbox to lock the Clip. This prevents it from being accidentally edited or dragged to a new position.

- **Change Gain** - use this control to increase/decrease the gain of the Clip. Double click in the cell to edit the value.

- **Mute** - select this to mute the Clip. Its color changes in the track view and it is no longer audible during playback.

A wide range of Clip selection features, Clip functions, and Clip display options are available from the Clips menu. Use "What's This?" help for information on individual menu entries.

You can access this window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Clips**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Audio Montages

Clip

6.4.7 Focused Clip window

The Focused Clip window is an accordion control that contains some sophisticated editing commands. Each collapsible panel can be expanded by clicking on its title and allows you to access a range of tools for working with the selected or "focused" Clip.

Using the Focused Clip window

To use the Focused Clip window you first need to select a Clip. Once you have selected a Clip with your cursor its name will highlight in red. You can now choose one of the Focused Clip accordion's tabs. The accordion will open to reveal options for that particular pane's selection of editing shortcuts and tools. There are a number of panes including:

- **Edit** - provides you with a range of common playback shortcuts and editing tools for working with Clips.

- **Cue Point** - a cue point is a defined position marker that belongs to a Clip. It helps to align Clips together by making Clips snap to other Clips at the cue point position. Cue points are displayed as dotted vertical lines. This pane provides you with a number of shortcuts for creating and working with Cue Points.

- **Envelope** - allows you to access a number of controls for editing and working with a Clip's different envelopes.

- **Fade-in** - provides you with a range of tools for adjusting a Clip's Fade-in curve.

- **Fade-out** - provides you with a range of tools for adjusting a Clip's Fade-out curve.

- **Colors** - up to 20 colors may be assigned to help distinguish particular Clips visually. Select a Clip then click on a color. Custom colors can be defined with great subtlety from Options > Colors... Select the Custom color to be defined, then the parts to which it is applied, then define the color value using the color wheel (or type a set of RGB - Red, Green, Blue values). See Audio Montage Colors for more information.

When you choose a different Clip the properties in the Focused Clip window will update. So for example, if you select "Envelope" the corresponding controls will update to reflect the selected Clip's envelope information.

The Focused Clip window can be docked or floated. You can reveal or hide the focused Clip window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Focused clip**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Clip

Audio Streaming Settings

Customize Commands

Audio Montage Colors

6.4.8 Notes

This specific tool window allows you to keep notes about the current Audio Montage session.

You can type into the window and use the standard HTML text editor controls to format your text, add images, hyperlinks and lists. These notes are saved along with the Audio Montage file.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Notes**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

6.4.9 Scripting window (Audio Montage Workspace)

This window allows you to write and execute scripts in the Audio Montage Workspace.

The inbuilt text editor helps you when writing scripts by highlighting the different parts of the script with colors, making it more readable. A script can also be written in another text

editor, and loaded via the File menu. To run a script, from within the script window choose **Functions** > **Execute script**.

For a general description of scripting, see Scripting. For an overview of the scripting language, see ECMAScript Reference.

You can access this dialog from the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Script**.

Note: To view trace or log messages when executing scripts that contain the logWindow() function, ensure that the Log window is visible and that it's warning filter buttons - ! ⚠ ⓘ are selected.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Scripting

ECMAScript Reference

Log window

6.4.10 Groups

This specific tool window displays a list of Groups within the current Audio Montage. Groups are a selection of Clips that you can quickly re-select via the Groups window. You can group a number of Clips and nest Groups inside each other. It is also possible to disable individual Groups and color Groups so you can identify them easily.

Grouping Clips

To create a Group from a number of Clips:

1. Select the Clips you want to group.
2. Make sure you have the Groups window displayed.
3. Select "Group selected clips" from the drop down menu or click the Group icon.
4. Enter a name for the Group in the dialog that appears and click OK.

The new Group appears in the Group list. All Clips included in the Group will also have the name of the Group prepended to the Clip names in each track. Clicking on a Group in the Groups window will select all the Clips in your Montage so you can edit them together.

Adding Clips to an existing Group

If you already have a Group, and want to add one or several more Clips to it:

1. Select the Clips you want to add.
2. In the Groups view, select "Group selected clips", or click the corresponding icon.

3. In the dialog that appears, select the Group to which you want to add the Clips.
4. Click OK.

Removing Groups

1. Select the Group by clicking on its name in the Groups view list.
2. Pull down the Grouping menu and select "Remove selected group". The Group is removed (the Clips are not affected).

Using Groups

There are some important things to remember when using Groups:

- A Clip cannot be part of more than one Group. If you add a Clip to a Group, it is automatically removed from any other Group it is part of.
- It is possible to create nested Groups, by adding a Group to another Group.
- You can temporarily "turn off" a Group by deactivating the checkbox to the left of a Group in the list. When a Group is deactivated, you can move individual Clips in the Group, as if they were not grouped.
- You can select a specific color for a Group to make it easy to discern it in the track view.
- Double-click to rename a Group
- You can nest a Group inside another Group by dragging it
- The Render option is available to render all Groups as individual files (Render dialog)

You can access this window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Groups**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Render Montage window

6.4.11 DVD-Audio

This specific tool window allows you to author a DVD-Audio disk from a collection of audio montages.

You can add your finished audio montages into this window, check the DVD-Audio layout for conformity and output a DVD-Audio disk all from this window. **Note:** All included montages need to be in DVD-A mode (this can be set via Audio Montage properties).

Although DVD-A has a different specification from an Audio CD, its track concept makes is similair to an Audio CD - this is why creating a DVD-A is almost like creating a CD with

WaveLab's Audio Montage. It is worth noting that throughout WaveLab, where "CD" is mentioned (for example, in messages), one can usually substitute "DVD-A".

To create a DVD-Audio disk, follow these basic steps:

- **Create an audio montage** - you need at least one audio montage in order to create a DVD-A. You can import any currently open ones or add them as *.mon files. Note that the Audio Montage must have DVD tracks defined with red (track) markers.
- **Add Montage(s)** - choose **File** > **Add Audio Montage(s)** from the DVD-Audio window to add one or more montages to your DVD-Audio. Each montage will then be displayed along with its details in the DVD-Audio window. You can drag the montages vertically to determine the track order. Montages can also be added by dragging montage files directly into the DVD_Audio track listing window.
- **Choose Options** - from the DVD-Audio window choose **Options** > **Options...** to open DVD-Audio options dialog. You can then define settings for your DVD-A including its name and volume information details, whether it will be played back on PAL or NTSC, and the duration and effects/transitions that included images will use. Images to be included can be laid out in a picture track in your audio montage or a single default image can be chosen. If a default image is used it will displayed at the beginning of each montage
- **Check and burn your DVD-A** - once you are finished laying out your DVD-A and choosing any options, you can check that it conforms to DVD-Audio specifications by choosing **Edit** > **Check DVD-Audio conformity...**. If everything is OK, choose **Edit** > **Burn DVD-Audio...** to open the DVD-Audio Creation dialog where you can then render out the disk as a collection of files ready for burning. Once these files are successfully outputted the Write Data CD/DVD dialog will open automatically allowing you to burn your DVD-Audio files to an optical DVD.

Note that WaveLab automatically creates some basic DVD menus to access the audio tracks from the DVD-Player.

For more information on authoring DVD-Audio disk see Authoring CDs and DVDs.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **DVD-Audio**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

DVD-Audio Creation

DVD-Audio options

Write Data CD/DVD

6.4.12 Navigator

This specific tool window displays a representative view of the entire active montage and enables you to quickly navigate around it.

Each track is represented by a horizontal colored block. By default, stereo tracks are dark blue and monotracks are purple or pink (any custom colors you set for a track or via the Groups window are also displayed).

The visible Workspace is shown by a surrounding black rectangle. Dragging this rectangle scrolls the main montage window, enabling you to easily navigate to a location in your montage. You can also resize the rectangle vertically and horizontally to adjust the zoom setting of the active window. This window can be useful when you have a large montage with many Clips and tracks.

- Left click on a Clip block to zoom on it
- Right click anywhere to completely zoom out

You can access this window in the Audio Montage Workspace by choosing **Workspace** > **Specific tool windows** > **Navigator**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

Groups

6.4.13 Zoom

This specific tool window displays a magnified view of the selected track and allows you to accurately adjust the cross fade point for two adjacent Clips.

You can adjust the zoom level of the selected track, manually shift the Clip's position or allow WaveLab to find the best location to begin a cross fade between two Clips. You can adjust the amount that WaveLab will search for the best cross fade position to avoid phase cancellation issues, and also view the envelope points of the Clips. Mouse operations are identical to those in the main Clip view.

You can access this window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Zoom**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

6.4.14 Effects

This specific tool window allows you to add VST effect plug-ins, both to individual Clips and tracks. The difference between Clip effects and track effects are that Clip effects affect individual Clips only, whereas track effects affect all Clips on a track.

Each independent Clip and/or audio track in the Montage can be independently processed by up to ten VST effect plug-ins. Effects are configured either as Inserts, when all sound is processed by the effect, or as Send effects where the balance between the unprocessed sound and the effect send level can be adjusted or controlled by effect envelope curves (clip effects only). Note that not all effects are capable of "Send", and if this feature is not available, it is disabled.

The window only shows the effects of the focused Clip, or the focused track according to the active mode. Each plugin and track can be understood as being associated with a small "Master Section". A plug-in can be inserted and sorted, and you double-click on an effect to Edit it. When you close an effect window, its parameters are saved into the montage and it can be be undone if necessary. Use "What's This?" help on the column headers for more details.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Effects**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

6.4.15 Snapshots

This dialog enables you to save a number of "views" of your montage. You can then recall a particular view or snapshot at any time.

When you save a snapshot you capture the current zoom factor, the current cursor position, the current scroll position and the current time range displayed. You can name each snapshot and update an existing snapshot too. To recall a snapshot, just click on its title in the list. This restores all of its view settings. You can also choose to recall all or just one of its view properties by clicking the appropriate icon.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Snapshots**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

6.5 Shared tool windows

Shared tool windows are windows that can only exist in one place at a time and are shared between Workspaces. They perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Shared tool windows:

Phasescope
VU Meter
Oscilloscope
Bit Meter
Spectroscope
Spectrometer
Wavescope
Timecode
Background task monitor
Log window
Master Section

Related topics

About Tool Windows
Metering
Docking windows

6.5.1 Phasescope

The Phasescope indicates the phase and amplitude relationship between two stereo channels. It is only really relevant when monitoring stereo audio material.

Reading the Phasescope

It can be interpreted as follows:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.
- A random but roughly elliptical shape indicates a well balanced stereo signal. If the shape "leans" to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the Phasescope will show a straight line, angled 45 degrees to the other side).
- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90 degrees on the other.
- Generally, the more you can see a "thread" shape, the more bass there is in the signal; the more "spray-like" the display, the more high frequencies are in the signal.

Phase Correlation meter

At the bottom of the display, you will find a Phase Correlation meter, which shows the same information but in a different way:

- The green line shows the current phase correlation, while the two red lines show the recent peak minimum and maximum values, respectively (these are the default colors, which you can change).
- With a mono signal, the meter would show +1, indicating that both channels are perfectly in phase.
- Similarly, -1 indicates that the two channels are the same, but one is inverted.
- Generally, for a "good" mix, the meter should show a value between 0 and +1.

Unlike the main Phasescope, the Phase Correlation meter is also available in "Analyze audio selection" mode, showing an average value for the selected range.

Changing settings

You can define display colors, peak hold time, and the resolution or number of samples to display from the **Functions** > **Settings...** menu.

The Phasescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

6.5.2 VU Meter

The vu-meter is used to display the peak and average loudness/decibel level of your Audio File. It also displays the pan, or the balance between the left and right channels in a stereo file.

The upper part of the level/pan meter window shows the peak level and average loudness, in the following way:

- The Peak Level meters display the peak levels of each channel, graphically and numerically. By default, the meter segments and numerical peak values are displayed in green for low levels, yellow for levels between -6dB and -2dB, and red for levels above -2dB. You can change both the colors and the range boundaries if you wish via the Level/Pan Meter Settings dialog. This can be accessed through the **Functions** > **Settings...** menu, or by using the icon.
- The VU (Volume Unit) meters measure the average loudness (RMS) of each channel. These meters have a built-in inertia, evening out loudness variations over a user-defined

time span. If you are monitoring playback or audio input, you will also note two vertical lines following each VU meter bar, seemingly "trying to reach" the current RMS value. These lines indicate the average of the most recent minimum RMS values (left line) and the average of the most recent maximum RMS values (right line). To the left, the difference between the minimum and maximum average values is displayed (the level value in brackets) - this gives you an overview of the dynamic range of the audio material.

- If you are monitoring real-time audio (playback or input), the maximum peak and loudness values are displayed numerically to the right of the meter bars. Numbers in brackets to the right of the Maximum Peak values indicate the number of successive clips (0dB signal peaks). Recording levels should be set so that they only rarely clip. If the master level is set too high, the sound quality and frequency response will be compromised at high recording levels, with unwanted clipping effects. If the level is set too low, noise levels may be high relative to the main sound being recorded.

Pan meters

The lower part of the window shows the pan (the difference in level between the left and right channel, only applicable when monitoring stereo audio):

- The upper pan meters show the peak level difference between the channels, graphically and numerically. Note that the pan meters are "two-sided"; the level bars can go to the left or right, indicating which channel is the loudest. The two sides are shown in different colors (which can be changed via the Settings dialog as explained previously).

- The lower pan meters show the average difference in loudness between the channels, in a similar way. This gives you a visual indication of whether a stereo recording is properly centered, for example.

- If you are monitoring real-time audio (playback or input), the maximum balance difference value (peak and loudness) for each channel is displayed numerically to the left and right of the meter bars. The VU meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces.It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

6.5.3 Oscilloscope

The Oscilloscope offers a highly magnified view of the waveform around the playback cursor position.

If you are analyzing a stereo file, the Oscilloscope normally shows the separate levels of the two channels. However, if you activate the option "Show Sum and Subtraction" on the Functions menu (or click the +/- icon), the upper half of the Oscilloscope shows the sum of the two channels and the lower half shows the subtraction.

Making settings

By opening the Settings dialog you can adjust the display colors, and choose to activate Auto-zoom or not. With Auto-zoom activated, the display is optimized so that the highest level reaches the top of the display at all times. The Settings dialog can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

The Oscilloscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

6.5.4 Bit Meter

The Bit Meter shows the resolution or the number of bits used in the digital audio being monitored. Normally the maximum number of bits in an Audio File is the same as the resolution of the Audio File (e.g. a 16 bit Audio File shows that up to 16 bits are used), but sometimes this is not the case.

As soon as you perform any kind of real-time processing on an Audio File, the audio data is treated at a much higher resolution (32 bit floating point), to allow for pristine audio quality. Examples of such processing include level adjustments, effects, the mixing of two or more files, etc. In fact, the only time when a 16 bit file is played back at 16 bit resolution is if you play it without any fades or effects, and with the Master Faders set to 0.00 (no level adjustment). You can try this out for yourself by playing back a 16 bit Audio File and viewing the Bit Meter - as soon as you adjust the Master Faders, note that 24 bits are used and the "inter" indicator is lit (more on this below).

How to read the Bit Meter

- The innermost meters (closest to the bit scale) show the number of bits in use. You can adjust this display in the Settings dialog.
- The outer meters are "history" meters, showing how many bits were recently in use. You can adjust the hold time in the Settings dialog.
- The "over" segment indicates clipping, similar to a clip indicator.
- If the "below" segment is lit, there are more than 24 bits. The bit meter will show the 24 higher bits, and the "below" segment indicates the existence of extra, lower bits. Note that audio is always processed with more than 24 bits internally.
- If the "inter" segment is lit, this indicates that the audio data cannot really be expressed on a regular 24 bit scale (there are floating point values "in between" bits, thus the label "inter"). This is typically the case if you apply effects; thus the "inter" segment allows you to differentiate processed from non-processed 24 bit PCM files.

When to use the Bit Meter

- To check whether dithering is necessary or not. As a rule, if you're playing back or mixing down to 16 bits, and the Bit Meter shows that more than 16 bits are used, you should apply dithering.

- To see the "actual" resolution of an Audio File. For example, even though a file is in 24 bit format, only 16 bits may be used. Or, a 32 bit file may only use 24 bits (in this case, the "below" segment would not be lit). For this purpose, the Bit Meter is best used in "Analyze selection" mode.

- To see whether a "zeroed" plug-in still affects your signal, or whether a plug-in uses 16 bit internal processing, or more generally, to detect any signal modifier stage in the audio chain between reading and playing.

To adjust the Bit Meter settings, select "Settings..." from the Options pop-up menu or by using the icon.

The Bit meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Dithering

6.5.5 Spectroscope

The Spectroscope shows a continuous graphical representation of the frequency spectrum, analyzed into 60 separate frequency bands, represented as vertical bars. Peak levels are shown as a short horizontal lines above the corresponding band, indicating recent peak/-maximum values. The Spectroscope offers a quick spectrum overview. For a more detailed analysis of the audio spectrum, use the Spectrometer. The Spectroscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

Spectrometer

6.5.6 Spectrometer

The Spectrometer uses FFT (Fast Fourier Transform) techniques to display a continuous frequency graph, providing a very precise and detailed real-time frequency analysis.

- The current frequency spectrum is shown as a linear graph.
- Spectrum "peaks" are shown as a short horizontal lines, indicating recent peak/maximum values.

Snapshots

By using the "Add snapshot" and "Erase last snapshot" buttons, you can take and erase snapshots of the current spectrum. These will be superimposed over the current peak spectrum graph, in a color you can customize, until you click the icon again to take a new snapshot. One use for this is to check the effects of adding EQ, for example. Up to five snapshots can be superimposed on the display; the sixth snapshot replaces the earliest one in the display, and so on. (Don't confuse the order of snapshots with the numbered buttons - these are Spectrometer presets, see below.)

Zooming

You can adjust the frequency scale and range in the Settings dialog as described below, but it's also possible to temporarily zoom in on a frequency area of interest. This is done by clicking and dragging a rectangle in the Spectroscope display. When you release the mouse button, the display is zoomed in so that the enclosed frequency range fills the window. To return to full-scale display, select "Zoom-out fully" from the Functions menu, or double-click anywhere on the display.

Making settings

You can adjust the behavior and display of the meters as desired, and assign up to five sets of Spectrometer settings to the Preset buttons, for instant access. Open the Settings dialog by selecting "Settings" from the Functions menu or clicking the "tool" icon. Note that you can apply your settings without closing the dialog, by clicking the Apply button.

If you want to store your settings for later use (or assign them to a Preset button), select "Save as..." from the pop-up menu in the lower part of the dialog, and specify a name for the preset in the file dialog that appears. Now, you can choose to make the settings instantly available for selection in the FFT Meter window, by using the "Assign to preset button" submenu on the pop-up menu.

- When you are finished, click OK to close the dialog.

Selecting Spectrometer presets

If you have assigned your settings to the Preset buttons in the Settings dialog, you can quickly switch between different level scales and display modes, by clicking one of the Preset icons [1]-[5], or selecting the desired preset from the Options pop-up menu.

Exporting FFT data as ASCII text

When using the Spectrometer in off-line mode ("Monitor Edit cursor Position" or "Analyze audio election" mode) you can export the displayed FFT data as a text file, by selecting

"Export FFT data as ASCII" from the Options pop-up menu. The resulting text file can then be imported into applications that allow graph plotting from text files (Microsoft Excel, for example).

The Spectrometer can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Spectroscope

6.5.7 Wavescope

The Wavescope meter displays a real-time waveform drawing of the audio signal being monitored.

It can be useful when recording or rendering a file if "Monitor File rendering" mode is active.

Making settings

You can adjust settings for the display via the Wavescope Settings dialog. This can be accessed via **Functions** > **Settings...** menu, or by using the icon. Here you can set various color options for the background, grid and waveform display, as well as setting the waveform rendering speed and vertical zoom. If "Clear waveform when reaching right of pane" is checked the waveform display is cleared each time the cursor reaches the right end of the display. If unchecked, the previous waveform is overwritten.

Tip: Wavescope is a useful meter for visualizing audio during recording. The Wavescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

6.5.8 Timecode

This shared tool window displays the current time of the playback head (or the cursor position if not playing back), in the timecode format currently selected in the Time Format dialog.

The current time can be in a range of standard SMPTE formats, CD or DVD specific formats, or a custom one - see Time Format dialog.

The most common use for this window is for working with video or film. You can use it to display the playback head position of the current Audio File in a SMPTE format to frame level resolution (i.e. hours:minutes:seconds:frames). You can also use it to display CD frames (minutes:seconds:frames) when you are authoring CD or DVDs.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Timecode**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Time Format dialog

CD Frame

SMPTE timecode

Authoring CDs and DVDs

6.5.9 Background task monitor

This specific tool window allows you to view all background rendering processes that are in progress.

You can adjust the priority with which they are processed, pause or cancel them. It is useful if you have a number of lengthy processes underway and want to free up some processing power to focus on editing. You can either lower the priority of a task so it does not use as much of the computer processor's capacity, or pause the task temporarily.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Background task monitor**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window by choosing **Utilities** > **Background task monitor**.

You can choose to automatically open the Background task monitor when a rendering process begins. To enable this option, select the checkbox via **Options** > **Global Preferences** > **Options** > **Make the Background Task Monitor visible when a task starts** . Since this window is a Shared_tool_window, the location where it opens may vary.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Control Window

Rendering

6.5.10 Log window

This shared tool window allows you to view log messages that have been issued by WaveLab.

When using WaveLab's scripting language for example, the *logWindow( )* function outputs messages to this window. There are a number of toggle buttons and these allow you to filter the types of messages displayed.

Note: When using the logWindow( ) method you need to have the "Show informal notes" button selected for the messages to be visible. If no log messages are displayed, make sure all the toggle buttons are selected - ! ⚠ .

You can access this window via **Workspace** > **Shared tool windows** > **Log**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window via **Utilities** > **Log**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Control Window

Specific tool windows

Scripting

6.6 Audio Montage Colors

You can create and assign different color schemes to different Clips and parts of a Clip, to help you find your way through the Montage.

Default colors are provided, but you can also define custom colors. To define a new color scheme, select the color name to be defined (click once and overtype to rename), select the parts to which it is applied, then define the color value using the color wheel (or type a set of RGB - Red, Green, Blue values).

You can use the Copy color and Paste buttons to transfer colors between various parts of the montage. In addition, you can drag groups of color settings onto another location in the list - you are asked to confirm. You could, for example, copy "Custom 1" to "Custom 3" as a time saver, before customizing details of "Custom 3".

Color settings made in this dialog apply to all montages.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

6.7 Audio Montage contextual information

When working in the Audio Montage window there is a multitude of editing actions available. In order to learn them all you can take advantage of the contextual information displayed in the status bar (which is enabled by default). This display at the bottom of the main window shows hints to all the possible actions you can perform as you move the cursor around the main Audio Montage window. The info line will also display positions and lengths when you are moving, copying or resizing Clips.

The object you are directly over is indicated, along with any possible actions you can perform and any modifier keys that will enable further functions. These hints are displayed using icons and text descriptions from left to right.

This option is enabled/disabled in the Audio Montage Workspace via **Options** > **Audio Montage preferences** > **All Audio Montages** > **Display indications of possible actions**.

Icons

The following symbols are used in the contextual information display:

- - if this appears, this means there is a context menu available
- - indicates a left mouse click
- - indicates a double click
- - indicates the object can be dragged in any direction inside the montage
- - indicates the object can be dragged in any direction inside or outside the montage
- - indicates the object can be dragged vertically
- - indicates the object can be dragged horizontally

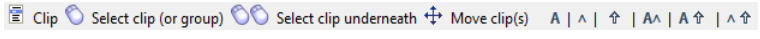
Modifier keys

All hints are displayed along with any possible modifier key combinations. Modifier keys are represented by the following symbols:

- - **[Alt]/[Option]** key
- - **[Shift]** key
- - **[Ctrl]/[Command]** key

Example

From left to right, the example below describes the following possible actions:



- a context menu is available
- a single left mouse click will select the Clip
- a double click will select the Clip underneath (if it exists)

- clicking and dragging will move the click in any direction

There are a number of modifier keys available also including (from left to right):

- the **[Alt]/[Option]** key
- the **[Ctrl]/[Command]** key
- the **[Shift]** key

Various combinations also exist:

- the **[Alt]/[Option]** key + the **[Ctrl]/[Command]** key
- the **[Alt]/[Option]** key + the **[Shift]** key
- the **[Ctrl]/[Command]** key + the **[Shift]** key

Related topics

Chapter 7

Batch Processing

This special Workspace allows you to process any number of audio or audio montage files with Master Section plug-ins and presets, offline effects, and other plug-ins that are unique to Batch Processing. Each file is processed and then saved to a folder of your choice. You can choose a different file format, rename the file according to a set of rules, and run an external application when the batch is finished.

You can process as many files as you wish taking advantage of multi-processing on multi-core processors, if available. You can save batch process files: this allows you to run batches repeatedly if required.

For example, you may have a folder of 24-bit Audio Files which you wish to Normalize, add a fade out to and dither down to 16-bit 44.1kHz. You could save this as a batch process file and then each time you update the original higher sample rate files, you can re-run the batch.

Creating and running a batch process

To set up and run a batch, follow these steps:

- **Create a new batch processor file** - use the **Special** > **Add to Batch Processor** menu entry in another Workspace, or simply use the **File** > **New** command. You can save the newly created batch at any time for later reuse.

- **Add files to the batch** - from within the Batch processor Workspace either use the File Browser, the **Edit** > **Add files from folder...** command, drag a file directly from another Workspace or your computer's file browser, or right-click in the "Files to process" pane and choose **Insert all open Audio Files**.

- **Add plug-ins and effects** - drag any plug-in processors/effects you wish from the Batch Processing Plug-ins to the "Audio Plug-in chain" pane. Right click on any effect (and select **Edit**) to adjust its settings, and drag the plug-ins to change the order in which they are processed.

- **Define an output folder** - from the "Output" tab define a location for the resulting files. In addition you can select "Renaming" to change the name of output files according to a number of conditions. The resulting names are previewed in the "File to process" pane.

- **Choose a file format** - use the "Format" tab to define the file format that files are saved in.

- **Run an external task** - you can choose to run an external application after batch processing has completed using the "External Tool" tab. The external tool must be set up first via the Configure external tools dialog. You can access this from **Tools** > **Configure external tools**.

- **Define any options** - choose any options using the "Options" tab and/or annotate your batch using the "Comments" tab.

- **Choose any Multitasking options** - if you have a multi-core processor in your computer you can also choose how many tasks are completed simultaneously. You can choose how many cores of your processor to dedicate to batch processing, adjusting this setting even during processing.

- **Start processing** - when you are happy with all the settings, choose **Process** > **Start** or click the shortcut button to begin processing the batch. You can monitor the progress of each task (and pause, cancel or reassign priorities on lengthier tasks) using the Background tasks window. From the built-in Progress pane you can choose to pause or cancel the batch at any time.

Batch the batch processor

As you may have noticed in the Plug-ins window, you can access Master Section plug-ins and many of WaveLab's built in offline processes. In addition there are some plug-ins which are unique to the Batch Processor Workspace and each is explained in more detail in Batch Processing Plug-ins. Obviously the order in which you process plug-ins is important. You can change the sequence by dragging the plug-in to the position you wish within the "Audio Plug-in chain" list and releasing the mouse button.

For example, a typical batch processing effects chain might resemble the following:

DC Remover > **Normalizer** > **Denoiser** > **Compressor** > **Limiter** > **MPEG Encoder**

The batch processor operates using a number of processing passes. The number of passes depends on the type of plug-ins you use. Some plug-ins require multiple passes of the Audio File in order to determine how to process the audio, others can process in a single pass. Besides the plug-ins shared with the Master section, there are three other types of plug-in types available in the Batch Processor Workspace:

- **Monopass** - these plug-ins only require one pass when processing.

- **Multipass** - these plug-ins require two or more passes (an analysis and a process pass). The Normalizer is an example of a Multipass effect.

- **Metapass** - these plug-ins are unique to the Batch Processor and require several passes before processing the audio. After analysing ALL Audio Files, a metapass plug-in takes into account the analysis performed by itself on ALL these files, before processing them individually in a second round. The Meta Normalizer is an example of a Metapass effect.

Once all passes have been completed, the resulting file is written to your hard disk. It is useful to note that WaveLab keeps the audio in the 32-bit domain for all calculations before writing a file to disk. To optimize performance, all batch process calculations are performed and stored in RAM rather than using temporary files.

Understanding the signal path

To understand how the signal flow of your plug-in chain is processed you can look closely at the audio signal path display, which consists of a number of colored arrows in columns and rows. Each arrow represents how each plug-in processes any audio passed into it. Each vertical column represents a processing pass. Use the "What's this?" arrow on that column for more information.

Re-using a batch

Once you have created and run a batch file you can re-use it by:

- Choosing **Edit** > **Reset the status of all files** and then running your batch again. This will process all of the files in the batch.
- Alternatively you can selectively re-run a batch process on certain files by right-clicking on the "Files to process" pane, choosing "Reset status" or "Remove" and then running your batch again.

Related topics

Specific tool windows

Shared tool windows

Batch Processing Plug-ins

Batch Conversion

Batch Renaming

7.1 Specific tool windows

Specific tool windows are windows that are specific to the current Workspace. They perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Batch Processing Workspace specific tool windows

The following specific tool windows are available within the Batch Processor Workspace:

Master Project

File Browser

Batch Processing Plug-ins

7.1.1 File Browser

This dialog allows you to browse files directly from within WaveLab rather than using your operating system's file browser.

It provides you with all the standard browsing functions (such as List and Icon views)as well as additional controls to audition Audio Files and any marker defined regions.You can use it to open or insert an entire file or a specific region of a file by dragging and releasing the file in the location you wish to insert the file. In the Audio Montage Workspace you can also choose to only view certain types of WaveLab specific files. The File Browser can be very useful in speeding up the process of auditioning a long list of sound files. It can also be used to drag to windows such as Data CD/DVD, including dragging folders.

The playback functions

Play file Once you have selected a file in the File Browser list, you can audition it by clicking Play in the File Browser's toolbar. Click again to stop.

Auto-play You can also have files play back automatically as soon as they are selected. To do this, activate Auto-Play mode.

Tip: To audition a long list of Audio Files, enable Auto-play and use the UP and DOWN cursor keys to step through each Audio File in quick succession.

Opening and inserting a file or region

Once you have chosen a file you wish to open, double-click it to open the file in the current Workspace. You can also open it by dragging it onto an empty tabbed group, or onto the tab bar of a tabbed group.

You can also drag the file over a wave view to **insert** it at a given point. When you drag a file over a wave view, a beam is displayed so as to clearly display the insertion point.

When browsing Audio Files, the file browser also has a region panel. When you select an Audio File, its related regions are listed in this panel. You can then drag the region from the list, to insert only that part of the audio. Double-clicking on a region has the effect of opening the file in the Workspace and selecting that region. You can access this dialog via **Workspace** > **Specific tool windows** > **File Browser**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

7.1.2 Batch Processing Plug-ins

The Batch Processor Workspace allows you to add a sequence of plug-ins that can be used to process a batch of Audio Files. Some of these plug-ins are standard plug-ins available from the Master Section; others are offline processes available in the Audio File Workspace, and there are others that are only available within batch processing.

To add a plug-in: First create a new Audio Plug-in chain window: click on the Create an empty document icon in the Batch Processors Workspace. Select a category of plug-ins from the Plug-ins accordion (Double-click to add the plug-in at the end, or drag the plug-in into position in the batch processor chain).

To remove a plug-in: Right-click on the plug-in to use the pop-up menu "Remove" option, or drag it back onto the plugin window.

Master Section presets

These are Master Section presets (not to be confused with individual plugin presets, so it's always a chain, even if it's a chain with one member). As these presets can contain multiple effects, they can be put to powerful use in batch processing.

Master Section Plug-ins

These are all the plug-ins available from the Master Section, sorted in the same manner.

Monopass Plug-ins

Monopass plug-ins only require one pass when processing. A Monopass plug-in effect processes the signal then outputs it to any subsequent plug-in. This is like real-time plug-ins.

The following Monopass plug-ins are available in the Batch Processor Workspace:

Audio Injector

Trimmer

Resizer

Fade-In

Instructor

Time Stretch

Pitch Correction

Multipass Plug-ins

Multipass plug-ins require two or more passes (an analysis pass followed by a process pass) before processing the audio. Some are unique to the Batch Processor while others are also found as Offline Processors in the Audio File Workspace.

The following Multipass plug-ins are available in the Batch Processor Workspace:

Normalize Level

Stereo to Mono

Pan Normalizer

Loudness normalizer

Fade-out

DC Remover

Loudness Restorer

Metapass Plug-ins

Metapass plug-ins are unique to the Batch Processor and require several passes before processing the audio. After analyzing the audio, a metapass plug-in takes into account all other plug-ins in the effects chain before processing the audio.

The following Metapass plug-ins are available in the Batch Processor Workspace:

Meta-Normalizer

Meta-Leveler

Related topics

Audio Plug-ins

Offline processing

7.1.2.1 Audio Injector

This dialog is exclusive to the Batch Processor Workspace and allows you to insert an Audio File at the beginning and/or end of the Audio File being processed. The inserted file can also be cross faded with the original Audio File, if required.

You can access this dialog from from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **AudioInjector**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.1.2.2 Trimmer

This dialog is exclusive to the Batch Processor Workspace and allows you to remove a specified duration (from 0ms to 60s) of audio from the head and/or tail of an Audio File.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **Trimmer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.1.2.3 Resizer

This dialog is exclusive to the Batch Processor Workspace and allows you to adjust the duration of an Audio File.

You can set a desired length of file and choose whether to insert silence after the end of the chosen duration.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **Resizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.1.2.4 Fade-In/Out

This dialog is exclusive to the Batch Processor Workspace and allows you to fade the beginning (Fade-In) or the end (Fade-Out) of a batch audio file. You can choose the length and shape of the fade, its duration and the gain you wish it to start/end at.

You can access the fade out dialog from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Fade-Out**.

You can access the fade in dialog from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **Fade-In**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.1.2.5 Instructor

This dialog is exclusive to the Batch Processor Workspace and is a special utility plug-in that allows you to "instruct" the next plug-in in the batch with information about the audio it needs to process. This is useful for situations where you want to use plug-ins that require an analysis stage. In effect, the Instructor plug-in turns a monopass plug-in into a dual pass one.

Some Monopass plugins, such as Denoisers or DeBuzzer, need to learn about the audio they are to process before they can begin processing correctly. The "Instructor" plugin can help in this situation, since it can "teach" the next plug-in in the audio chain about the audio it is about to process.

The Instructor plugin **must** be used as a pair:

1. The first instance replicates the start of the audio stream. This means the next plug-in in the chain will receive the start of the audio stream twice.

2. The second instance of the plug-in comes after the plugin being "instructed". It simply cuts out the extra audio injected by the first instance of the Instructor plug-in.

This means, for example, that the Denoiser plug-in has time to sufficiently analyze the audio stream before the second stream start is injected. The "badly" processed first part of the stream is skipped by the second instance of the Instructor plug-in.

You can set the Instructor plug-in to replicate up to 20 seconds of audio. Be careful not to set a value that is longer than the shortest file in the batch, otherwise a short file will be over truncated by the second instance of the plug-in.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **Instructor**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.1.2.6 Time Stretch

This dialog is exclusive to the Batch Processor Workspace and allows you to change the duration of an audio selection (usually without changing its pitch). For more information on the Time Stretch dialog parameters see Time stretching.

You can access this dialog from **Batch Processing Workspace** > **Plug-ins window** > **Monopass Plug-ins** > **Time Stretch**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Time stretching

Batch Processing Plug-ins

7.1.2.7 Pitch Correction

This dialog is exclusive to the Batch Processor Workspace and allows you to detect and change the pitch of a sound without affecting its length. For more information on the Pitch Correction dialog parameters see Pitch correction.

You can access this dialog from **Batch Processing Workspace** > **Plug-ins window** > **MonoPass Plug-ins** > **Pitch Correction**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Pitch correction

Batch Processing Plug-ins

7.1.2.8 Normalize Level

This dialog allows you to change the peak level of your Audio File.

Enter the desired peak level (in dB) that you wish the audio selection to have. You can also use "Find current peak level" to obtain a report on the peak level of the current audio selection (or the peak level in the whole file, if "Process whole file if there is no selection" is selected in the preferences). You can choose to apply the same gain to both stereo channels (Stereo Link) or mix down to a mono file (Mix to Mono) with the assurance that no clipping will occur when both channels are mixed together.

To normalize the selected audio, enter the desired peak level and click Apply.

You can access this dialog by choosing **Process** > **Level Normalizer...** in the Audio File Workspace.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Value editing

7.1.2.9 Stereo to Mono

This dialog is exclusive to the Batch Processor Workspace and allows you to mix a stereo signal down to a mono one while being certain not to clip while mixing channels.

You can choose to use the same peak level that the stereo file contains, or set the gain to be applied and/or the maximum level to be reached in the resulting mono file.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **Stereo to Mono**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.1.2.10 Pan Normalizer

This dialog allows you to ensure that both channels of a stereo file have the same level or loudness. Since loudnesss is more important than peak volume in the perception of sound,

this is a powerful tool to get the best possible stereo balance.

This process uses two passes to first analyze the audio and then render any level changes required.

You must have a stereo selection in a stereo file to apply this process.

You can access this dialog from the Audio File Workspace via **Process** > **Pan Normalizer...**.

This function is also available as a batch processing plug-in. You can access it from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **Pan Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

7.1.2.11 Loudness normalizer

This dialog allows you to adjust the loudness of a file. Because it relates to loudness as opposed to maximum peak level, this tool works quite differently from the Level Normalizer. A typical application is to specify a loudness, for example -12dB, and the Loudness Normalizer processes the audio to match this level of loudness.

Like Gain Change, increasing the loudness to a certain value can introduce clipping. To avoid this occurring, a peak limiter (the Peak Master plug-in) can be used as part of the process. The Loudness Normalizer raises the loudness and limits peaks in the signal at the same time (if needed), to achieve the target loudness.

In the case of stereo files, both channels are processed independently. This process happens over several stages; analysis precedes the final rendering. The Loudness Normalizer dialog gives access to useful statistics about the file and the Loudness distribution tool, and you can choose to remove any DC offset in the file.

You can access this dialog from the Audio File Workspace via **Process** > **Loudness normalizer...**.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Loudness Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

DC Offset

Loudness distribution

7.1.2.12 DC Remover

This dialog is exclusive to the Batch Processor Workspace and allows you to eliminate any DC Offset from an Audio File.

It can be useful to apply this plug-in first in a batch before other plug-ins to avoid further processing a file containing any DC Offset. For example, an Audio File that has a DC offset will not be at its loudest possible volume when normalized, because the offset consumes headroom.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **DC Remover**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

DC Offset

7.1.2.13 Loudness Restorer

This dialog is exclusive to the Batch Processor Workspace and allows you to restore the loudness of an Audio File.

The Loudness Restorer plug-in **must** be used as a pair - the first instance is used to read the audio signal, the second instance uses this information to adjust the loudness or peak level of the file.

To use the plug-in:

1. Place an instance of the plug-in in the batch chain and choose "Capture" as the mode.
2. Place a second instance of the plug-in after the first one and choose either "Restore peak level" or "Restore loudness (RMS)".

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **Loudness Restorer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.1.2.14 Meta-Normalizer

This dialog is exclusive to the Batch Processor Workspace and allows you to normalize a batch of files all to the same level.

The core purpose of this plug-in is to achieve the **same** loudness in all files (the highest loudness found, if possible), while being certain that no file will clip. For each file, a specific gain is computed by the plug-in once all files have been analyzed and prior to actually applying any gain to achieve the common loudness. If it is not possible to match the highest found loudness, the level of the file with the highest loudness is reduced, so that other files can get the same loudness.

You can choose between peak level or loudness (RMS) as a method of determing the "equal" level between all files in the batch. This "equal" level is then used to apply a gain to each file in the batch.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Metapass Plug-ins** > **Meta-Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.1.2.15 Meta-Leveler

This dialog is exclusive to the Batch Processor Workspace and allows you to match the level of a batch of files.

The core purpose of this plug-in is to to apply the **same** gain to all files, while being certain that a specific peak level will never be exceeded in any file. The unique gain that you wish to apply is (possibly) reduced by the plug-in, once all files in the batch have been analyzed, and prior to actually applying the gain across the batch.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Metapass Plug-ins** > **Meta-Leveler**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.2 Shared tool windows

Shared tool windows are windows that can only exist in one place at a time and are shared between Workspaces. They perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Shared tool windows:

Background task monitor

Log window

Related topics

About Tool Windows

Docking windows

7.2.1 Background task monitor

This specific tool window allows you to view all background rendering processes that are in progress.

You can adjust the priority with which they are processed, pause or cancel them. It is useful if you have a number of lengthy processes underway and want to free up some processing power to focus on editing. You can either lower the priority of a task so it does not use as much of the computer processor's capacity, or pause the task temporarily.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Background task monitor**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window by choosing **Utilities** > **Background task monitor**.

You can choose to automatically open the Background task monitor when a rendering process begins. To enable this option, select the checkbox via **Options** > **Global Preferences** > **Options** > **Make the Background Task Monitor visible when a task starts** . Since this window is a Shared_tool_window, the location where it opens may vary.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Control Window

Rendering

7.2.2 Log window

This shared tool window allows you to view log messages that have been issued by WaveLab.

When using WaveLab's scripting language for example, the *logWindow( )* function outputs messages to this window. There are a number of toggle buttons and these allow you to filter the types of messages displayed.

Note: When using the logWindow( ) method you need to have the "Show informal notes" button selected for the messages to be visible. If no log messages are displayed, make sure all the toggle buttons are selected - ! ⚠ .

You can access this window via **Workspace** > **Shared tool windows** > **Log**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window via **Utilities** > **Log**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Control Window

Specific tool windows

Scripting

7.3 Batch Processing Plug-ins

The Batch Processor Workspace allows you to add a sequence of plug-ins that can be used to process a batch of Audio Files. Some of these plug-ins are standard plug-ins available from the Master Section; others are offline processes available in the Audio File Workspace, and there are others that are only available within batch processing.

To add a plug-in: First create a new Audio Plug-in chain window: click on the Create an empty document icon in the Batch Processors Workspace. Select a category of plug-ins from the Plug-ins accordion (Double-click to add the plug-in at the end, or drag the plug-in into position in the batch processor chain).

To remove a plug-in: Right-click on the plug-in to use the pop-up menu "Remove" option, or drag it back onto the plugin window.

Master Section presets

These are Master Section presets (not to be confused with individual plugin presets, so it's always a chain, even if it's a chain with one member). As these presets can contain multiple effects, they can be put to powerful use in batch processing.

Master Section Plug-ins

These are all the plug-ins available from the Master Section, sorted in the same manner.

Monopass Plug-ins

Monopass plug-ins only require one pass when processing. A Monopass plug-in effect processes the signal then outputs it to any subsequent plug-in. This is like real-time plug-ins.

The following Monopass plug-ins are available in the Batch Processor Workspace:

Audio Injector

Trimmer

Resizer

Fade-In

Instructor

Time Stretch

Pitch Correction

Multipass Plug-ins

Multipass plug-ins require two or more passes (an analysis pass followed by a process pass) before processing the audio. Some are unique to the Batch Processor while others are also found as Offline Processors in the Audio File Workspace.

The following Multipass plug-ins are available in the Batch Processor Workspace:

Normalize Level

Stereo to Mono

Pan Normalizer

Loudness normalizer

Fade-out

DC Remover

Loudness Restorer

Metapass Plug-ins

Metapass plug-ins are unique to the Batch Processor and require several passes before processing the audio. After analyzing the audio, a metapass plug-in takes into account all other plug-ins in the effects chain before processing the audio.

The following Metapass plug-ins are available in the Batch Processor Workspace:

Meta-Normalizer

Meta-Leveler

Related topics

Audio Plug-ins

Offline processing

7.3.0.1 Audio Injector

This dialog is exclusive to the Batch Processor Workspace and allows you to insert an Audio File at the beginning and/or end of the Audio File being processed. The inserted file can also be cross faded with the original Audio File, if required.

You can access this dialog from from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **AudioInjector**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.3.0.2 Trimmer

This dialog is exclusive to the Batch Processor Workspace and allows you to remove a specified duration (from 0ms to 60s) of audio from the head and/or tail of an Audio File.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **Trimmer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.3.0.3 Resizer

This dialog is exclusive to the Batch Processor Workspace and allows you to adjust the duration of an Audio File.

You can set a desired length of file and choose whether to insert silence after the end of the chosen duration.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **Resizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.3.0.4 Fade-In/Out

This dialog is exclusive to the Batch Processor Workspace and allows you to fade the beginning (Fade-In) or the end (Fade-Out) of a batch audio file. You can choose the length and shape of the fade, its duration and the gain you wish it to start/end at.

You can access the fade out dialog from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Fade-Out**.

You can access the fade in dialog from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **Fade-In**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.3.0.5 Instructor

This dialog is exclusive to the Batch Processor Workspace and is a special utility plug-in that allows you to "instruct" the next plug-in in the batch with information about the audio it needs to process. This is useful for situations where you want to use plug-ins that require an analysis stage. In effect, the Instructor plug-in turns a monopass plug-in into a dual pass one.

Some Monopass plugins, such as Denoisers or DeBuzzer, need to learn about the audio they are to process before they can begin processing correctly. The "Instructor" plugin can help in this situation, since it can "teach" the next plug-in in the audio chain about the audio it is about to process.

The Instructor plugin **must** be used as a pair:

1. The first instance replicates the start of the audio stream. This means the next plug-in in the chain will receive the start of the audio stream twice.
2. The second instance of the plug-in comes after the plugin being "instructed". It simply cuts out the extra audio injected by the first instance of the Instructor plug-in.

This means, for example, that the Denoiser plug-in has time to sufficiently analyze the audio stream before the second stream start is injected. The "badly" processed first part of the stream is skipped by the second instance of the Instructor plug-in.

You can set the Instructor plug-in to replicate up to 20 seconds of audio. Be careful not to set a value that is longer than the shortest file in the batch, otherwise a short file will be over truncated by the second instance of the plug-in.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Monopass Plug-ins** > **Instructor**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.3.0.6 Time Stretch

This dialog is exclusive to the Batch Processor Workspace and allows you to change the duration of an audio selection (usually without changing its pitch). For more information on the Time Stretch dialog parameters see Time stretching.

You can access this dialog from **Batch Processing Workspace** > **Plug-ins window** > **Monopass Plug-ins** > **Time Stretch**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Time stretching

Batch Processing Plug-ins

7.3.0.7 Pitch Correction

This dialog is exclusive to the Batch Processor Workspace and allows you to detect and change the pitch of a sound without affecting its length. For more information on the Pitch Correction dialog parameters see Pitch correction.

You can access this dialog from **Batch Processing Workspace** > **Plug-ins window** > **MonoPass Plug-ins** > **Pitch Correction**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Pitch correction

Batch Processing Plug-ins

7.3.0.8 Normalize Level

This dialog allows you to change the peak level of your Audio File.

Enter the desired peak level (in dB) that you wish the audio selection to have. You can also use "Find current peak level" to obtain a report on the peak level of the current audio selection (or the peak level in the whole file, if "Process whole file if there is no selection" is selected in the preferences). You can choose to apply the same gain to both stereo channels (Stereo Link) or mix down to a mono file (Mix to Mono) with the assurance that no clipping will occur when both channels are mixed together.

To normalize the selected audio, enter the desired peak level and click Apply.

You can access this dialog by choosing **Process** > **Level Normalizer...** in the Audio File Workspace.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Value editing

7.3.0.9 Stereo to Mono

This dialog is exclusive to the Batch Processor Workspace and allows you to mix a stereo signal down to a mono one while being certain not to clip while mixing channels.

You can choose to use the same peak level that the stereo file contains, or set the gain to be applied and/or the maximum level to be reached in the resulting mono file.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **Stereo to Mono**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.3.0.10 Pan Normalizer

This dialog allows you to ensure that both channels of a stereo file have the same level or loudness. Since loudnesss is more important than peak volume in the perception of sound, this is a powerful tool to get the best possible stereo balance.

This process uses two passes to first analyze the audio and then render any level changes required.

You must have a stereo selection in a stereo file to apply this process.

You can access this dialog from the Audio File Workspace via **Process** > **Pan Normalizer...**.

This function is also available as a batch processing plug-in. You can access it from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **Pan Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

7.3.0.11 Loudness normalizer

This dialog allows you to adjust the loudness of a file. Because it relates to loudness as opposed to maximum peak level, this tool works quite differently from the Level Normalizer. A typical application is to specify a loudness, for example -12dB, and the Loudness Normalizer processes the audio to match this level of loudness.

Like Gain Change, increasing the loudness to a certain value can introduce clipping. To avoid this occurring, a peak limiter (the Peak Master plug-in) can be used as part of the process. The Loudness Normalizer raises the loudness and limits peaks in the signal at the same time (if needed), to achieve the target loudness.

In the case of stereo files, both channels are processed independently. This process happens over several stages; analysis precedes the final rendering. The Loudness Normalizer dialog gives access to useful statistics about the file and the Loudness distribution tool, and you can choose to remove any DC offset in the file.

You can access this dialog from the Audio File Workspace via **Process** > **Loudness normalizer...**.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Loudness Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

DC Offset

Loudness distribution

7.3.0.12 DC Remover

This dialog is exclusive to the Batch Processor Workspace and allows you to eliminate any DC Offset from an Audio File.

It can be useful to apply this plug-in first in a batch before other plug-ins to avoid further processing a file containing any DC Offset. For example, an Audio File that has a DC offset will not be at its loudest possible volume when normalized, because the offset consumes headroom.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **DC Remover**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

DC Offset

7.3.0.13 Loudness Restorer

This dialog is exclusive to the Batch Processor Workspace and allows you to restore the loudness of an Audio File.

The Loudness Restorer plug-in **must** be used as a pair - the first instance is used to read the audio signal, the second instance uses this information to adjust the loudness or peak level of the file.

To use the plug-in:

1. Place an instance of the plug-in in the batch chain and choose "Capture" as the mode.
2. Place a second instance of the plug-in after the first one and choose either "Restore peak level" or "Restore loudness (RMS)".

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **Loudness Restorer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.3.0.14 Meta-Normalizer

This dialog is exclusive to the Batch Processor Workspace and allows you to normalize a batch of files all to the same level.

The core purpose of this plug-in is to achieve the **same** loudness in all files (the highest loudness found, if possible), while being certain that no file will clip. For each file, a specific gain is computed by the plug-in once all files have been analyzed and prior to actually applying any gain to achieve the common loudness. If it is not possible to match the highest found loudness, the level of the file with the highest loudness is reduced, so that other files can get the same loudness.

You can choose between peak level or loudness (RMS) as a method of determing the "equal" level between all files in the batch. This "equal" level is then used to apply a gain to each file in the batch.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Metapass Plug-ins** > **Meta-Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

7.3.0.15 Meta-Leveler

This dialog is exclusive to the Batch Processor Workspace and allows you to match the level of a batch of files.

The core purpose of this plug-in is to to apply the **same** gain to all files, while being certain that a specific peak level will never be exceeded in any file. The unique gain that you wish to apply is (possibly) reduced by the plug-in, once all files in the batch have been analyzed, and prior to actually applying the gain across the batch.

You can access this dialog from the Batch Processor Workspace via **Plug-ins window** > **Metapass Plug-ins** > **Meta-Leveler**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Processing Plug-ins

Chapter 8

Podcasts

About Podcasts

Once set up, you can produce high quality audio podcast episodes directly from WaveLab.

Audio podcasts are ordered and commented sets of digital Audio Files, generally in the form of 'episodes' with a consistent theme or presenter. Episodes may be downloaded from the Internet, usually via an automated feed. Compression is an important consideration in exporting an audio podcast file, since the listener is likely to encounter bandwidth bottlenecks, so the file formats .mp4a, .mp3 and Ogg Vorbis are commonly used for podcasts.

The Podcast window

The Podcast window is divided into two panes. The upper pane shows the information for the Feed or an Episode, depending on the item selected in the list below. This is where you will be adding files, Internet links or textual information to the Podcast feed and its episodes. The lower pane shows an item list of the basic feed and all Episodes included in the Podcast.

Adding, moving and deleting episodes

There are two ways to add an episode to a Podcast:

- Select "New" from the Podcast window's Episodes menu. This will add a new untitled episode without any information being present.

- Select an existing episode in the Podcast window's item list, then select "Duplicate" from the Episodes menu. This adds a new episode, copying all information from the existing episode to the new one. Episodes can be re-ordered using the "Move up/down" functions in the Episodes menu, or by dragging them to a new location in the item list. See Dragging operations for more information.

To delete an episode, select it in the item list and use the "Delete selected" function in the Episodes menu. Alternatively, you can exclude an Episode from the Podcast by unchecking the "Enabled" box.

There must be at least one episode present in the Podcast.

Adding information to the Podcast

The upper pane of the Podcast window shows various fields and settings that are related to the basic Feed or Episode, depending on what you have selected in the lower pane's item list (the fields are slightly different for the Feed and Episodes).

Field labels in bold letters point at a field that is mandatory to fill. For quick publishing of your Podcast it is sufficient to fill these fields only.

The following settings are available for the Feed and Episode items:

- **Title**
  This sets the title of the feed (e.g. the topic of your podcast). This is a mandatory field for Feed and Episode.

- **Description**
  This gives space for a further description of the feed content. This is a mandatory field for Feed and Episode.

- **Internet link (URL)**
  Your feed's main link presented to the user. Use this to direct people to a certain website that is related to your feed. This is a mandatory field. Clicking will open the specified URL in your system's default Internet browser, so make sure that the link is correct. This is a mandatory field for Feed.

- **Picture (Feed only)**
  This picture cannot be larger than 144 x 400 pixels according to the RSS standard, so the picture is automatically resized if required. Clicking will open the specified picture in your system's default image viewer.

- **Audio file (Episode only)**
  This sets the path to the Audio File you want to add to the episode. The Audio File could be any file type that is supported by the web browser's media reader, but an .mp3 file is best for compatibility. Clicking the button lists the Audio Files already open in WaveLab. Select one for your episode. Alternatively, you can drag the icon of an Audio File into the Audio File pane. Clicking opens the specified file in your system's default media player or viewer, as a preview or check.

- **Publication date and time**
  This sets the publication date and time of the Feed or Episode. Clicking the "Now" button will transfer your system's current date and time.

For the Feed, if "As most recent episode" is checked, this automatically matches the date and time of the most recent Episode.

Related topics

FTP site

Episodes menu

Publish menu

Specific tool windows

8.1 FTP site

This dialog allows you to enter information for uploading your Podcast and all related files to an online FTP server.

You must specify all log-in details for your FTP site, the relative path to your podcast XML file and the web site address where the link to your Podcast is found, including the path to the feed. If in doubt about your FTP settings they can normally be supplied by your ISP.

You can access this dialog via **Publish** > **FTP Site...** in the Podcast Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Podcasts

8.2 Episodes menu

This menu provides commands for managing individual Episodes within a Podcast.

- **New**
  This adds a new untitled episode without any information being present.

- **Duplicate selected**
  This adds a new episode, copying all information from the existing episode to the new one.

- **Delete selected**
  To delete an episode, select it in the item list and use this function. Alternatively, you can exclude an Episode from the Podcast by unchecking the "Enabled" box.

- **Move up/down**
  Episodes can be re-ordered using this function, or by dragging them to a new position in the item list.

8.3 Publish menu

This is where you define where your Podcast is going to be uploaded to via FTP.

FTP site

The WaveLab's FTP site manager stores all information required for the Podcast upload process. Selecting "FTP site..." from the Podcast window's Publish menu brings up the FTP site dialog. See FTP site for more information.

Uploading the Podcast

Once you have specified the FTP site settings, uploading the Podcast is rather simple by using the functions in the Podcast window's Publish menu.

- **Update all items on FTP**

  This uploads/updates the XML Podcast file on the FTP; it also uploads all items' media files, but only if they are not already present on the FTP. This is the most common function to upload and update your Podcast.

- **Upload/Replace all items on FTP**

  This is the same as above, but it always uploads/replaces all of the media files belonging to the item. This is useful if you have changed the audio data, for example.

- **Update selected item on FTP**

  This uploads/updates the XML Podcast file on the FTP. It also uploads the media file of the currently selected item in the list, but only if it is not already present on the FTP.

- **Upload/Replace selected item on FTP**

  This is the same as above, but it always uploads/replaces the media file of the currently selected item in the list. This is useful if you have changed the audio data, for example.

- **View published Podcast**

  This opens your Podcast (from the URL specified in your FTP site settings), using your default browser.

FTP site

This is where you specify your Podcast FTP settings. See FTP site.

Tip

Please note that "all items" in fact means all items that have their "Enabled" box in the item list checked. If not enabled, these episodes are simply omitted from the Podcast.

Related topics

FTP site

8.4 Specific tool windows

Specific tool windows are windows that are specific to the current Workspace. They perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Podcast Workspace specific tool windows

The following specific tool windows are available within the Podcast Workspace:

Master Project

File Browser

8.4.1 File Browser

This dialog allows you to browse files directly from within WaveLab rather than using your operating system's file browser.

It provides you with all the standard browsing functions (such as List and Icon views)as well as additional controls to audition Audio Files and any marker defined regions.You can use it to open or insert an entire file or a specific region of a file by dragging and releasing the file in the location you wish to insert the file. In the Audio Montage Workspace you can also choose to only view certain types of WaveLab specific files. The File Browser can be very useful in speeding up the process of auditioning a long list of sound files. It can also be used to drag to windows such as Data CD/DVD, including dragging folders.

The playback functions

Play file Once you have selected a file in the File Browser list, you can audition it by clicking Play in the File Browser's toolbar. Click again to stop.

Auto-play You can also have files play back automatically as soon as they are selected. To do this, activate Auto-Play mode.

Tip: To audition a long list of Audio Files, enable Auto-play and use the UP and DOWN cursor keys to step through each Audio File in quick succession.

Opening and inserting a file or region

Once you have chosen a file you wish to open, double-click it to open the file in the current Workspace. You can also open it by dragging it onto an empty tabbed group, or onto the tab bar of a tabbed group.

You can also drag the file over a wave view to **insert** it at a given point. When you drag a file over a wave view, a beam is displayed so as to clearly display the insertion point.

When browsing Audio Files, the file browser also has a region panel. When you select an Audio File, its related regions are listed in this panel. You can then drag the region from the list, to insert only that part of the audio. Double-clicking on a region has the effect of opening the file in the Workspace and selecting that region. You can access this dialog via **Workspace** > **Specific tool windows** > **File Browser**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Chapter 9

Control Window

This is a special Workspace that allows you to create your own control window layout. This window does not edit any files, but is used as a placeholder to organize certain tool windows (for example a selection of meters). This is especially useful if you have multiple monitors: you can place the Control Window on your secondary display and use it to manage all the tool windows you commonly refer to.

This window has some special features:

- the ability to host sub-windows that monitor events happening in other windows. You can dock the following windows: All Meters, Master Section, Master Project, Log window, Background task monitor, Import Audio CD and Data CD/DVD.
- if you work with multiple monitors, the Control Window can occupy a dedicated screen.
- the ability to organize all your audio meters in one place, as an assembly of tabbed sub-windows.

Adding Tabbed Groups

The Control Window uses the tab group system to allow for sophisticated panel arrangements. You can arrange tabs horizontally or vertically, split and divide them. For setups with a single monitor, independent floating windows are recommended for small screens, and docked windows for large screens.

To setup a number of tab groups in the Control Window:

- Choose a predefined arrangement of tabs from **Workspace** > **Tab Group Shortcuts** >
- Add an individual Tab Group using the menu commands **Workspace** > **Add Tab Group at right** or **Workspace** > **Add Tab Group at left**
- or to sub-divide a window, right-click on any empty pane and choose either "Split horizontally" or "Split vertically"

Docking/Undocking windows

The process for docking/undocking windows in the Control Window is slightly different to that in other Workspaces. To dock a window in the Control Window:

1. Undock the window so it is floating as an independent window.
2. Choose a pane in which you want to place the window. This is done by clicking on an empty pane; an arrow icon will appear to indicate it is selected.
3. Choose the following command from the floating window's menu: **Window** > **Place in Control window**.

You can also simply drag a tab from one tab group to another, to move it there.

Once docked in the Control Window you can move a window around by dragging it to any other free space in the control window.

To undock a window from the Control Window either:

- right click on the tab title bar or the window itself and choose "Set as independent floating window" from the context menu or,
- click on the button at the top right of the Control Window

For more information on docking/undocking windows see Docking windows.

Related topics

Specific tool windows

Shared tool windows

Metering

Master Section

Master Project

Scripting

Tab Groups

9.1 Specific tool windows

Specific tool windows are windows that are specific to the current Workspace. They perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Control Window specific tool window

The following specific tool window is available within the Control Window:

Master Project

Related topics

About Tool Windows

Docking windows

9.2 Shared tool windows

Shared tool windows are windows that can only exist in one place at a time and are shared between Workspaces. They perform useful tasks in relationship to the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Shared tool windows:

Phasescope

VU Meter

Oscilloscope

Bit Meter

Spectroscope

Spectrometer

Wavescope

Timecode

Background task monitor

Log window

Master Section

Related topics

About Tool Windows

Metering

Docking windows

9.2.1 Phasescope

The Phasescope indicates the phase and amplitude relationship between two stereo channels. It is only really relevant when monitoring stereo audio material.

Reading the Phasescope

It can be interpreted as follows:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).

- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.

- A random but roughly elliptical shape indicates a well balanced stereo signal. If the shape "leans" to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the Phasescope will show a straight line, angled 45 degrees to the other side).

- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90 degrees on the other.

- Generally, the more you can see a "thread" shape, the more bass there is in the signal; the more "spray-like" the display, the more high frequencies are in the signal.

Phase Correlation meter

At the bottom of the display, you will find a Phase Correlation meter, which shows the same information but in a different way:

- The green line shows the current phase correlation, while the two red lines show the recent peak minimum and maximum values, respectively (these are the default colors, which you can change).

- With a mono signal, the meter would show +1, indicating that both channels are perfectly in phase.

- Similarly, -1 indicates that the two channels are the same, but one is inverted.

- Generally, for a "good" mix, the meter should show a value between 0 and +1.

Unlike the main Phasescope, the Phase Correlation meter is also available in "Analyze audio selection" mode, showing an average value for the selected range.

Changing settings

You can define display colors, peak hold time, and the resolution or number of samples to display from the **Functions** > **Settings...** menu.

The Phasescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

9.2.2 VU Meter

The vu-meter is used to display the peak and average loudness/decibel level of your Audio File. It also displays the pan, or the balance between the left and right channels in a stereo file.

The upper part of the level/pan meter window shows the peak level and average loudness, in the following way:

- The Peak Level meters display the peak levels of each channel, graphically and numerically. By default, the meter segments and numerical peak values are displayed in green for low levels, yellow for levels between -6dB and -2dB, and red for levels above -2dB. You can change both the colors and the range boundaries if you wish via the Level/Pan Meter Settings dialog. This can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

- The VU (Volume Unit) meters measure the average loudness (RMS) of each channel. These meters have a built-in inertia, evening out loudness variations over a user-defined time span. If you are monitoring playback or audio input, you will also note two vertical lines following each VU meter bar, seemingly "trying to reach" the current RMS value. These lines indicate the average of the most recent minimum RMS values (left line) and the average of the most recent maximum RMS values (right line). To the left, the difference between the minimum and maximum average values is displayed (the level value in brackets) - this gives you an overview of the dynamic range of the audio material.

- If you are monitoring real-time audio (playback or input), the maximum peak and loudness values are displayed numerically to the right of the meter bars. Numbers in brackets to the right of the Maximum Peak values indicate the number of successive clips (0dB signal peaks). Recording levels should be set so that they only rarely clip. If the master level is set too high, the sound quality and frequency response will be compromised at high recording levels, with unwanted clipping effects. If the level is set too low, noise levels may be high relative to the main sound being recorded.

Pan meters

The lower part of the window shows the pan (the difference in level between the left and right channel, only applicable when monitoring stereo audio):

- The upper pan meters show the peak level difference between the channels, graphically and numerically. Note that the pan meters are "two-sided"; the level bars can go to the left or right, indicating which channel is the loudest. The two sides are shown in different colors (which can be changed via the Settings dialog as explained previously).

- The lower pan meters show the average difference in loudness between the channels, in a similar way. This gives you a visual indication of whether a stereo recording is properly centered, for example.

- If you are monitoring real-time audio (playback or input), the maximum balance difference value (peak and loudness) for each channel is displayed numerically to the left and right of the meter bars. The VU meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces.It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

9.2.3 Oscilloscope

The Oscilloscope offers a highly magnified view of the waveform around the playback cursor position.

If you are analyzing a stereo file, the Oscilloscope normally shows the separate levels of the two channels. However, if you activate the option "Show Sum and Subtraction" on the Functions menu (or click the +/- icon), the upper half of the Oscilloscope shows the sum of the two channels and the lower half shows the subtraction.

Making settings

By opening the Settings dialog you can adjust the display colors, and choose to activate Auto-zoom or not. With Auto-zoom activated, the display is optimized so that the highest level reaches the top of the display at all times. The Settings dialog can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

The Oscilloscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

9.2.4 Bit Meter

The Bit Meter shows the resolution or the number of bits used in the digital audio being monitored. Normally the maximum number of bits in an Audio File is the same as the resolution of the Audio File (e.g. a 16 bit Audio File shows that up to 16 bits are used), but sometimes this is not the case.

As soon as you perform any kind of real-time processing on an Audio File, the audio data is treated at a much higher resolution (32 bit floating point), to allow for pristine audio quality. Examples of such processing include level adjustments, effects, the mixing of two or more files, etc. In fact, the only time when a 16 bit file is played back at 16 bit resolution is if you play it without any fades or effects, and with the Master Faders set to 0.00 (no level adjustment). You can try this out for yourself by playing back a 16 bit Audio File and viewing the Bit Meter - as soon as you adjust the Master Faders, note that 24 bits are used and the "inter" indicator is lit (more on this below).

How to read the Bit Meter

- The innermost meters (closest to the bit scale) show the number of bits in use. You can adjust this display in the Settings dialog.

- The outer meters are "history" meters, showing how many bits were recently in use. You can adjust the hold time in the Settings dialog.
- The "over" segment indicates clipping, similar to a clip indicator.
- If the "below" segment is lit, there are more than 24 bits. The bit meter will show the 24 higher bits, and the "below" segment indicates the existence of extra, lower bits. Note that audio is always processed with more than 24 bits internally.
- If the "inter" segment is lit, this indicates that the audio data cannot really be expressed on a regular 24 bit scale (there are floating point values "in between" bits, thus the label "inter"). This is typically the case if you apply effects; thus the "inter" segment allows you to differentiate processed from non-processed 24 bit PCM files.

When to use the Bit Meter

- To check whether dithering is necessary or not. As a rule, if you're playing back or mixing down to 16 bits, and the Bit Meter shows that more than 16 bits are used, you should apply dithering.
- To see the "actual" resolution of an Audio File. For example, even though a file is in 24 bit format, only 16 bits may be used. Or, a 32 bit file may only use 24 bits (in this case, the "below" segment would not be lit). For this purpose, the Bit Meter is best used in "Analyze selection" mode.
- To see whether a "zeroed" plug-in still affects your signal, or whether a plug-in uses 16 bit internal processing, or more generally, to detect any signal modifier stage in the audio chain between reading and playing.

To adjust the Bit Meter settings, select "Settings..." from the Options pop-up menu or by using the icon.

The Bit meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Dithering

9.2.5 Spectroscope

The Spectroscope shows a continuous graphical representation of the frequency spectrum, analyzed into 60 separate frequency bands, represented as vertical bars. Peak levels are shown as a short horizontal lines above the corresponding band, indicating recent peak/-maximum values. The Spectroscope offers a quick spectrum overview. For a more detailed analysis of the audio spectrum, use the Spectrometer. The Spectroscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

Spectrometer

9.2.6 Spectrometer

The Spectrometer uses FFT (Fast Fourier Transform) techniques to display a continuous frequency graph, providing a very precise and detailed real-time frequency analysis.

- The current frequency spectrum is shown as a linear graph.
- Spectrum "peaks" are shown as a short horizontal lines, indicating recent peak/maximum values.

Snapshots

By using the "Add snapshot" and "Erase last snapshot" buttons, you can take and erase snapshots of the current spectrum. These will be superimposed over the current peak spectrum graph, in a color you can customize, until you click the icon again to take a new snapshot. One use for this is to check the effects of adding EQ, for example. Up to five snapshots can be superimposed on the display; the sixth snapshot replaces the earliest one in the display, and so on. (Don't confuse the order of snapshots with the numbered buttons - these are Spectrometer presets, see below.)

Zooming

You can adjust the frequency scale and range in the Settings dialog as described below, but it's also possible to temporarily zoom in on a frequency area of interest. This is done by clicking and dragging a rectangle in the Spectroscope display. When you release the mouse button, the display is zoomed in so that the enclosed frequency range fills the window. To return to full-scale display, select "Zoom-out fully" from the Functions menu, or double-click anywhere on the display.

Making settings

You can adjust the behavior and display of the meters as desired, and assign up to five sets of Spectrometer settings to the Preset buttons, for instant access. Open the Settings dialog by selecting "Settings" from the Functions menu or clicking the "tool" icon. Note that you can apply your settings without closing the dialog, by clicking the Apply button.

If you want to store your settings for later use (or assign them to a Preset button), select "Save as..." from the pop-up menu in the lower part of the dialog, and specify a name for the preset in the file dialog that appears. Now, you can choose to make the settings instantly available for selection in the FFT Meter window, by using the "Assign to preset button" submenu on the pop-up menu.

- When you are finished, click OK to close the dialog.

Selecting Spectrometer presets

If you have assigned your settings to the Preset buttons in the Settings dialog, you can quickly switch between different level scales and display modes, by clicking one of the Preset icons [1]-[5], or selecting the desired preset from the Options pop-up menu.

Exporting FFT data as ASCII text

When using the Spectrometer in off-line mode ("Monitor Edit cursor Position" or "Analyze audio election" mode) you can export the displayed FFT data as a text file, by selecting "Export FFT data as ASCII" from the Options pop-up menu. The resulting text file can then be imported into applications that allow graph plotting from text files (Microsoft Excel, for example).

The Spectrometer can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Spectroscope

9.2.7 Wavescope

The Wavescope meter displays a real-time waveform drawing of the audio signal being monitored.

It can be useful when recording or rendering a file if "Monitor File rendering" mode is active.

Making settings

You can adjust settings for the display via the Wavescope Settings dialog. This can be accessed via **Functions** > **Settings...** menu, or by using the icon. Here you can set various color options for the background, grid and waveform display, as well as setting the waveform rendering speed and vertical zoom. If "Clear waveform when reaching right of pane" is checked the waveform display is cleared each time the cursor reaches the right end of the display. If unchecked, the previous waveform is overwritten.

Tip: Wavescope is a useful meter for visualizing audio during recording. The Wavescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

9.2.8 Timecode

This shared tool window displays the current time of the playback head (or the cursor position if not playing back), in the timecode format currently selected in the Time Format dialog.

The current time can be in a range of standard SMPTE formats, CD or DVD specific formats, or a custom one - see Time Format dialog.

The most common use for this window is for working with video or film. You can use it to display the playback head position of the current Audio File in a SMPTE format to frame level resolution (i.e. hours:minutes:seconds:frames). You can also use it to display CD frames (minutes:seconds:frames) when you are authoring CD or DVDs.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Timecode**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Time Format dialog

CD Frame

SMPTE timecode

Authoring CDs and DVDs

9.2.9 Background task monitor

This specific tool window allows you to view all background rendering processes that are in progress.

You can adjust the priority with which they are processed, pause or cancel them. It is useful if you have a number of lengthy processes underway and want to free up some processing power to focus on editing. You can either lower the priority of a task so it does not use as much of the computer processor's capacity, or pause the task temporarily.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Background task monitor**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window by choosing **Utilities** > **Background task monitor**.

You can choose to automatically open the Background task monitor when a rendering process begins. To enable this option, select the checkbox via **Options** > **Global Preferences** > **Options** > **Make the Background Task Monitor visible when a task starts** . Since this window is a Shared_tool_window, the location where it opens may vary.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Control Window

Rendering

9.2.10 Log window

This shared tool window allows you to view log messages that have been issued by WaveLab.

When using WaveLab's scripting language for example, the *logWindow( )* function outputs messages to this window. There are a number of toggle buttons and these allow you to filter the types of messages displayed.

Note: When using the logWindow( ) method you need to have the "Show informal notes" button selected for the messages to be visible. If no log messages are displayed, make sure all the toggle buttons are selected - .

You can access this window via **Workspace** > **Shared tool windows** > **Log**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window via **Utilities** > **Log**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Control Window

Specific tool windows

Scripting

9.3 Metering

WaveLab contains a variety of audio meters to help in monitoring and analyzing audio.Meters can be used to monitor audio during playback, rendering, recording and also to analyze a specific selection of audio. There are seven different audio meters in WaveLab, each with its own separate window. The meters are accessed via the Meters menu, the Shared Tool windows menu, or from the Meters control bar.

Monitor Modes

You can choose the audio source and the mode for displaying information in the meters. You can access the following monitoring functions from the Meters menu or via the Meters command bar:

- **Monitor Playback** - the meters display the audio output of the Master Section **after the dithering section, unlike the Master Section's own meter**.
- **Monitor Audio Input** - the meters display the audio input you have chosen in the Audio Streaming Settings. This is useful for monitoring audio while recording.

- **Monitor File Rendering** - the meters display what is being written to disk during file rendering, taking into account Master Section settings, with average, minimum and maximum peak values computed. After rendering, the meter(s) freeze, and remain frozen until you refresh or change the monitor mode.

- **Monitor Edit cursor position** - the meters display static information about the audio beneath the edit cursor. Note that the Master Section settings are not taken into account in this mode.

- **Analyze audio selection** - this allows you to make a selection and have the meters display the average values over the selected range as a static display. The Master Section settings are not taken into account in this mode. When you change the selection, you need to update the meter displays by selecting "Refresh selection analysis" from the Meters menu (or by clicking the Refresh button on the Meters control bar).

If you have one of the continuous display modes chosen, you can also select to freeze the meters at any time using the "Freeze meters" command.

Using meter windows

There can only be one instance of each Audio Meter. For example, if you dock one meter in a Workspace, it will be automatically removed from the place where it was previously located.

Audio Meters can appear in the Audio File, Audio Montage Workspaces as well as the Control Window. They can be used:

- as a docked window in a Workspace
- as a tabbed window in the control window
- as an independent floating window. In this mode, it can be useful to make the window "frameless" from **Window** > **Hide frame**, to save screen space. In this case, the whole menu is accessed by right-clicking.

The axis of most Audio Meters may be rotated, to view the graphics horizontally or vertically.

Some meters can also have their visual style and display parameters customized via their corresponding settings dialog. This can be accessed through the **Functions** > **Settings...** menu in each meter window, or by using the icon.

Related topics

VU Meter

Spectroscope

Oscilloscope

Bit Meter

Phasescope

Spectrometer

Wavescope

9.3.1 VU Meter

The vu-meter is used to display the peak and average loudness/decibel level of your Audio File. It also displays the pan, or the balance between the left and right channels in a stereo file.

The upper part of the level/pan meter window shows the peak level and average loudness, in the following way:

- The Peak Level meters display the peak levels of each channel, graphically and numerically. By default, the meter segments and numerical peak values are displayed in green for low levels, yellow for levels between -6dB and -2dB, and red for levels above -2dB. You can change both the colors and the range boundaries if you wish via the Level/Pan Meter Settings dialog. This can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

- The VU (Volume Unit) meters measure the average loudness (RMS) of each channel. These meters have a built-in inertia, evening out loudness variations over a user-defined time span. If you are monitoring playback or audio input, you will also note two vertical lines following each VU meter bar, seemingly "trying to reach" the current RMS value. These lines indicate the average of the most recent minimum RMS values (left line) and the average of the most recent maximum RMS values (right line). To the left, the difference between the minimum and maximum average values is displayed (the level value in brackets) - this gives you an overview of the dynamic range of the audio material.

- If you are monitoring real-time audio (playback or input), the maximum peak and loudness values are displayed numerically to the right of the meter bars. Numbers in brackets to the right of the Maximum Peak values indicate the number of successive clips (0dB signal peaks). Recording levels should be set so that they only rarely clip. If the master level is set too high, the sound quality and frequency response will be compromised at high recording levels, with unwanted clipping effects. If the level is set too low, noise levels may be high relative to the main sound being recorded.

Pan meters

The lower part of the window shows the pan (the difference in level between the left and right channel, only applicable when monitoring stereo audio):

- The upper pan meters show the peak level difference between the channels, graphically and numerically. Note that the pan meters are "two-sided"; the level bars can go to the left or right, indicating which channel is the loudest. The two sides are shown in different colors (which can be changed via the Settings dialog as explained previously).

- The lower pan meters show the average difference in loudness between the channels, in a similar way. This gives you a visual indication of whether a stereo recording is properly centered, for example.

- If you are monitoring real-time audio (playback or input), the maximum balance difference value (peak and loudness) for each channel is displayed numerically to the left and right of the meter bars. The VU meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces.It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

9.3.2 Spectroscope

The Spectroscope shows a continuous graphical representation of the frequency spectrum, analyzed into 60 separate frequency bands, represented as vertical bars. Peak levels are shown as a short horizontal lines above the corresponding band, indicating recent peak/-maximum values. The Spectroscope offers a quick spectrum overview. For a more detailed analysis of the audio spectrum, use the Spectrometer. The Spectroscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

Spectrometer

9.3.3 Oscilloscope

The Oscilloscope offers a highly magnified view of the waveform around the playback cursor position.

If you are analyzing a stereo file, the Oscilloscope normally shows the separate levels of the two channels. However, if you activate the option "Show Sum and Subtraction" on the Functions menu (or click the +/- icon), the upper half of the Oscilloscope shows the sum of the two channels and the lower half shows the subtraction.

Making settings

By opening the Settings dialog you can adjust the display colors, and choose to activate Auto-zoom or not. With Auto-zoom activated, the display is optimized so that the highest level reaches the top of the display at all times. The Settings dialog can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

The Oscilloscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

9.3.4 Bit Meter

The Bit Meter shows the resolution or the number of bits used in the digital audio being monitored. Normally the maximum number of bits in an Audio File is the same as the resolution of the Audio File (e.g. a 16 bit Audio File shows that up to 16 bits are used), but sometimes this is not the case.

As soon as you perform any kind of real-time processing on an Audio File, the audio data is treated at a much higher resolution (32 bit floating point), to allow for pristine audio quality. Examples of such processing include level adjustments, effects, the mixing of two or more files, etc. In fact, the only time when a 16 bit file is played back at 16 bit resolution is if you play it without any fades or effects, and with the Master Faders set to 0.00 (no level adjustment). You can try this out for yourself by playing back a 16 bit Audio File and viewing the Bit Meter - as soon as you adjust the Master Faders, note that 24 bits are used and the "inter" indicator is lit (more on this below).

How to read the Bit Meter

- The innermost meters (closest to the bit scale) show the number of bits in use. You can adjust this display in the Settings dialog.
- The outer meters are "history" meters, showing how many bits were recently in use. You can adjust the hold time in the Settings dialog.
- The "over" segment indicates clipping, similar to a clip indicator.
- If the "below" segment is lit, there are more than 24 bits. The bit meter will show the 24 higher bits, and the "below" segment indicates the existence of extra, lower bits. Note that audio is always processed with more than 24 bits internally.
- If the "inter" segment is lit, this indicates that the audio data cannot really be expressed on a regular 24 bit scale (there are floating point values "in between" bits, thus the label "inter"). This is typically the case if you apply effects; thus the "inter" segment allows you to differentiate processed from non-processed 24 bit PCM files.

When to use the Bit Meter

- To check whether dithering is necessary or not. As a rule, if you're playing back or mixing down to 16 bits, and the Bit Meter shows that more than 16 bits are used, you should apply dithering.
- To see the "actual" resolution of an Audio File. For example, even though a file is in 24 bit format, only 16 bits may be used. Or, a 32 bit file may only use 24 bits (in this case, the "below" segment would not be lit). For this purpose, the Bit Meter is best used in "Analyze selection" mode.
- To see whether a "zeroed" plug-in still affects your signal, or whether a plug-in uses 16 bit internal processing, or more generally, to detect any signal modifier stage in the audio chain between reading and playing.

To adjust the Bit Meter settings, select "Settings..." from the Options pop-up menu or by using the icon.

The Bit meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Dithering

9.3.5 Phasescope

The Phasescope indicates the phase and amplitude relationship between two stereo channels. It is only really relevant when monitoring stereo audio material.

Reading the Phasescope

It can be interpreted as follows:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.
- A random but roughly elliptical shape indicates a well balanced stereo signal. If the shape "leans" to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the Phasescope will show a straight line, angled 45 degrees to the other side).
- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90 degrees on the other.
- Generally, the more you can see a "thread" shape, the more bass there is in the signal; the more "spray-like" the display, the more high frequencies are in the signal.

Phase Correlation meter

At the bottom of the display, you will find a Phase Correlation meter, which shows the same information but in a different way:

- The green line shows the current phase correlation, while the two red lines show the recent peak minimum and maximum values, respectively (these are the default colors, which you can change).
- With a mono signal, the meter would show +1, indicating that both channels are perfectly in phase.

- Similarly, -1 indicates that the two channels are the same, but one is inverted.
- Generally, for a "good" mix, the meter should show a value between 0 and +1.

Unlike the main Phasescope, the Phase Correlation meter is also available in "Analyze audio selection" mode, showing an average value for the selected range.

Changing settings

You can define display colors, peak hold time, and the resolution or number of samples to display from the **Functions** > **Settings...** menu.

The Phasescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

9.3.6 Spectrometer

The Spectrometer uses FFT (Fast Fourier Transform) techniques to display a continuous frequency graph, providing a very precise and detailed real-time frequency analysis.

- The current frequency spectrum is shown as a linear graph.
- Spectrum "peaks" are shown as a short horizontal lines, indicating recent peak/maximum values.

Snapshots

By using the "Add snapshot" and "Erase last snapshot" buttons, you can take and erase snapshots of the current spectrum. These will be superimposed over the current peak spectrum graph, in a color you can customize, until you click the icon again to take a new snapshot. One use for this is to check the effects of adding EQ, for example. Up to five snapshots can be superimposed on the display; the sixth snapshot replaces the earliest one in the display, and so on. (Don't confuse the order of snapshots with the numbered buttons - these are Spectrometer presets, see below.)

Zooming

You can adjust the frequency scale and range in the Settings dialog as described below, but it's also possible to temporarily zoom in on a frequency area of interest. This is done by clicking and dragging a rectangle in the Spectroscope display. When you release the mouse button, the display is zoomed in so that the enclosed frequency range fills the window. To return to full-scale display, select "Zoom-out fully" from the Functions menu, or double-click anywhere on the display.

Making settings

You can adjust the behavior and display of the meters as desired, and assign up to five sets of Spectrometer settings to the Preset buttons, for instant access. Open the Settings dialog by selecting "Settings" from the Functions menu or clicking the "tool" icon. Note that you can apply your settings without closing the dialog, by clicking the Apply button.

If you want to store your settings for later use (or assign them to a Preset button), select "Save as..." from the pop-up menu in the lower part of the dialog, and specify a name for the preset in the file dialog that appears. Now, you can choose to make the settings instantly available for selection in the FFT Meter window, by using the "Assign to preset button" submenu on the pop-up menu.

- When you are finished, click OK to close the dialog.

Selecting Spectrometer presets

If you have assigned your settings to the Preset buttons in the Settings dialog, you can quickly switch between different level scales and display modes, by clicking one of the Preset icons [1]-[5], or selecting the desired preset from the Options pop-up menu.

Exporting FFT data as ASCII text

When using the Spectrometer in off-line mode ("Monitor Edit cursor Position" or "Analyze audio election" mode) you can export the displayed FFT data as a text file, by selecting "Export FFT data as ASCII" from the Options pop-up menu. The resulting text file can then be imported into applications that allow graph plotting from text files (Microsoft Excel, for example).

The Spectrometer can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Spectroscope

9.3.7 Wavescope

The Wavescope meter displays a real-time waveform drawing of the audio signal being monitored.

It can be useful when recording or rendering a file if "Monitor File rendering" mode is active.

Making settings

You can adjust settings for the display via the Wavescope Settings dialog. This can be accessed via **Functions** > **Settings...** menu, or by using the icon. Here you can set

various color options for the background, grid and waveform display, as well as setting the waveform rendering speed and vertical zoom. If "Clear waveform when reaching right of pane" is checked the waveform display is cleared each time the cursor reaches the right end of the display. If unchecked, the previous waveform is overwritten.

Tip: Wavescope is a useful meter for visualizing audio during recording. The Wavescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Chapter 10

Master Section

The Master section contains WaveLab's "Real Time Engine" and is the final element in the signal path, before your audio material is passed on to the audio hardware, or before it is rendered to disk as an Audio File. The Master section is where you can set the master volume level and add effects processors.

The Master Section is a "shared tool window". This means there is a unique instance of it: it can only show up at one place at a time. You can access it from Workspace menu > Shared tool windows. If the Master section is not visible, click on to dock it in the relevant Workspace.

The Master Section includes:

- **Effects slots** For adding real-time effect plug-in processors such as Crystal Resampler, Steinberg Studio EQ, etc. WaveLab provides a number of effects slots. Click on to access plug-ins. If you need more effects than you have slots available, you can render the tracks and add further effects in a subsequent pass before storing the final Audio File.

Plug-ins can be individually by-passed for playback by toggling to , but this does not affect the rendering process.

To give priority to a single effect (that is, to switch the other effects off temporarily), press 'S' (Solo bypass).

The icon allows you to store, access and manage plug-in presets. For Steinberg VST3 effects, an interesting set of Factory presets are provided from the drop-down submenu.

Other buttons in this row allow you to hide or display the effect's control panel and switch an effect on ( ) or off ( ) - for both playback and rendering.

- **Master level faders** The right and left faders are normally linked, but you can press "Unlink" to control the output levels independently. Select "Mono" to sum the audio channels into a single mono output. "Reset peaks" clears the last peak level indicator.

The tools at the bottom pane are:

- **Smart Bypass** - this opens the Smart bypass dialog, which allows you to choose whether to bypass all the active effects in the Effects slots, including faders. The purpose of this feature is to allow you to compensate for any level difference introduced by the Master Section. **Note that this is for playback only, not for file rendering**.

- **Reset all** - removes all the active effects from the Effects slots and sets the master output to 0dB.
- **Settings menu** - provides access to various options.
- **Rendering functions** - brings up the Render Wave window. This allows you to process the Audio File or Audio Montage with the effects you have selected and create a temporary or final Audio File. By rendering the effects, they become a permanent part of the file, rather than using them in real-time to test a set of effects on the file. The rendering options differ slightly so they are appropriate for Audio Files and Audio Montages.
- **Master section On/Off** - removes the Master section from the signal path. **Note that this is for playback only, not for file rendering**.

Tip: When working in the Master Section window, the mouse wheel can be used to adjust the master volume. You have to point the cursor in the Master Section for this to work.

Docking/Undocking Master Section

The Master Section is a shared tool window and can be docked and un-docked like other Shared tool windows. There are however, some exceptions to its behavior, because of its central importance in WaveLab.

If you dock the Master Section in a Workspace, then close that Workspace, the Master Section will be automatically un-docked and re-displayed as an independent floating window. This is done so that the Master Section remains visible at all times and is not closed with the Workspace (the normal behavior for a Shared tool window). If however, you place it in the Control Window then it is not docked but part of a tab group. As such if you then close the Control Window it will also hide the Master Section.

The "Global" menu (in all menu bars), always has an entry "Master Section", to easily reveal the Master Section if it is ever hidden in this manner. If the Master Section is docked or in a tab group, its main host will be made visible again. The Master Section can be docked in some Workspaces (Audio File, Montage), but not in some others (for example, the Podcast Workspace).

Related topics

Render Wave window

Render Montage window

Plug-ins settings

Audio Plug-ins

10.1 Render Wave window

This dialog allows you to process the Audio File or selection you are working on, applying any effects you have active.

It applies all active plug-ins to the selected audio region or the whole file, and you can optionally change the format of the rendered file. This process is sometimes also referred to as "mixing" or "bouncing down".

You can select options for which regions of the session are rendered, whether to create a new file or process in place, and whether to mute the effects chain when finished. You can also choose to copy marker locations to the new file.

You can access this dialog in the Audio File Workspace via the Render button from the Master Section window.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Rendering

Audio Plug-ins

Audio File Format Dialog

Master Section

10.2 Render Montage window

This dialog allows you to process the montage session you are working on, mixing down multiple tracks into a mono or stereo file and applying any active plug-in effects you have active. This process is sometimes also referred to as mixing or bouncing down.

You can select options for which regions of the session are rendered, whether to create a named file or a temporary, untitled file, and whether to mute the effects chain when finished.

There are various options to render all or parts of the Montage, into one or more files - use "What's this..."

You can access this dialog in the Audio Montage Workspace via the Render button from the Master Section window.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Rendering

Audio Plug-ins

Audio File Format Dialog

Master Section

10.3 Plug-ins settings

This dialog allows you to access a number of options for managing your VST plug-ins.

You can specify where WaveLab should search for your VST plug-ins and which ones it should ignore. It also allows you to choose how your VST plug-in knobs respond to mouse interactions and how frequently they are updated visually.

If you use your own file structure to organize and store VST Plug-ins, this dialog allows you to have full control over which ones are loaded or not. This can be particularly useful if you want to disable a particular plug-in that you suspect of not functioning properly, or to only load a certain set of plug-ins for a specific project, for example.

You can access this dialog from the Master Section via **Options** > **VST Plug-in settings...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Plug-ins

Chapter 11

Interface Elements

This section contains links to all of the various types of windows and dialogs within WaveLab.

Related topics

Dialogs

Shared tool windows

Specific tool windows

11.1 Dialogs

There are many dialog windows within WaveLab that allow you to access all its features and processing power. You can get help on a particular dialog by using the Help button from within the dialog. Using the "What's this?" tool will help you to access detailed information on a dialog's parameters. See Getting Help for more information on.

The following is a list of all the dialogs within WaveLab that have help entries available:

Insert link

Align Clips

Recording channels

Audio CD Report

Audio file comparer

Audio File Format Dialog

Audio Montage properties

Split Clip at silences

Audio properties dialog

Audio Range

Audio Streaming Settings

Auto Split

Batch Conversion
Basic Audio CD Settings
Batch Renaming
CD Text Editor
CD-Text Browser
CD Wizard
Copy audio information
Customize Commands
Data CD/DVD
DVD-Audio options
DVD-Audio Creation
Document list dialog
Ducking options
Effect morphing
Edit playback times
Error detection and correction
Configure external tools
Folder Preferences dialog
Frequency range
FTP site
Change Level
File attributes (OGG/WMA)
Global analysis
Import Audio CD
Insert Audio Files
Level envelope
Loop Tone Uniformizer
Loudness distribution
Loudness normalizer
Global Preferences
Convert marker type
Load Master Section Preset
Save Master Section Preset
Audio Montage Clone
Write Audio CD
Meta-normalizer

Audio Montage Preferences
MP2 encoding
MP3 attributes
MP3 encoding
Normalize Level
OGG encoding
Open files list
Pan Normalizer
Mix dialog
Pitch bend
Pitch correction
Pitch quantize
Playback speed
Podcast options
Text format
Repeat Clip
Recently used files
Recording Dialog
Record create marker
Rename File dialog
Render ASIO input to file
Render Montage window
Render Wave window
Loudness envelope options
Sample attributes
Modified files dialog
Save Audio File dialog
Select where and how Audio Files will be saved
Save Batch Processor
Save File As
Save Preset As
Save picture of active window
Scripting window (Audio Montage Workspace)
Scripting window (Audio File Workspace)
Definition of shortcut(s)
Signal generator

Silence Generator
Smart bypass
Special File Format
Spectrogram options
Surround Panner
Create Document Template
Template List Dialog
Time Format dialog
Time stretching
Track dispatching
Value Format
File attributes
Audio File editing preferences
Write Audio CD
WMA encoding
Workspace layout
Write Data CD/DVD
Loop Tweaker
Zoom factor

11.1.1 Insert link

This dialog allows you to insert a hyperlink to a web page in your podcast episode description.

In "Title" enter the text that will form the active link.

In "URL" enter the Uniform Resource Locator (i.e. web address) you wish it to open when clicked. Remember to include the full URL, for example: `http://www.personalwebsite.com/index.html`

You can access this dialog from the Podcast Workspace via **Episode** in the list of Items, then click on above the "Description" field. The description of the feed content is a mandatory field.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Podcasts

11.1.2 Align Clips

This dialog allows you to align Clips along the focused track at specific intervals, with or without a space between them. You must select at least two Clips to use this function.

You can access this dialog in the Audio Montage Workspace's Clip window via **Functions** > **Align selected clips of focused track...** or via its icon.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Clips

11.1.3 Recording channels

This dialog allows you to select which input channels you want to record from. If you have an audio card with multiple inputs, you can record up to eight channels simultaneously.

You can access this dialog via **Record** > **Set input**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Recording Dialog

11.1.4 Audio CD Report

This dialog allows you to generate a detailed report about the active audio CD. This report includes a full track listing with ISRC codes (if entered), track times and CD-Text.

The Audio CD report can be output in HTML or Adobe PDF format or printed directly from WaveLab. You can choose the details of what is displayed and include your custom logo. You might send the Audio CD report to your client, an album artwork designer or to the CD replication house when presenting them with a master CD, for example.

You can control the formatting of values in your report via the Value Format window. To open this dialog click on the "Value Format" link.

You can access this dialog in the Audio Montage Workspace's via **CD window** > **Functions** > **Generate Audio CD report...**

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Value Format

Authoring CDs and DVDs

11.1.5 Audio file comparer

This analysis tool allows you to compare two Audio Files for differences between them. It can create a "delta file" that contains these differences. It can also drop markers at positions where differences are found.

This can be useful for:

- Judging the effect of using an equalizer by comparing the file before and after. The delta file will show what was added or removed.
- Check the noise added by a processor.
- Compare two digitally recorded files to check for dropouts.

Note - Make sure that the two documents you want to compare are open.

You can access this dialog via **Analysis** > **File comparator...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.6 Audio File Format Dialog

This dialog allows you to set the type of Audio File to be saved. Some file types will make some options unavailable.

Clicking in Attributes (meta-data) produces the File Attributes dialog for adding non-audio information associated with a file. This option is only available for Wave, MPEG-1 Layer 3 (MP3), Windows WMA and Ogg Vorbis file types, and for each of these, the Edit command leads to a different dialog box.

You can also save and restore settings. See Presets.

You can access this dialog via **File** > **Save as...**> **Output Format** . The dialog is also available from other locations where Audio Files are saved, for example, via the Master Section's render dialog.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Presets

11.1.7 Audio Montage properties

This dialog allows you to set/change the audio properties and number of channels of the active montage file. It appears when you first create an audio montage file and can also be edited later. It allows you to select an output mode for your project, choose how many channels you wish to work with and set the sample rate of the montage.

Changing the values in this dialog will not process the audio in your montage in any way. However if you change the sample rate, the file will play back at the wrong pitch.

You can access this dialog from the Audio Montage Workspace by choosing **Edit** > **Audio Montage properties...**. It is also displayed when you create a new audio montage by clicking on the "Create an empty document" icon or via **File** > **New...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Workspaces

11.1.8 Split Clip at silences

This dialog allows you to split a single Clip into new ones according to any silence detected in its audio.

Effects and envelopes are ignored, only the source material is analyzed. You can define several factors which control the threshold at which silence is detected including the duration of a silence, the signal level, and the gap between silent regions in the Clip. You can choose to use a single- or two-stage analysis process and also to separate the resulting new Clips.

You can access this dialog in the Audio Montage Workspace via **Focused Clip** > **Edit** > **Split at silences...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

11.1.9 Audio properties dialog

This dialog reports the audio properties of the active Audio File. It allows you to change the number of audio samples per second (the sample rate) and the accuracy of samples in the audio stream (the bit-rate).

Changing the values in this dialog does not process the file in any way (in contrast to using Save as). However, the following rules apply:

- If you change the sample rate, the file will play back at the wrong pitch.
- If you change the bit resolution the file will be converted to the new resolution the next time you save it.

Note that there is no Undo for this operation. If you save with a lower bit resolution, the file is permanently converted.

You can access this dialog in the Audio File Workspace by choosing **Edit** > **Audio properties...**. It is also displayed when you create a new Audio File using **Edit** > **New...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.10 Audio Range

This dialog allows you to define accurately a specific audio range. This audio range is then used to create a selection. There are many options you can choose to easily select a specific part of your Audio File.

You can also audition your selection using the various playback buttons provided.

In order to select a region of the Audio File, first select a From and To point using one of the methods available. Additional options allow you to control your selection further.

You can access this dialog in the Audio File and Audio Montage Workspaces by choosing **Edit** > **Select** > **Edit...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.11 Audio Streaming Settings

This dialog allows you to specify how WaveLab's internal input and output channels are connected to your sound card and which device you intend to use for audio playback and recording. You can define the buffer settings for your device as well as setting up connections to external gear such as external effects units. You should select at least two channels for stereo playback and recording.

- **Control Panel** - clicking this will launch your sound card settings application (usually installed with the audio card). In most cases this will provide settings for buffer size, digital formats, additional I/O connections, etc. depending on your sound card and driver.

- **Playback tab** - this allows you to select and name audio ports used for playback.If you are monitoring on a surround system, specify your surround speaker outputs here.

- **Recording tab** - this allows you to select and name your audio ports used for recording and input monitoring. The inputs you define here will then be available in the recording window.

- **External gear tab** - this allows you to select inputs and outputs to external audio processing equipment. For example, you can use this to buss audio out to an outboard effects unit and back into /app.

- **Options tab** - use this tab to specify the number of buffers and control driver functionality (e.g. release the driver when the application is in the background, so that it can be used by other applications). You can also specify pre-roll and post-roll times.

Windows MME/Built-In Audio If you do not have a third-party audio card you can select the Windows MME or Built-in Audio (Mac) options.

You can also use MME with most third party audio cards, with the advantage that you can record and play at different sample rates. However, Windows MME drivers don't allow Audio Monitoring in the record dialog, and other drivers generally offer better sound quality and performance.

You can access this dialog from Workspaces that support the playback of Audio Files by choosing **Options** > **Audio Streaming Settings...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Recording Dialog

11.1.12 Auto Split

This dialog allows you to automatically split an Audio File into separate Audio Files according to certain specified rules.

You can split files:

- ...at marker points
- ...at specific time intervals
- ...at regions containing silences
- ...at beat points using beat detection
- ...according to a description of regions stored in a text file

The Auto Split function can be used in many situations such as:

- cutting up a single Audio File of a recording session into its individual takes (using the silence threshold)
- cutting up a drum loop into its individual drum hit samples (using beat detection)
- outputting individual tracks from an album master file (using marker detection or "Learn regions from text file")
- converting marker types from one type to another (by using the option "Do not split, only create markers at splitting points")
- silencing the regions between audio information in an instrumental take (using "Mute audio outside regions")

The Auto Split tool is made up of several pages of dialogs which take you through the process of splitting your audio. Note that the sequence of pages is not always the same and depends on the choices you make.

You can access this dialog in the Audio File Workspace via **Tools** > **Auto Split...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Split Clip at silences

11.1.13 Batch Conversion

This dialog allows you to convert the file format of a group of Audio Files.

This could be useful for example, if you have a large number of Audio Files that you wish to encode as MP3 files, or a group of stereo files that you need to downsample and convert to mono. You can also take maximum advantage of your computer's architecture if you have a multiple processor cores, by spreading the processing load across them. These settings are selected in the Multitasking tab.

To use the Batch Converter tool:

1. Click on the ✚ plus sign icon to add files, (or just drag the files directly into the pane)
2. Define an output file location and file format
3. Choose any options or performance settings from the Options and Multi-tasking tabs
4. Click Start to begin (you can also choose an option to start the process automatically when dragging the file in).

You can access this dialog by choosing **Utilities** > **Batch conversion of Audio Files...**

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File Format Dialog

11.1.14 Basic Audio CD Settings

This dialog allows you to edit various settings for the Basic Audio CD. You can add optional UPC/EAN codes to your CD, adjust the pause time before the first track and add additional silence at the beginning and end of tracks. You can also choose to play the audio through the Master Section and its effects before burning your audio CD. See Basic Audio CD for more information.

You can access this dialog from the Basic Audio CD window in the Audio File Workspace by choosing the **Options** > **Options...** or by clicking on the spanner icon.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Authoring CDs and DVDs

Basic Audio CD

11.1.15 Batch Renaming

This dialog appears in various locations in WaveLab and allows you to rename a batch of files, Clips or markers.

The Batch Renaming dialog offers a powerful set of facilities to convert, remove, format, replace and insert text into existing names. You can use simple options to match text, or you can build your own regular expressions, as required.

Batch renaming can be useful with large projects for example, so you can apply easily identifiable names to all referenced files, markers and Clips belonging to the project.

Batch renaming can be used for the following operations:

- to rename files
- to rename Clips in Audio Montages
- to rename markers in Audio Files and Audio Montages.

These functions share most of the same features, with some minor differences. The Batch Renaming dialog has three pages. The first page defines which files/clips/markers should be renamed; the second dialog page defines how the renaming should be executed. The third page shows you a preview of the names that will be produced, before you commit to the changes.

You can access this dialog in various places throughout WaveLab:

- from any Workspace: via **Utilities** > **Batch file renaming...**
- from the Marker window: via **Functions** > **Batch marker renaming...**
- from the Clips window in the Audio Montage Workspace: via**Functions** > **Batch clip renaming...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Conversion

Marker Window

About Regular Expressions

Regular Expressions

11.1.16 CD Text Editor

This dialog allows you to add CD-Text data to an audio CD. CD-Text is an extension of the Red Book Compact Disc standard and allows you to store text information on an audio CD. Information such as the disk title, songwriter, composer and disc ID, can be added for example. The text data you enter is displayed by CD players that support the CD-Text format.

In WaveLab you can add information about the disk itself and each individual track. This information is entered in a series of text fields that scroll together horizontally. There is one pane of fields for the disk itself and a pane for each track. You can scroll between these with a slider bar. Buttons allow you to copy between fields to make entering information easier.

Tip: Make sure the "Enable CD-Text Burning" checkbox is ticked before burning an audio CD if you wish to write CD-Text data.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

CD

Authoring CDs and DVDs

Write Audio CD

11.1.17 CD-Text Browser

This dialog allows you to view CD text information about the selected disk when importing an audio CD using the Import Audio CD dialog.

You can access this dialog from the Import Audio CD dialog via **Functions**> **Examine CD-Text...**. You can also access it via its icon in the Import Audio CD window.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Import Audio CD

11.1.18 CD Wizard

This dialog allows you to quickly create an audio CD track listing from the active montage. The Clips in the montage are used to automatically generate CD track and splice markers from the Clip regions and cross fade points. You can adjust any pause time before tracks, adjust the gaps between track start markers and the track audio playing as well as generating any ISRC and UPC/EAN codes. Once you have chosen the options you require click "Apply" to generate the CD track listing in the CD window. You can further edit the track listing that is created and then burn your audio CD using the Write Audio CD dialog.

For more information on creating Audio CDs see Authoring CDs and DVDs.

You can access this dialog from the Audio Montage Workspace's CD window by choosing **Functions** > **CD Wizard...** or by clicking on the icon.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

CD

Authoring CDs and DVDs

Write Audio CD

ISRC

11.1.19 Copy audio information

This dialog allows you to copy information about the name and location of the selected Audio File, including any selection information and cursor position. The information is copied to the clipboard and can be pasted into an external text application.

This can be a useful feature to use when making notes on a particular session or when you need accurate file path/selection information when writing a script.

You can access this dialog in both the Audio File and Audio Montage Workspaces via **Edit** > **Copy Audio information...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

11.1.20 Customize Commands

This dialog allows you to customize your own shortcuts for WaveLab. It shows a list of already assigned shortcuts for WaveLab commands and menu options. Each shortcut is restricted to a specific context, so you can re-use the same shortcut combination in different places. The exception is the Master Section where all shortcuts are global to the application.

To edit shortcuts, double click on the shortcut text or its placeholder, or select a command and click Edit shortcut... See Definition of shortcut(s) for more information.

Please note that some shortcuts cannot be edited, these are unavailable (grayed out). New keyboard shortcuts are displayed in blue.

- **Showing/hiding in menus** - you can decide whether to hide/show certain commands in their respective menu by deselecting/selecting their checkbox in the "Menu" column.

- **Triggering via a keyword** - you can enter a keyword which you can later use to activate a command by typing it into the Keyword field command bar.

- **Triggering from an external MIDI device** - you can assign a command to be triggered by an external MIDI device using the "MIDI Trigger" checkbox. For example, this could be useful for issuing transport commands from your midi keyboard. You can specify a sequence of up to three midi events.

Please note that according to the context not all types of shortcuts are available. For example, triggering via a keyword is not available within the recording context.

- **Display in command bar** - For certain commands (those with icons displayed) you can also choose whether to display them in their corresponding command bar. To do this, select/deselect the option in the "Bar" column. You can define one key shortcut per command, each shortcut can be a sequence of up to four keystrokes. Note that for

the Macintosh only, if a shortcut is made up of two or more keystrokes, the main menu only displays the first keystroke (this is a Mac OS X limitation).

To reset some or all types of shortcuts to their factory defaults use the "Reset" button.

To generate a reference of your customized shortcuts, use the "Summary" button to either view a webpage summary or to print out a list of all your command shortcuts.

You can access this window from within various other dialogs in WaveLab when you see the icon, or by choosing **Options** > **Customize Commands...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Definition of shortcut(s)

Command bars

11.1.21 Data CD/DVD

This dialog allows you to arrange files and folders into a structure that you can later write as a "data" disk.

You can prepare a disk for a variety of media (CD-DA, DVD or Blu-ray Disk). The list of media devices displays the amount of data that will fit on each type of disk. As you add files, an indicator on the left (and a text display at the bottom) display how "full" your new disk is. You can enter a name for your disk and change the disk file structure before writing your data to an ISO image or writeable optical disk.

Note: Although you can prepare a disk in any format, the types of physical disk you can burn will be limited by the optical drive connected to your computer. If you do not have a Blu-ray Disk writer connected, for example, you will not be able to write a Blu-ray Disk you may have prepared. However, you can still save it as an ISO Disk image and transfer it for remote burning.

To prepare a data disk, follow these steps:

Adding files

Add files and folders to the disk by:

- dragging them directly from your computer's file browser.
- dragging a file from any Workspace by either dragging it by its title bar tab or using the document button .
- choosing **File** > **Special** > **Add to Data CD/DVD** from any of the editing Workspaces to add the currently focused file.

Removing files

To remove files from the file structure:

- select the files/folders you wish to remove and click on the red cross icon. To remove a group of files, hold down **[Shift]** while selecting the files/folders to remove. Please note that this *does not* delete the files from the computer, only a reference to them.
- to remove all files/folders from the layout choose **Reset**.

Editing the file structure

To edit the arrangement of the files in your disk:

- drag files/folders to a different location in the file list pane. To move multiple files/folders hold down **[Shift]** while selecting the files/folders you wish to move.
- drag file/folders over a folder icon to move them into that directory.
- to add a new folder click on the folder icon.
- to edit the name of a file/folder, double click on its title.

Burning your disk

Once you are happy with your disk layout you can write it to a disk or disk image:

- select a media type from the list of media devices. This will determine the maximum size and type of medium you will be writing to, whether it be CD, DVD, Blu-ray Disk or an ISO image of any of these. You can change the media type at any stage.
- enter the name that you want your disk to display when it is mounted as a data volume.
- to write your disk, click write CD/DVD icon. This opens the write CD/DVD dialog which enables you to burn your data disk to an ISO image or to an optical disk using a connected CD/DVD/Blu-ray Disk writer.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Write Audio CD

Blu-ray Disk

ISO image

11.1.22 DVD-Audio options

This dialog allows you to choose settings for creating a DVD-Audio disk. You can enter disk identification and volume information, choose the TV format (PAL or NTSC), provide still picture tracks and adjust various options.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **DVD-Audio** > **Options...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

DVD-A

11.1.23 DVD-Audio Creation

This dialog allows you to choose settings when creating a DVD-Audio disk.

You can choose to just test the DVD-A, how to render the audio for each Audio Montage and choose an output folder for the DVD-Audio files that are created. Once the files have been rendered to the hard disk by WaveLab, the Write Data CD/DVD dialog opens automatically, allowing you to burn your DVD-Audio files to an optical DVD.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **DVD-Audio** > **Burn DVD-Audio...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

DVD-A

11.1.24 Document list dialog

This dialog allows you to add a file to a document in a different Workspace.

You can add the file to a new or existing Workspace document. Open documents are displayed in the list box, or you can browse to find an existing document, or create a new one.

For example, to add your Audio File to a Podcast, choose **File** > **Special** > **Add to Podcast...**. The Podcast can be a new or existing Podcast. Open Podcasts are displayed in the list box.

You can access this dialog by choosing **File** > **Special** > **Add to [Workspace]...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Podcasts

11.1.25 Ducking options

This dialog allows you to choose options when using WaveLab's ducking feature. The ducking function allows you to use a Clip's volume envelope to modulate the volume envelope of an adjacent channel's Clip. The most obvious application of this feature is when mixing background music with a voice-over. When the voice-over starts, the level of the music is automatically lowered and when the voice-over stops, the level of the music is restored to the original level. This is done by automatically creating volume envelope curves.

To use this feature you first need to select the Clip you wish to alter the volume envelope of and then from the dialog choose the track containing the Clip you wish to duck. The focused Clip must be located within the time range of the Clip you wish to use as the modulator.

You can access this dialog in the Audio Montage Workspace via **Focused Clip** > **Envelope** > **Duck according to other track...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

11.1.26 Effect morphing

This dialog within the Audio File Workspace allows you to gradually mix two audio ranges that have different effects/processing applied to them.

Effect morphing allows you to smoothly "morph" or blend from one effect to another, or from an unprocessed audio segment to a processed audio segment. Effect Morphing always involves two audio ranges. Normally, these are two versions of the same audio range, of which one is unprocessed and the other processed.

Effect morphing works by using WaveLab's Undo buffer to mix a copy of the processed audio with the unprocessed version. You can also use any other bit of audio from the application's clipboard that has the same length as the one you wish to morph with.

To set up a basic effect morph, proceed as follows:

1. Start by making a selection range over the time you wish the effect morphing to take place.
2. Now process this range using any Master Section effect(s), or use some other off-line processing. You cannot use processing/effects that alter the length of the selection, e.g. time stretching.
3. Open the Effect Morphing dialog by choosing **Edit**>**Effect Morphing...**.
4. Adjust your envelope points over time between 0% and 100%. This will determine the level and direction of the morph. For example, starting at 100% and ending at 0% will fade out the effect you just applied.

5. Choose "Unprocessed selection" for this example. As mentioned previously you can also use any audio of the same duration from the clipboard.

6. Click Apply to apply the Effect morphing.

You can access this dialog in the Audio File Workspace via **Edit** > **Effect morphing...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

11.1.27 Edit playback times

This dialog allows you to set parameters that are used to evaluate the CD when using the command "Play all CD Track starts".

You can set the track pre-roll time, toggle one-second of silence before playback, and set the time each track is played before jumping to the next track.

You can access this dialog from the Audio Montage Workspace via **CD** > **Options** > **Edit playback times...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

11.1.28 Error detection and correction

This specific tool window allows you to search for unwanted clicks and digital artifacts in an Audio File. Several detection and restoration methods are available. You can detect, mark and name, jump to, play back and remove individual audio errors.

The settings for error detection and correction are organized within the following groups of controls:

- **Detection tab** - choose the method used to detect any errors. Each different method has its own settings to determine when it will detect an error. Click Detection 1 and 2 will search for clicks in certain frequency ranges while Digital Click Detection looks for clicks that are caused by clicks typical to digital errors.

- **Correction tab** - specify the method used to correct any errors.

- **Search Area tab** - specify the range of audio you wish to search for errors.

- **Options tab** - this provides a range of preferences for playing back, viewing, and marking any errors found.

- **Browse & Correct panel** - step through your defined search region detecting each error in turn. You can choose to correct any errors or mark them for later correction. There are controls to skip between error markers and make fine adjustments to the selection. You can also automatically detect and correct all marked errors in the search region.

Strategies to detect and correct errors

There are several strategies that you can employ when detecting and correcting errors:

- Define an audio selection where you have identified an error, then click "Correct", or "Mark for subsequent correction".
- Use the function "Detect next error" to let WaveLab automatically find the next error, and then click "Correct", or "Mark for subsequent correction".
- Use the function "Detect all errors" to let WaveLab automatically find all errors in the predefined range. Then browse the marked errors, removing or adjusting each audio range in turn, and click "Correct" to correct any specific errors, or you can use the option "Correct all".

You can access this dialog by choosing **Workspace** > **Specific tool windows** > **Error Correction**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Marker types

11.1.29 Configure external tools

This dialog allows you to configure WaveLab to work with external applications. You can pass command line arguments to the external applications so that they can process the current file/folder you are working on, or even the WaveLab's settings folder. This function can be useful if you wish to edit an Audio File in another application, or wish to compress all your settings files into a backup zip file, for example.

Note that an external tool only works within the Workspace in which it is defined. This is by design, so that each type of Workspace can have its own external "toolkit".

To define an external tool you need to:

- Give the command a title
- Specify the path to the application's executable file
- Choose which arguments you wish to pass to the external application
- Additionally you can specify an initial folder to direct the external application to
- Select how you wish WaveLab to behave before the external tool is launched.

Once you have defined an external tool in this way, you can run it by selecting its title from the Tools menu from within its Workspace.

You can access this dialog by choosing **Tools** > **Configure external tools...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

11.1.30 Folder Preferences dialog

This dialog allows you to set the folders where you prefer to save and open files, as well as options about how to automatically update them, or not, while using WaveLab.

Select a folder category from the list and then either type in the path to the directory or browse to select the folder via a standard file dialog.

Each Workspace has its own set of preferred folders, which are set from the Folder Preferences dialog from that Workspace.

If you open this dialog from the Control Window, you can set the preferred folders for all Workspaces. To use this dialog, first select a folder category (from the left) and then define its path and options (on the right).

A note on temporary files - this folder category specifies where WaveLab should store its temporary files. These are used for operations such as Undo. If you have multiple drives, you should allocate each temporary folder to a unique physical drive. WaveLab then selects the source and destination folders on different physical drives, which speeds up file operations considerably.

You can access this dialog by choosing **Options** > **Folders...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

11.1.31 Frequency range

This dialog allows you to adjust the range of frequencies displayed in the 3D Frequency Analysis window. Use the spin controls to adjust the highest and lowest frequencies you wish to view. This can be useful for isolating a particular frequency range which you are trying to analyze.

You can also choose to display frequencies using logarithmic or linear scaling, alter the color of the background and waveform, plus change the scale of the ruler.

You can access this dialog via **Analysis** > **3D Frequency Analysis** > **Edit frequency range** (click on the frequency button).

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

3D Frequency Analysis

Value editing

11.1.32 FTP site

This dialog allows you to enter information for uploading your Podcast and all related files to an online FTP server.

You must specify all log-in details for your FTP site, the relative path to your podcast XML file and the web site address where the link to your Podcast is found, including the path to the feed. If in doubt about your FTP settings they can normally be supplied by your ISP.

You can access this dialog via **Publish** > **FTP Site...** in the Podcast Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Podcasts

11.1.33 Change Level

This dialog allows you to apply a gain to change the level of an Audio File.

You can also use "Find current peak level" to obtain a report on the peak level of the current audio selection (or the peak level in the whole file, if "Process whole file if there is no selection" is selected in the preferences). This can be useful if you wish to calculate how much you can increase the overall gain of a file without introducing clipping (exceeding 0dB), for example.

To change the level, enter the desired gain and click Apply.

You can access this dialog by choosing **Process** > **Change level...** in the Audio File Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Value editing

11.1.34 File attributes (OGG/WMA)

This dialog allows you to enter/edit the meta-tags for an OGG Vorbis (OGG) or Window Media Audio (WMA) encoded file. Double click on the tags provided to edit an entry.

In the Audio File Workspace you can access this dialog by opening a WMA or Ogg Vorbis file and clicking in the Meta-Data specific tool window. You can also access it by choosing **Save as...** > **Output format** > **Attributes** > **Edit...** when saving a WMA or Ogg Vorbis file, and from most places where you are able to select an output file format, like the Render dialog.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Metadata window

Audio File Format Dialog

OGG encoding

Ogg Vorbis files

11.1.35 Global analysis

This dialog allows you to perform advanced analysis on your audio to identify areas with specified properties. It can be used to find problem areas such as glitches or clipped samples, or just to check general information such as the pitch of a sound.

How it works When you analyze a section of an Audio File, WaveLab scans it and extracts information which it displays in the dialog. It also "pin-points" sections of the file that meet specific characteristics, for example, sections being very loud or almost silent. You can then browse between these points, set markers or zoom in on them.

Types of analysis On most of the tabs, you will find settings determining exactly how the analysis will be performed. Each tab focuses on a particular analysis area:

- **Peaks** - this tab is for finding individual samples with very high decibel values.
- **Loudness** - this tab is for finding sections that are perceived by the human ear as louder or weaker in volume. WaveLab uses an accurate method (RMS - Root Mean Square), to measure a consecutive section of samples and then average their value.
- **Pitch** - this tab is used for finding the exact average pitch of a section of audio. The method works best on monophonic material (single notes, without chords or harmonies) and assumes the analyzed section has a relatively stable pitch. As a general rule try to analyze the sustain portion of a sound, rather than the attack.
- **Extras** - this tab is for informing you about DC offsets and the significant bit resolution of the file. This can be useful, for example, if you wanted to check whether a 16 bit file really uses 16 bits (or if it was actually recorded with only 8 bit resolution and then upscaled to 16 bits).
- **Errors** - this tab is to help you find glitches and sections where the audio has clipped (gone beyond 0 dB).For more control over the analysis, use Error detection and correction. **Checking and browsing the results** Checking the results on the Pitch and Extra tabs is simple, since only one value is returned for the whole section of analyzed audio. For all the other tabs, the analysis methods returns a series of exact points ("hot points") in the file or section of audio. Use the relevant buttons in each tab to select the "hot points" you wish to look at in more detail. Using the slider you can then browse or skip between these points, adding markers or focusing the waveform display at that particular point in the file.

You can access this dialog in the Audio File Workspace via **Analysis** > **Global Analysis...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Error detection and correction

3D Frequency Analysis

11.1.36 Import Audio CD

This dialog allows you to import one or more tracks from an audio CD.

Once you have identified the source CD, you can select the track or tracks to import. You must specify a folder location and file format before the file or files can be saved to disk.

Actions tab Specify the optical disk drive from the drop-down list, and the disk reading speed. The title and track details are automatically completed. If no track names appear, click to refresh CD reading.

From this list you can select the tracks (or define the start and end points of a single track). The "Rename tracks" menu option automatically adds the track order number to the name in a range of different styles. You can select a track name for text editing by double-clicking.

In "Output", accept the default Folder location or choose another. Clicking produces the Audio File Format dialog for changing details of the Audio File's format.

Options tab This accesses a list of tickable options to refine how the tracks are read and output to file.

You can access this dialog via **Utilities** > **Import Audio CD tracks...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File Format Dialog

11.1.37 Insert Audio Files

This dialog allows you to choose how imported Audio Files will be arranged in the active audio montage.

It displays a list of the Audio Files you have selected to import and allows you to add or remove files, rearrange their order, choose whether to create new tracks for each Clip or lay them out on a single track, and where to insert the Clips. To adjust the spacing between Clips that you import, see Audio Montage Preferences.

This dialog appears when adding two or more Audio Files to a montage. You can access it from the Montage Workspace via **Insert context menu** > **Insert Audio Files...**, or when you drag Audio Files directly from your computer's file system.You can also access it from the Montage Workspace via **Workspace** > **Specific tool windows** > **Files** > **File** > **Add Audio Files(s) at cursor...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Podcasts

File

Audio Montage Preferences

11.1.38 Level envelope

This dialog allows you to create a volume envelope which can be applied to a selected range or a whole Audio File. This could be useful if you need to even-out loud and quiet parts, for example.

You can edit the envelope by:

- Double clicking on the line to add a point: double click on a point to remove it.
- Selecting multiple points (using **[Shift]** click) allows you to move the selection as a group.

You can also use the controls to delete or reset points in the envelope. Use the envelope smoothing button to toggle the envelope points between a straight (polygonal) line or a curved path.

Click Apply to apply the envelope to the audio selection or file.

You can access this dialog in the Audio File Workspace via **Process** > **Level envelope...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Adjusting Envelopes

11.1.39 Loop Tone Uniformizer

This dialog allows you to create sounds that will loop, from audio that is seemingly "un-loopable". These are normally sounds that constantly decay in level or continuously change in timbre. The Loop Tone Uniformizer applies processing to the sound that evens out changes in level and timbral characteristics in order for a sound to loop properly. For example, this could be useful for creating looped samples for a softsynth or hardware sampler.

To use the Loop Tone Uniformizer you need to have a loop defined using a pair of loop markers . The original length of the loop is not changed.

Uniformizers

This tab allows you to specify the method/s used to even out the sound you wish to loop. Choose between "Slice mixing" and/or "Chorus smoothing". For a full explanation of how each of these methods operate use the "What's this?" help function.

For slice mixing, you may need to experiment to see how many slices are needed. Generally the more slices you use, the more natural the sound will be (up to a certain point).

Pre Cross fade

The Loop Tone Uniformizer also includes the option to cross fade the end of the loop with the start of the newly processed section so that transition into the newly looped section is smoother during playback. Use the envelope drag points or value sliders to adjust the cross fade.

A note about Post Cross fading

If you have used the Loop Tone Uniformizer within a region of an Audio File, the transition from the end of the loop into the original file in many cases won't be very natural. This can be fixed as follows:

1. Close the Loop Tone Uniformizer and open the Loop Tweaker.
2. Click on the Cross fade tab and deactivate the Cross fade option (the check box).
3. Click on the Post Cross fade tab and make sure Cross fade is activated.
4. Set up the post cross fade parameters and click Apply.

It can also be a useful tip to lock the loop markers after you use this command on a selection within an Audio File. Once the loop is processed in place, moving the loop markers will upset playback of the loop.

You can access this dialog in the Audio File Workspace via **Process** > **Loop Tone Uniformizer...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Markers

Adjusting Envelopes

11.1.40 Loudness distribution

This dialog allows you to measure the most frequent loudness values found in an Audio File (this is not the same as average loudness). The graphic peaks found by the analysis represent these values.

The tool gives an answer to the question: "How often does a given loudness (vertical scale, in dB) appear in the whole file?". The percentage figure is relative to other peaks. Some simple examples:

- 1. If, for instance, you have a sine of 0 dB / 2 second followed by a sine of -6 dB / 2 seconds. This means the global audio material has as much 0 dB material as - 6dB material: you will see 2 peaks (0 / -6 dB), each at 100%.
- 2. If, for instance, uou have a sine of 0 dB / 1 second followed by a sine of -6 dB / 3 seconds. This means there are 3 times more -6 dB material than 0 dB material. You will see a peak of about 33% at 0 dB, and another peak of 100% at -6 dB.

This analysis is useful for learning about how the loudness of your music is distributed.

You can access this dialog in the Wave File Workspace via **Analysis** > **Loudness distribution...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

11.1.41 Loudness normalizer

This dialog allows you to adjust the loudness of a file. Because it relates to loudness as opposed to maximum peak level, this tool works quite differently from the Level Normalizer. A typical application is to specify a loudness, for example -12dB, and the Loudness Normalizer processes the audio to match this level of loudness.

Like Gain Change, increasing the loudness to a certain value can introduce clipping. To avoid this occurring, a peak limiter (the Peak Master plug-in) can be used as part of the process. The Loudness Normalizer raises the loudness and limits peaks in the signal at the same time (if needed), to achieve the target loudness.

In the case of stereo files, both channels are processed independently. This process happens over several stages; analysis precedes the final rendering. The Loudness Normalizer dialog gives access to useful statistics about the file and the Loudness distribution tool, and you can choose to remove any DC offset in the file.

You can access this dialog from the Audio File Workspace via **Process** > **Loudness normalizer...**.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Loudness Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Loudness

DC Offset

Loudness distribution

11.1.42 Global Preferences

This dialog allows you to view and change options that are common throughout the application's Workspaces.

You can adjust the following settings:

- **General** - this tab allows you to change the location of WaveLab's settings files and the user interface language. You need to restart the application for changes to take effect.
- **Display** - this tab allows you change many aspects of the user interface that apply across the whole application. These options provide useful information and usability functions but can be deactivated to streamline the interface. "Use Window Switcher" is an import panel that controls the appearance of the Window Switcher panel and whether you want particular Workspaces to appear.

- **Formats** - this tab allows you to adjust settings for some of the audio formats and units that WaveLab uses.
- **CD Burning** - this tab allows you to set a number of parameters for CD burning.
- **Options** - this tab allows you to control application-wide start-up options and transport settings. You can also reset default answers. This can be useful for example, if you deactivated any "Don't show this message again" warning messages that you now wish to restore.
- **Global shortcuts** - this tab allows you edit key sequences for shortcuts available across all Workspaces.

You can access this dialog via **WaveLab** > **Preferences...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Workspaces

11.1.43 Convert marker type

This dialog allows you to convert marker types in the current file. You can choose to convert all markers in the file or only those within a current waveform selection.

You can convert up to three different types of markers to other types. There is also the ability to set conditions that a marker's name must match in order for it to be converted. These conditions can be expressed as text matches or regular expressions. A regular expression is a precise way of matching strings of text. When you choose to use regular expressions for your conditionals an additional pop-up menu allows you to access a list of common regular expressions. You can also save presets of common conversions you make using the standard presets control.

You can access this dialog from the Marker window via **Functions** > **Convert marker types...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Marker Window

Markers

About Regular Expressions

11.1.44 Load Master Section Preset

This dialog allows you to choose whether plug-ins, dithering, and master level settings are overwritten or used (the default action) when loading an existing Master Section preset.

This can be useful if you wish to keep some/all of the current plug-ins and level settings you have in the Master Section when you load another preset. For example, you may wish to maintain the current master level and dithering settings but replace the effects chain that you have set up with one you commonly use from another preset. You can use the checkboxes provided to adjust which settings from the new Master Section preset are used or ignored.

You can access this dialog from the Master Section by opening any Master Section preset from the preset menu when you have the "Open option box when selecting preset" option selected. You can also access it from the Status Bar.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Master Section

11.1.45 Save Master Section Preset

This dialog allows you to save the current settings of the Master Section as a preset.

You can choose whether plug-ins, dithering, and master levels are saved or ignored. This can be useful for example, for saving chains of effects plug-ins (and their settings) that you commonly use.

You can access this dialog from the Master Section by choosing **Master Section preset menu** > **Save as...**. You can also access it from the Status Bar.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Master Section

Audio Plug-ins

11.1.46 Audio Montage Clone

This dialog allows you to clone your active Audio Montage and its associated Audio Files.

You can choose a number of options that determine how WaveLab will consolidate the montage and its related audio Clips when making the clone.

You can access this dialog from the Audio Montage Workspace via **File** > **Save special** > **Clone completely...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.47 Write Audio CD

This dialog allows you to define some options when creating a new audio montage from an Audio File that is open in the Audio File Workspace. You can choose whether to use a selection or the whole Audio File to create a new Montage file, and how to interpret any markers that are present.

You can access this dialog in the Audio File Workspace via **Edit** > **Create Montage from Audio File...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

11.1.48 Meta-normalizer

This dialog allows you to adjust the loudness of each Clip so that they get the same loudness.

You can use it to equalize the volume of all the Clips within a montage, either by peak level or loudness. This can be an easy way to ensure that all Clips playback at a similar volume. You can choose to only equalize the volume of selected Clips, and effects can be excluded if you wish.

This tool is to achieve the **same** level of loudness in all Clips (the highest loudness found, if possible), while ensuring that no Clip will cause sample clipping.

For each Clip, a specific gain is computed by the algorithm after **all** Clips have been analyzed (and prior to changing the Clip gain to achieve the common loudness). If it is not possible to match the highest found loudness, the level of the Clip with the highest loudness is reduced, so that other Clips can achieve the same loudness.

To avoid clipping the Master Section you can also normalize the mixdown output of the montage (before it goes into the Master Section), and/or the Master Section output.

You can access this dialog from the Audio Montage Workspace by choosing **Tools** > **Meta-normalizer...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Master Section

11.1.49 Audio Montage Preferences

This dialog allows you to define settings for working in the Audio Montage Workspace.

Using the two tabs you can define settings for both the active audio montage and for all audio montages. You can adjust default settings for how WaveLab works with audio Clips and effects plug-ins in an audio montage, as well as to define backup and display options.

You can access this dialog from the Audio Montage Workspace via **Options** > **Audio Montage preferences...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Workspaces

11.1.50 MP2 encoding

This dialog allows you to specify the encoding options when you save an MP2 Audio File. You can select the bit rate and stereo encoding format, as well as define other options.

You can access this dialog via **Save as...** > **Output Format (select type MPEG-1 Layer 2)** > **Encoding** > **Edit...**, and from most places where you can select an output file format, such as the Render dialog.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

MP2 files

11.1.51 MP3 attributes

This dialog allows you to enter/edit the meta-tags (ID3 tags) for an MPEG 3 encoded file.

You can access this dialog via **Edit** > **File attributes...** when you have an MP3 file open in the Audio File Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

MP3 files

11.1.52 MP3 encoding

This dialog allows you to specify the encoding options when you save an MP3 Audio File.

You can select which encoder to use and adjust the bit rate and quality of the file. You can also choose the format and method the encoder uses to output the file, as well as select other advanced options.

You can access this dialog by choosing **Save as...** > **Output Format (select type MPEG-1 Layer 3)** > **Encoding** > **Edit...**, and from most places where you can select an output file format, such as the Render dialog.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

MP3 files

11.1.53 Normalize Level

This dialog allows you to change the peak level of your Audio File.

Enter the desired peak level (in dB) that you wish the audio selection to have. You can also use "Find current peak level" to obtain a report on the peak level of the current audio selection (or the peak level in the whole file, if "Process whole file if there is no selection" is selected in the preferences). You can choose to apply the same gain to both stereo channels (Stereo Link) or mix down to a mono file (Mix to Mono) with the assurance that no clipping will occur when both channels are mixed together.

To normalize the selected audio, enter the desired peak level and click Apply.

You can access this dialog by choosing **Process** > **Level Normalizer...** in the Audio File Workspace.

This function is also available as a batch processing plug-in. You can access it from **Batch Processing Workspace** > **Plug-ins window** > **Multipass Plug-ins** > **Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

Value editing

11.1.54 OGG encoding

This dialog allows you to specify the encoding options when you save an Ogg Vorbis Audio File.

You can choose between using an average or variable bit rate to adjust the size and quality of the file.

You can access this dialog via **Save as...** > **Output Format (select type Ogg Vorbis)** > **Encoding** > **Edit...**, and from most places where you can select an output file format, such as the Render dialog.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Ogg Vorbis files

11.1.55 Open files list

This dialog allows you to view a list of all currently open files and documents in WaveLab.

Documents from all the different Workspaces are displayed and can be open, closed, renamed, saved and in the case of Audio Files, auditioned. The list can also be filtered to only display the files you are interested in.

When accessed by a shortcut key, this dialog is a quick way to jump from one file to another if you have many of them open and spread across various Workspaces. This dialog is designed for productivity.

Keeping this dialog open means that it will automatically be updated if any document is closed/opened, saved or edited.

You can navigate the list using the arrow keys, or by using a-z on your keyboard.

You can access this dialog via **Global** > **List of all open files...**

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.56 Pan Normalizer

This dialog allows you to ensure that both channels of a stereo file have the same level or loudness. Since loudnesss is more important than peak volume in the perception of sound, this is a powerful tool to get the best possible stereo balance.

This process uses two passes to first analyze the audio and then render any level changes required.

You must have a stereo selection in a stereo file to apply this process.

You can access this dialog from the Audio File Workspace via **Process** > **Pan Normalizer...**.

This function is also available as a batch processing plug-in. You can access it from the Batch Processor Workspace via **Plug-ins window** > **Multipass Plug-ins** > **Pan Normalizer**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.57 Mix dialog

This dialog allows you to specify a change in gain while pasting audio.

You can adjust the gain of the copy of audio contained in the clipboard, and/or the audio at the destination so you can control how the two components mix together.

You can access this dialog from the Audio File Workspace via **Edit** > **Paste special** > **Mix...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.58 Pitch bend

This dialog allows you to change the pitch of a sound over time. Note that changing the pitch using Pitch bend will affect its duration unless length preservation is activated.

Using an envelope you can 'draw' the curve you wish the pitch to follow. The pitch offset is displayed along the vertical ruler of the envelope and the range the envelope effects can be adjusted using the spin control. Positive pitch values produce sounds with a higher pitch and of shorter length, negative values produce sounds with a lower pitch and of longer length.

If length preservation is activated, you can choose the algorithm that is used to perform the pitch bend operation. Depending on the type of audio material you are processing, choose the appropriate mode. For more information on the different modes, use the "What's This?" tool. You can also adjust the quality used when processing the pitch bend. The quality setting and the selected mode will affect the processing time for this effect.

This function could be used for creating the classic 'tape stop' effect, or for blending the tempo/pitch of one track into another, for example.

You can access this dialog from the Audio File Workspace via **Process** > **Pitch bend...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

11.1.59 Pitch correction

This dialog allows you to detect and change the pitch of a sound without affecting its length.

- **Amount of Shift** - use these controls to find out the current pitch of the audio and if desired, calculate the required shift to match a specific pitch.

- **Method** - use these controls to adjust the method and quality you wish to use when processing the pitch change. You can also choose how the length of the sound is affected by the operation. By default this process will not change the length of the sound.

This tool could be useful for fixing an off-key vocal note in a live recording, or tuning the pitch of a kick drum sample to fit a particular song, for example.

You can access this dialog from the Audio File Workspace via **Process** > **Pitch Correction...**.

Pitch correction is also available as a batch processing plug-in in the Batch Processor Workspace. In addition, Pitch correction for a Clip is available from the Focused Clip window (via **Edit** > **Pitch shift...**).

Note that for Batch processing and Clip processes, the envelope option is not available.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

11.1.60 Pitch quantize

This dialog allows you to correct the pitch of an Audio File.

The musical notes in the audio selection are quantized to the nearest semitone. Pitch quantize works best on recordings that have a single fundamental frequency, such as voice or single instruments. It can be useful for 're-tuning' the pitch of an imperfect recorded vocal performance, for example.

The dialog allows you to choose the reference frequency (normally 440 Hz) and the time taken for the pitch correction to reach the nearest semitone. An explanation of when and how to adjust the Slur time is given using the 'What's this?' tooltips.

You can also choose to preserve the formants of the original material. Choosing this option can sometimes help to produce a more realistic result when correcting the pitch of vocals or acoustic instrument sounds.

You can access this dialog from the Audio File Workspace via **Process** > **Pitch quantizing...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

Formant

11.1.61 Playback speed

This dialog allows you to change the playback speed for open audio documents with or without affecting the pitch of the audio. Playback speed can be set for both the Audio File Workspace and the Audio Montage Workspace (all Clips in a montage are affected).

Each open wave or montage file can be assigned its own independent playback speed. For example, you can have several wave windows open and change the playback speed for one of them, without affecting the playback speed of the other windows when you switch to them. This requires that you deactivate the option "Transport settings are global to all windows" in the Global preferences > Options tab.

This function can be useful in a number of ways:

- A common application is to speed up the listening time of an interview or conference speech.
- You can increase playback speed to quickly locate specific positions in the audio
- You can decrease the playback speed in order to practice a tricky recording take.

By optionally defining presets you can easily switch between your favorite playback speeds. Once defined, you can quickly jump between them using the Speed context menu in the transport's command bar.

If you are experiencing problems using different playback speeds, there are a number of options to help you balance processing quality with CPU usage.

You can access this dialog in the Audio File and Audio Montage Workspaces via **Transport** > **Playback speed** > **Edit speed...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Transport controls

Command bars

Context menus

Global Preferences

11.1.62 Podcast options

This dialog allows you to define a number of options for working with Podcasts. You can choose how WaveLab handles pictures added to your Podcast, select a different timezone to publish for, and/or define an external HTML editor.

You can access this dialog from the Podcast Section via **Options** > **Options...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.63 Text format

This dialog allows you to export text information from WaveLab in a choice of text formats, or you send the text directly to your printer.

You can access this dialog from various locations in WaveLab, wherever text is produced for output.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.64 Repeat Clip

This dialog allows you to make a number of copies of a Clip and lay them out at various intervals on the current track of your audio montage. You can specify the number of clones to produce, and control their placement, alignment and spacing.

You can access this dialog in the Audio Montage Workspace by right-clicking on the track's waveform and choosing **Repeat clip...** from the Clip context menu.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

Clip

11.1.65 Recently used files

This dialog allows you to view all the files you have recently used in WaveLab. You can use it to find a specific file and open it. Information on the full path of each file, its name and the date it was last accessed, is displayed.

To find a file quickly, begin typing the name of the file. You can also choose to remove individual files from the list or those that can no longer be located by the application. The number of recently opened files remembered by WaveLab is set in the Global Preferences.

You can access this dialog from all Workspaces via **File** > **Edit list of recently opened files...**. You can also open it from various menus that list recently used files such as the **File** > **Favorites files...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Global Preferences

11.1.66 Recording Dialog

This dialog enables you to set up and record an Audio File from the input device specified in the Audio Streaming Settings dialog.

To set up a recording:

- In the "File to create" section, select Temporary File, or select Named File and browse to the folder where the recorded audio will be saved.

- Click the down arrow to select a preset audio format, or click the audio format text to display the Audio File Format dialog, and set up the format for your recorded audio. Here, you can create new Audio File Format presets, if required, for later use.
- Choose from Level or Spectrum display. If you choose Level, click the Settings button to bring up the Level/Pan Meter Settings dialog.
- Click Record. While the recording is in progress, you can create named markers by clicking on the marker icons in the Actions section.
- Click Stop when the recording is complete. An unsuitable recording is easily discarded, or you can Close the dialog to edit the recorded file in the Audio File Workspace.

In WaveLab you also have further options available:

- **Method** - use this tab to define options for starting/stopping/pausing the recording automatically. You can select an input device and choose to start a recording at a specific time or stop it after a specific duration. In WaveLab you can also choose to drop markers automatically and split a file into while recording.
- **Options** - use this tab to determine various options affecting how the recording dialog window behaves while recording.
- **Values** - use this tab to define numerical values for the various recording options available.

You can access this dialog from the Transport controls button or by choosing **Transport** > **Record**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Streaming Settings

Audio File Format Dialog

Record create marker

11.1.67 Record create marker

This dialog allows you to name markers to drop while you are recording an Audio File. You can enter a name for each marker and an optional time offset.

Creating named markers can be useful for junping to these points quickly when later reviewing or editing the recording.

You can access this dialog while recording from the Recording dialog, dropping a marker by clicking the button.

You must have the "Confirm name of markers to drop" option selected for this dialog to appear. This option is located in the Options tab of the Record dialog.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Recording Dialog

11.1.68 Rename File dialog

This dialog allows you to rename a file, and/or optionally change the folder where it is stored.

The file suffix may be selected from the drop-down list. All open documents that have a reference to the file are updated. Associated peak and marker files are also updated.

This function is particularly useful when renaming Audio Files. If you rename an Audio File outside WaveLab (using your computer's file browser), the next time you open an Audio Montage with a reference to that file, WaveLab will not find it.

To avoid this, use this dialog to rename files and all references to the file from the open Audio Montage are automatically updated to use the newly named file.

You can access this dialog by choosing **File** > **Rename...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.69 Render ASIO input to file

This dialog allows you to record an Audio File directly to disk from an ASIO input. The audio is collected from the audio input, not from the Audio File or Montage. The ASIO input audio is rendered through the Master Section and any of its plug-ins and saved as a file, as when rendering normally. This is in fact, another way to record. When you record normally, no plug-ins are used, but more options are possible. On a Mac, this function should work without ASIO.

Note: It is impossible for WaveLab to know when the input should stop, so it is possible to record for hours unless you manually stop this process using the Transport controls.

You can access this dialog from the Master Section when you select the Asio Input plugin and then press the Master Section Render button.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Rendering

Recording Dialog

Master Section

11.1.70 Render Montage window

This dialog allows you to process the montage session you are working on, mixing down multiple tracks into a mono or stereo file and applying any active plug-in effects you have active. This process is sometimes also referred to as mixing or bouncing down.

You can select options for which regions of the session are rendered, whether to create a named file or a temporary, untitled file, and whether to mute the effects chain when finished.

There are various options to render all or parts of the Montage, into one or more files - use "What's this..."

You can access this dialog in the Audio Montage Workspace via the Render button from the Master Section window.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Rendering

Audio Plug-ins

Audio File Format Dialog

Master Section

11.1.71 Render Wave window

This dialog allows you to process the Audio File or selection you are working on, applying any effects you have active.

It applies all active plug-ins to the selected audio region or the whole file, and you can optionally change the format of the rendered file. This process is sometimes also referred to as "mixing" or "bouncing down".

You can select options for which regions of the session are rendered, whether to create a new file or process in place, and whether to mute the effects chain when finished. You can also choose to copy marker locations to the new file.

You can access this dialog in the Audio File Workspace via the Render button from the Master Section window.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Rendering

Audio Plug-ins

Audio File Format Dialog

Master Section

11.1.72 Loudness envelope options

This dialog allows you to adjust how the Loudness envelope waveform is displayed.

You can adjust the granularity of the analysis, or if the loudness analysis is based on small or longer segment. You can also see the loudness of specified frequency regions (by using the low/band/high pass filters to define which frequencies are displayed). This can be useful for isolating a particular "problem" range of frequencies which you want to focus on, for example.

This analysis is computationally intensive, so it may occasionally be slow to redraw.

You can access this dialog from both wave windows in the Audio File Workspace via the **Loudness tab** > **Edit settings...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.73 Sample attributes

This dialog allows you to define settings for an audio sample before loading it into a hardware or software sampler.

The settings do not process the sample in any way, they just give the file properties that the receiving sampler can use. This includes information about the pitch of the sample (which can be detected automatically by WaveLab), the key range that the sample should span, and the velocity range to occupy. For WAV and AIFF files, this information is stored in the file's header.

By default, there are no sample attributes in an Audio File. Therefore you must explicitly create this information by clicking on the "Create" button first. This dialog is useful if your sampler can take advantage of these additional tags. If supported by your sampler, it can save you time by allowing you to both edit and define a samples properties from within WaveLab.

You can access this dialog via **Workspace** > **Specific tool windows** > **Sample Attributes...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.74 Modified files dialog

This dialog allows you to save any currently open files that have been modified.

You can choose which files in the active Workspace you wish to save and which to ignore. If you have a large number of currently active files, it can be useful to choose "Select All" and then deselect any items you do not wish to save. Choosing "Cancel" safely returns you to the application without saving the modified files or quitting.

You can access this dialog throughout WaveLab via **File** > **Save all**.

This dialog is also shown when closing a Workspace with more than one modified file, or when closing the application.

In WaveLab you can also access this dialog from the Control Window via **File** > **Save all documents in all windows**

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.75 Save Audio File dialog

This dialog allows you to choose the name, file format and the location where you wish to save an Audio File.

The provided default location depends on what path you have specified for the 'Audio File Save folder' in Folder preferences. This can be overtyped, or you can click to browse folders, browse files, or explore the Operating System browser dialog.

You can also change the type of file format used to encode the file using the Audio File Format dialog. In addition there is an option to remember the file format you use so it appears the next time you save a file.

You can access this dialog from the Audio File Workspace and Montage Workspace via **File** > **Save as...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Folder Preferences dialog

Audio File Format Dialog

11.1.76 Select where and how Audio Files will be saved

This dialog allows you to choose where Audio Files are saved when saving each CD track as an Audio File from the Basic Audio CD window. You can choose the destination folder location and the format of the Audio Files which will be saved.

You can access this dialog in the Audio File Workspace via **Basic Audio CD window** > **File** > **Save each CD track as....**

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Basic Audio CD

Audio File Format Dialog

11.1.77 Save Batch Processor

This dialog allows you to save a copy of the batch processor file that you currently have open. You can specify a new file name and location. Optionally you can choose to save the list of files in the batch process, along with status of each file.

The default location depends on what path you have specified for the 'Batch Processor - Save folder' in Folder preferences. This can be overtyped, or you can click to browse folders, browse files or explore the Operating System browser dialog.

You can access this dialog in the Batch Processor Workspace via **File** > **Save as...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Folder Preferences dialog

11.1.78 Save File As

This dialog allows you to save a copy of a file that you currently have open. You can specify a new file name and location.

The default location depends on the path you have specified for the active Workspace's 'Save folder' in Folder preferences. This can be overtyped, or you can click to browse folders, browse files or explore the Operating System browse dialog.

You can access this dialog in various Workspaces in WaveLab via **File** > **Save as...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Folder Preferences dialog

11.1.79 Save Preset As

This dialog allows you to save a copy of a preset you have altered with a different name. You can also define and create a root folder to hold for all presets of its kind.

You can access this dialog in various places throughout WaveLab where you choose presets - that is, wherever you see this icon and a menu containing **Save as...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Presets

11.1.80 Save picture of active window

This dialog allows you to save a picture of the active WaveLab window.

You can save the picture with or without the enclosing window frame and the resulting picture can be copied to the clipboard or saved to a file. If the second option is selected, you can specify where the picture is saved and the format: BMP, JPG/JPEG and PNG formats are offered.

You can access this dialog from most Workspaces in WaveLab via **View** > **Save picture of active window...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.1.81 Scripting window (Audio Montage Workspace)

This window allows you to write and execute scripts in the Audio Montage Workspace.

The inbuilt text editor helps you when writing scripts by highlighting the different parts of the script with colors, making it more readable. A script can also be written in another text editor, and loaded via the File menu. To run a script, from within the script window choose **Functions** > **Execute script**.

For a general description of scripting, see Scripting. For an overview of the scripting language, see ECMAScript Reference.

You can access this dialog from the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Script**.

Note: To view trace or log messages when executing scripts that contain the logWindow() function, ensure that the Log window is visible and that it's warning filter buttons - ! ⚠ ⓘ are selected.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Scripting

ECMAScript Reference

Log window

11.1.82 Scripting window (Audio File Workspace)

This window allows you to write and execute scripts in the Audio File Workspace.

The inbuilt text editor helps you when writing scripts by highlighting the different parts of the script with colors, making it more readable. A script can also be written in another text editor, and loaded via the File menu. To run a script, from within the script window choose **Functions** > **Execute script**.

For a general description of scripting, see Scripting. For an overview of the scripting language, see ECMAScript Reference.

You can access this dialog from the Audio File Workspace via **Workspace** > **Specific tool windows** > **Script**.

Note: To view trace or log messages when executing scripts that contain the logWindow() function, ensure that the Log window is visible and that it's warning filter buttons - ! ⚠ ⓘ are selected.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Scripting

ECMAScript Reference

Log window

11.1.83 Definition of shortcut(s)

This dialog allows you to define your own customized shortcut(s) for a particular function (selected from Customize Commands). These custom shortcuts can save you time and speed up your workflow in WaveLab.

You can define a shortcut using a:

- **Key sequence** - any existing keyboard shortcut is displayed in the first keystroke field and can be overwritten. There are some standard shortcuts that cannot be redefined; in this case the related edit fields are disabled. Each shortcut is restricted to a specific Workspace, so you can use the same shortcut in different Workspaces.

Press the key combination you wish to use and it will be displayed. If a keyboard shortcut is already used by another command, a warning is given. Use the Clear button to erase any shortcut entries already present.

There are additional spaces to allow you to define sequences containing up to four key strokes. Please note these keys must be pressed/released one after the other, and the action is performed when the last key has been pressed.

- **Keyword** - enter a keyword which you can later use to activate the command by typing it into the Keyword field command bar.

- **MIDI event sequence** - use an external MIDI device (such as a MIDI keyboard or dedicated control surface) to define a sequence of up to three MIDI events. First you need to select the MIDI interface for incoming messages. Press the key or button on your MIDI device and the corresponding MIDI event number will be displayed. You can define a sequence by continuing to input further MIDI messages. You could use a Program or Bank change message as a "modifier", for example.

Defining MIDI shortcuts can be useful if you want to use your external MIDI device to control the transport in WaveLab, for example. You can also use your external MIDI device to control

almost any function within WaveLab. Select a MIDI device via **Preferences...** > **Options tab**, activate "Use MIDI shortcuts" and specify a MIDI input port so that MIDI events are sent rather than keyboard events.

You can access this dialog from the Customize commands window by selecting a command and choosing **Edit shortcut...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Customize Commands

Command bars

11.1.84 Signal generator

This dialog allows you to generate complex synthesized sounds in mono or stereo.

You can layer different waveform generators together and if outputting a stereo file, adjust different settings for both the left and right channels. There are a multitude of settings to adjust the character (Source tab), frequency (Frequency tab) and amplitude (Level tab) of the generated signals.

To generate a sound follow these basic steps:

- choose whether the file is to be mono or stereo, and define the number of samples, and bit depth using the Audio properties dialog.
- choose how many layers of signal generators you wish to use (up to 64).
- for each layer, use the source tab to choose the type of signal generator you wish to use, switch to the frequency tab to define a frequency and its envelope, and use the level tab to define a level envelope. If you have chosen a stereo file you can also choose to adjust left and right channels independently.
- once all settings are made, choose Generate and WaveLab calculates an Audio File according to your specifications. The file is generated and opens up in a new window.

Generating signals can be useful for a variety of testing and measurement purposes. For example, you could use the Signal Generator to create a sine wave sweep from say 20Hz to 20,000kHz to test your monitoring environment for resonant frequencies.

You can access this dialog in the Audio File Workspace via **Tools** > **Signal Generator...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio properties dialog

DTMF Generator

11.1.85 Silence Generator

This dialog allows you to generate "silence", either by replacing an audio selection, or by specifying a duration.

You can also specify a cross fade at each end of the silent region, for a smoother transition. Radio buttons determine where the silent period is placed in the audio.

In WaveLab you can also specify a file to be inserted instead of "true" silence. This can be useful for example, if you wish to keep the ambience, or background noise of a particular environment in a recording.

You can access this dialog in the Audio File Workspace via **Edit** > **Silence (advanced)...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Silence

11.1.86 Smart bypass

This dialog allows you to correct for any level change introduced by plug-in effects in the Master Section.

Some effects, when applied to the signal chain, can cause an increase or decrease in level between their input and output signals and this impairs objective comparisons between processed and unprocessed sounds. By compensating this level shift, an objective A/B comparison is possible. This is the core purpose of this feature, which is a must-have tool for Mastering.

Although some plug-ins have an output level or make-up gain control (such as most compressors for example), it can be quite difficult to determine the increase/decrease of level when multiple plug-ins are used in an effects chain. This dialog can determine this difference in level and adjust the output gain accordingly. You can choose to hear the original audio, the processed audio or the processed audio with the correction in level applied while playing back audio. You can also select the method used to calculate the amount of level correction used.

You can access this dialog from the Master Section via **Smart bypass**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Master Section

11.1.87 Special File Format

This dialog allows you to choose options as to how WaveLab will interpret any unrecognized file formats or files containing raw data that you wish to open. You can define the sample

format, byte order, sample rate, file extension and whether or not the file contains mono or stereo channels. You can also choose to ignore the file's header (start) or end of file data. Once you have specified a format for a particular type of file, WaveLab remembers the new format for the next time. This function can be useful for example, if you are working with a rarely used Audio File format like those used by some game platforms or mobile phone ring tones.

You can access this dialog in the Audio File Workspace via **File** > **Open as...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File Format Dialog

11.1.88 Spectrogram options

This dialog allows you to adjust how the Spectrogram is displayed.

You can adjust the visual style of the spectrogram and its resolution. You can also specify a cut-off point above which frequencies are displayed and to plot them using a linear or logarithmic scale. This can be useful for isolating a certain range of frequencies: for example, in sound restoration you may wish to focus on a high frequency band which is more accurately displayed using a linear scale.

You can access this dialog from both wave windows in the Audio File Workspace via the **Spectrum tab** > **Edit settings...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

The Spectrum display

Spectrum editor

11.1.89 Surround Panner

This dialog allows you to adjust the pan of your audio between surround sound channels. In the graphic display, the positions of the clip left/right audio channels are shown as small squares. The proportional signal levels from the individual speakers are indicated by colored lines from the speakers to the center of the display.

The graphic display shows the surround imaging of either the left/top (blue) clip audio channel, or the right/bottom (red) clip audio channel – the color of the speaker lines shows which channel is selected for viewing and editing.

If you are viewing the left/top channel, you will see a blue square indicating the position of the audio. The other, gray square (which is mirrored horizontally) represents the other channel. **Click this to view and edit this channel instead**, it will be shown in red and red speaker lines indicate the speaker levels.

Each track can have its own Surround Pan window and you can have several of these open at the same time if you need to.

To pan the sound, click the blue square and drag it to the desired position. This pans the audio of the left/top channel – the other channel is automatically mirrored horizontally.

If you right-click in the graphical display, a pop-up menu appears allowing you to choose from a number of positioning "presets". This can be a quick way to pan a signal fully left-front, for example.

This dialog is closely related to surround panning envelopes. Indeed, you can automate surround panning for individual Clips using envelopes. These envelopes can be selected from the Focused Clip editor panel. You can choose to view either the Left-Right, Front-Rear or the LFE curve. This is slightly different from using regular volume and pan envelopes.

The important point to understand: internally, there is a single surround pan envelope where **each envelope point contains a complete surround state (left-right position, front-rear position and LFE amount)**.

If you add an envelope point to the Left-Right surround pan envelope for example, you will find that a point is automatically added to the Front-Rear and LFE envelopes, at the same position in the Clip. This is because there is actually only a single surround pan envelope. When you add an envelope point you are really adding it to this "full" surround pan envelope, and it is shown in all three views. In other words, the Left-Right, Front-Rear and LFE curves are three views of the same multidimensional envelope.

Note that if you move an envelope point, all surround pan envelopes are affected!

You can use the Surround Pan window to "program" each envelope point – this makes it easy to set up automated surround panning for a Clip. While the Surround Pan window is open, click on a surround pan envelope point. Its position is reflected in the Surround Pan window. You can then simply edit the point in the Surround Pan window rather than on the envelope.

You can access this dialog in the Audio Montage Workspace by right clicking on the small surround panner window in the track view column.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

11.1.90 Create Document Template

This dialog allows you to create a template from an active Audio Montage, Podcast or Batch processor file. Starting a new document from a template can be a great time-saver.

Each template you create retains all the settings of the current document plus any tracks you have created, although any data in the file, such as Clip references, are not saved. The next time you create a new document you can choose a template file from a list which is then used as a basis for the new document.

You can use this dialog to create a set of templates that you commonly use. For example, if you are frequently mastering CDs in the Audio Montage Workspace you could create a montage template specifically for that purpose.

You can access this dialog from the Audio Montage, Podcast or Batch processor Workspace via **File** > **Save special** > **Save as template...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Template List Dialog

11.1.91 Template List Dialog

This dialog allows you to choose a previously saved template from an active Audio Montage, Podcast or Batch processor file. Starting a new document from a template can be a great time-saver.

Templates are saved using the Create Document Template dialog. You can choose to use an existing template as the basis for your new file or choose "None" to define a new blank one.

You can only access this dialog if you have created at least one template previously by choosing **File** > **New...** from the Audio Montage Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Create Document Template

11.1.92 Time Format dialog

This dialog allows you to specify the way in which time is displayed on the waveform time rulers and in several time-based fields related to the file.

You can set options using the following tabs:

- **Timecode** - this provides a list of frames per second for various SMPTE timecodes, and for CD resolution. Selecting "Other" allows you to enter a custom value. You can also choose which frames/units are displayed. For example, if you are working with PAL video you might choose 25 frames per second - or "25(SMPTE EBU)".

- **Clock** - this allows you to display time units or to present the clock units in a compact format.

- **Meter** - this offers time signature, tempo (bpm) and ticks per quarter note (to allow the time units to be compatible with your MIDI sequencer).

- **Offset** - this provides a number of ways to specify a time offset for the time ruler. This can be useful if you wish to move the "time zero" point of the file. For example, you might want to have the time display zero on the first note of a recorded performance whilst still maintaining the pre-roll part of the Audio File.

You can access this dialog in the Audio File Workspace via **View** > **Main view display** > **Time Ruler** > **Time format...**. Alternatively, you can access it by right clicking anywhere on the time ruler in either waveform window to produce the Time Ruler context menu, and choosing **Time format...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Timecode

Audio File editing

11.1.93 Time stretching

This dialog allows you to change the duration of an audio selection (usually without changing its pitch).

You can view exact time information on the original audio selection you wish to stretch and choose the amount you wish to "stretch" it in seconds, beats per minute or as a percentage. You can also choose which method WaveLab uses to stretch the audio and the quality/speed of the processing.

If you select "Use modulation envelope" you can vary the time stretch over time using an envelope editor that becomes available. When choosing a method, it is useful to experiment with the different algorithms available to find which one suits your source material the best. Use "What's this?" to find out more information on each method and determine which one will produce the best time stretch without introducing unwanted artifacts.

This function is useful if you want to speed up or slow down a recording to match other material, such as matching the tempo of two drum beats, or fitting an audio track to a section of video, for example.

Time stretching is also available as a batch processing plug-in in the Batch Processor Workspace. In addition, Time stretching for a Clip is available from the Focused Clip window.

Note that for batch processing and Clip processing, the modulation envelope option is not available. You can access this dialog from the Audio File Workspace by choosing **Process** > **Time stretching...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Adjusting Envelopes

11.1.94 Track dispatching

This dialog allows you to control the output of different tracks in your audio montage.

The purpose of this dialog is to provide an answer to the question "Into which output channel will the mono channel, or the left and right channels, be mixed?"

For instance, in the case of surround output, there can be six destinations for a track channel.

Track allocation is dependent on the mode and how many channels you have selected when defining your Audio Montage settings. This in turn defines how many outputs are available from this dialog. For playback, your soundcard must also have the available number of corresponding channels. These are set up in the Audio Streaming Settings.

You can choose single or multiple channel outputs for each stereo and mono track in your audio montage. For stereo tracks, you can define channels individually for both the left and right channels of a stereo track. You can also apply a low pass filter to the Low Frequency Effects ("sub-woofer") channel in order to eliminate unwanted higher frequencies from being sent to sub-woofer speakers.

You can access this dialog in the Audio Montage Workspace by clicking on the **Lf:Rf** icon at the left hand edge of each track.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

11.1.95 Value Format

This dialog allows you to control the formatting of the different types of data values in your CD report. These include the date, time, audio time, track indexing and serial number formats. For example, you may wish to display dates in the format "date/month/year" instead of with a full time and date stamp (the default). You can choose from the many formatting options available for each data type in its corresponding drop down menu.

You can access this dialog in the Audio Montage Workspace's CD window via **Functions** > **Generate Audio CD report...** > **Value Format**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

CD

Audio CD Report

11.1.96 File attributes

This dialog allows you to define the file attributes (or meta-data) to be embedded in your exported Audio File.

You can choose between entering standard WAV file meta-data tags or the extended file header information available in Broadcast Wave Format (BWF).

Fill in the relevant information using the text fields provided, although it is not mandatory to complete all fields.

In some cases there are buttons to assist you in entering data:

- - Uses the current date and time to fill in the Origination Date/Time field.
- - Uses WaveLab as the name of the originator software.

You can access this dialog in the Audio File Workspace via **Edit** > **File Attributes...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

WAV files

11.1.97 Audio File editing preferences

This dialog allows you to define settings for editing in the Audio File Workspace.

You can choose defaults for editing and playback, adjust the visual appearance of the waveform displays, and determine how WaveLab works with audio and peak files.

You can access this dialog in the Audio File Workspace via **Options** > **Audio File editing preferences...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File editing

11.1.98 Write Audio CD

This dialog allows you to write an audio CD to either a DDP image or a physical CD-ROM disk using your computer's CD-R or DVD-R drive, or burn an audio CD from an existing DDP Image. The dialog has controls for refreshing the list of attached devices, opening the CD/DVD drive and erasing a CD-RW/DVD-RW disk or the contents of the DDP Image directory. You can also choose to bypass audio from passing through the Master Section before it is written. If you choose to burn a DDP image, the dialog allows you to select the directory to output the DDP files, as well as the option to write a table of contents and customer information.Choosing a CD-R or DVD-R drive displays choices for setting the speed, number of copies and other options.

Important note for Mac OSX users

When using the CD-R/DVD-R drive on Apple OSX computers for importing or burning it is important to note the following. If you insert a optical CD or DVD disk in the drive while the drive is not selected in WaveLab for import or burning, WaveLab will not be able to "see" the drive. The disk drive is "taken over" by other applications under Mac OSX and is not seen by WaveLab when this happens. To avoid this occurring, make sure there is no disk in the drive when opening any CD window within WaveLab.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

DDP files

Authoring CDs and DVDs

11.1.99 WMA encoding

This dialog allows you to specify the encoding options when you save a WMA Audio File.

You can select which encoder to use and adjust the output sample rate and bit resolution. Depending on the encoder chosen, you can also define various quality options and whether the resulting file is stereo or mono.

You can access this dialog by choosing **Save as...** > **Output Format (select type Windows Media Audio)** > **Encoding** > **Edit...**, and from most places where you can select an output file format, like the Render dialog.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

WMA files

11.1.100 Workspace layout

This dialog allows you to define options when saving the layout of a Workspace.

You can choose to save the placement of the Workspace and all its windows and/or the layout of tabbed windows. It allows you to choose whether to retain or discard these elements of your current layout arrangement when saving a new layout.

You can access this dialog from all Workspaces via **Workspace** > **Layout** > **Save as...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Workspaces

11.1.101 Write Data CD/DVD

This dialog allows you to write a data CD/DVD to either a physical DVD/CD-ROM disk using your computer's CD-R or DVD-R drive, or burn an ISO disk image. The dialog has controls for refreshing the list of attached devices, opening the CD/DVD drive and erasing a CD-RW/DVD-RW disk.

If you choose to burn an ISO disk image the dialog allows you to select a filename for the resultant ISO file. Choosing a CD-R or DVD-R drive displays choices for setting the speed or to try a test burn.

Important note for Mac OSX users

When using CD-R/DVD-R drive on Apple OSX computers for importing or burning it is important to note the following. If you insert a optical CD or DVD disk in the drive while the drive is not selected in WaveLab for import or burning, WaveLab will not be able to "see" the drive. The disk drive is "taken over" by other applications under Mac OSX and is not seen by WaveLab when this happens. To avoid this occurring, make sure there is no disk in the drive when opening any CD window within WaveLab.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Authoring CDs and DVDs

11.1.102 Loop Tweaker

This dialog allows you to refine a region of audio to loop seamlessly.

The Loop Tweaker can just be used to "tweak" an existing loop selection so that it loops perfectly or you can also use it to create a loop from material which does not naturally repeat. To use the Loop Tweaker you must first have a loop defined using a pair of loop markers .

The Loop Tweaker consists of the following tabs:

Loop points adjustment

This is a visualization of the beginning and end of the waveform between the loop markers. You use this dialog to manually refine a loop selection by dragging on the waveform left/right or using the automatic search buttons to find the nearest "good" loop point. The aim is to align the waveforms so they meet in the middle at a zero-crossing point with the waveforms matching as closely as possible, being "in phase". When you adjust your loop start and end points within the dialog the start and end loop markers in the main waveform window will adjust accordingly. Note that this movement may or may not be visible depending on the amount you move the markers and the zoom factor you have selected.

It is helpful to have the transport looped during playback so you can hear the difference when you adjust the loop markers within the dialog. Use the Display options to adjust how the Loop Tweaker waveform is visualized. You can also use the temporary memory buttons to capture a number of different loop "candidates" and audition them one by one. Note that if you are not using a cross fade or post cross fade you do not need to click Apply when just tweaking loop points. You can also leave this dialog window open and manually adjust the position of the markers in the main waveform windows if you need to make large adjustments.

Cross fade

This tab allows you to apply a cross fade at the end of a loop by mixing a copy of the beginning of the loop with it. This can be useful to smooth the transition between the end of a loop and its beginning - especially when using material that doesn't naturally loop. Use the envelope

drag points or value sliders to adjust the cross fade envelope. Click Apply to process the cross fade.

Post cross fade

This tab allows you to cross fade the loop back into the audio after the end of the loop. This is done by mixing a copy of the loop back into the audio. Use the envelope drag points or value sliders to adjust the cross fade envelope. Click Apply to process the post cross fade.

Using the "Copy" button you can insert multiple copies of the loop into the active Audio File, creating a smooth audio sequence, since copies will loop without an audible transition.

You can access this dialog by choosing **Process** > **Loop Tweaker...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Markers

Adjusting Envelopes

Zero crossing

11.1.103 Zoom factor

This dialog allows you to set the level of magnification of the waveform in the main wave view with precision.

Choose the zoom factor unit and enter an amount using the spin control box, or change the amount by dragging up or down with the mouse or use the mouse wheel without clicking - see Value editing.

You can access this dialog by left-clicking on the zoom factor box in the Status Bar. A right-click on the zoom factor box gives you various zoom options.

See Zooming and scrolling for other methods of zooming - for example, you can press [arrow up] or [arrow down] to zoom in or out.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Zooming and scrolling

Value editing

Status Bar

11.2 Shared tool windows

Shared tool windows are windows that can only exist in one place at a time and are shared between Workspaces. They perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Shared tool windows:

Phasescope
VU Meter
Oscilloscope
Bit Meter
Spectroscope
Spectrometer
Wavescope
Timecode
Background task monitor
Log window
Master Section

Related topics

About Tool Windows
Metering
Docking windows

11.2.1 Phasescope

The Phasescope indicates the phase and amplitude relationship between two stereo channels. It is only really relevant when monitoring stereo audio material.

Reading the Phasescope

It can be interpreted as follows:

- A vertical line indicates a perfect mono signal (the left and right channels are the same).
- A horizontal line indicates that the left channel is the same as the right, but with an inverse phase.
- A random but roughly elliptical shape indicates a well balanced stereo signal. If the shape "leans" to the left, there is more energy in the left channel and vice versa (the extreme case of this is if one side is muted, in which case the Phasescope will show a straight line, angled 45 degrees to the other side).
- A perfect circle indicates a sine wave on one channel, and the same sine wave shifted by 90 degrees on the other.

- Generally, the more you can see a "thread" shape, the more bass there is in the signal; the more "spray-like" the display, the more high frequencies are in the signal.

Phase Correlation meter

At the bottom of the display, you will find a Phase Correlation meter, which shows the same information but in a different way:

- The green line shows the current phase correlation, while the two red lines show the recent peak minimum and maximum values, respectively (these are the default colors, which you can change).
- With a mono signal, the meter would show +1, indicating that both channels are perfectly in phase.
- Similarly, -1 indicates that the two channels are the same, but one is inverted.
- Generally, for a "good" mix, the meter should show a value between 0 and +1.

Unlike the main Phasescope, the Phase Correlation meter is also available in "Analyze audio selection" mode, showing an average value for the selected range.

Changing settings

You can define display colors, peak hold time, and the resolution or number of samples to display from the **Functions** > **Settings...** menu.

The Phasescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

11.2.2 VU Meter

The vu-meter is used to display the peak and average loudness/decibel level of your Audio File. It also displays the pan, or the balance between the left and right channels in a stereo file.

The upper part of the level/pan meter window shows the peak level and average loudness, in the following way:

- The Peak Level meters display the peak levels of each channel, graphically and numerically. By default, the meter segments and numerical peak values are displayed in green for low levels, yellow for levels between -6dB and -2dB, and red for levels above -2dB. You can change both the colors and the range boundaries if you wish via the Level/Pan Meter Settings dialog. This can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

- The VU (Volume Unit) meters measure the average loudness (RMS) of each channel. These meters have a built-in inertia, evening out loudness variations over a user-defined time span. If you are monitoring playback or audio input, you will also note two vertical lines following each VU meter bar, seemingly "trying to reach" the current RMS value. These lines indicate the average of the most recent minimum RMS values (left line) and the average of the most recent maximum RMS values (right line). To the left, the difference between the minimum and maximum average values is displayed (the level value in brackets) - this gives you an overview of the dynamic range of the audio material.

- If you are monitoring real-time audio (playback or input), the maximum peak and loudness values are displayed numerically to the right of the meter bars. Numbers in brackets to the right of the Maximum Peak values indicate the number of successive clips (0dB signal peaks). Recording levels should be set so that they only rarely clip. If the master level is set too high, the sound quality and frequency response will be compromised at high recording levels, with unwanted clipping effects. If the level is set too low, noise levels may be high relative to the main sound being recorded.

Pan meters

The lower part of the window shows the pan (the difference in level between the left and right channel, only applicable when monitoring stereo audio):

- The upper pan meters show the peak level difference between the channels, graphically and numerically. Note that the pan meters are "two-sided"; the level bars can go to the left or right, indicating which channel is the loudest. The two sides are shown in different colors (which can be changed via the Settings dialog as explained previously).

- The lower pan meters show the average difference in loudness between the channels, in a similar way. This gives you a visual indication of whether a stereo recording is properly centered, for example.

- If you are monitoring real-time audio (playback or input), the maximum balance difference value (peak and loudness) for each channel is displayed numerically to the left and right of the meter bars. The VU meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces.It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

11.2.3 Oscilloscope

The Oscilloscope offers a highly magnified view of the waveform around the playback cursor position.

If you are analyzing a stereo file, the Oscilloscope normally shows the separate levels of the two channels. However, if you activate the option "Show Sum and Subtraction" on the Functions menu (or click the +/- icon), the upper half of the Oscilloscope shows the sum of the two channels and the lower half shows the subtraction.

Making settings

By opening the Settings dialog you can adjust the display colors, and choose to activate Auto-zoom or not. With Auto-zoom activated, the display is optimized so that the highest level reaches the top of the display at all times. The Settings dialog can be accessed through the **Functions** > **Settings...** menu, or by using the icon.

The Oscilloscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

11.2.4 Bit Meter

The Bit Meter shows the resolution or the number of bits used in the digital audio being monitored. Normally the maximum number of bits in an Audio File is the same as the resolution of the Audio File (e.g. a 16 bit Audio File shows that up to 16 bits are used), but sometimes this is not the case.

As soon as you perform any kind of real-time processing on an Audio File, the audio data is treated at a much higher resolution (32 bit floating point), to allow for pristine audio quality. Examples of such processing include level adjustments, effects, the mixing of two or more files, etc. In fact, the only time when a 16 bit file is played back at 16 bit resolution is if you play it without any fades or effects, and with the Master Faders set to 0.00 (no level adjustment). You can try this out for yourself by playing back a 16 bit Audio File and viewing the Bit Meter - as soon as you adjust the Master Faders, note that 24 bits are used and the "inter" indicator is lit (more on this below).

How to read the Bit Meter

- The innermost meters (closest to the bit scale) show the number of bits in use. You can adjust this display in the Settings dialog.
- The outer meters are "history" meters, showing how many bits were recently in use. You can adjust the hold time in the Settings dialog.
- The "over" segment indicates clipping, similar to a clip indicator.
- If the "below" segment is lit, there are more than 24 bits. The bit meter will show the 24 higher bits, and the "below" segment indicates the existence of extra, lower bits. Note that audio is always processed with more than 24 bits internally.
- If the "inter" segment is lit, this indicates that the audio data cannot really be expressed on a regular 24 bit scale (there are floating point values "in between" bits, thus the label "inter"). This is typically the case if you apply effects; thus the "inter" segment allows you to differentiate processed from non-processed 24 bit PCM files.

When to use the Bit Meter

- To check whether dithering is necessary or not. As a rule, if you're playing back or mixing down to 16 bits, and the Bit Meter shows that more than 16 bits are used, you should apply dithering.

- To see the "actual" resolution of an Audio File. For example, even though a file is in 24 bit format, only 16 bits may be used. Or, a 32 bit file may only use 24 bits (in this case, the "below" segment would not be lit). For this purpose, the Bit Meter is best used in "Analyze selection" mode.

- To see whether a "zeroed" plug-in still affects your signal, or whether a plug-in uses 16 bit internal processing, or more generally, to detect any signal modifier stage in the audio chain between reading and playing.

To adjust the Bit Meter settings, select "Settings..." from the Options pop-up menu or by using the icon.

The Bit meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Dithering

11.2.5 Spectroscope

The Spectroscope shows a continuous graphical representation of the frequency spectrum, analyzed into 60 separate frequency bands, represented as vertical bars. Peak levels are shown as a short horizontal lines above the corresponding band, indicating recent peak/-maximum values. The Spectroscope offers a quick spectrum overview. For a more detailed analysis of the audio spectrum, use the Spectrometer. The Spectroscope can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace.It can also be docked in the Control Window.

Related topics

Metering

Spectrometer

11.2.6 Spectrometer

The Spectrometer uses FFT (Fast Fourier Transform) techniques to display a continuous frequency graph, providing a very precise and detailed real-time frequency analysis.

- The current frequency spectrum is shown as a linear graph.
- Spectrum "peaks" are shown as a short horizontal lines, indicating recent peak/maximum values.

Snapshots

By using the "Add snapshot" and "Erase last snapshot" buttons, you can take and erase snapshots of the current spectrum. These will be superimposed over the current peak spectrum graph, in a color you can customize, until you click the icon again to take a new snapshot. One use for this is to check the effects of adding EQ, for example. Up to five snapshots can be superimposed on the display; the sixth snapshot replaces the earliest one in the display, and so on. (Don't confuse the order of snapshots with the numbered buttons - these are Spectrometer presets, see below.)

Zooming

You can adjust the frequency scale and range in the Settings dialog as described below, but it's also possible to temporarily zoom in on a frequency area of interest. This is done by clicking and dragging a rectangle in the Spectroscope display. When you release the mouse button, the display is zoomed in so that the enclosed frequency range fills the window. To return to full-scale display, select "Zoom-out fully" from the Functions menu, or double-click anywhere on the display.

Making settings

You can adjust the behavior and display of the meters as desired, and assign up to five sets of Spectrometer settings to the Preset buttons, for instant access. Open the Settings dialog by selecting "Settings" from the Functions menu or clicking the "tool" icon. Note that you can apply your settings without closing the dialog, by clicking the Apply button.

If you want to store your settings for later use (or assign them to a Preset button), select "Save as..." from the pop-up menu in the lower part of the dialog, and specify a name for the preset in the file dialog that appears. Now, you can choose to make the settings instantly available for selection in the FFT Meter window, by using the "Assign to preset button" submenu on the pop-up menu.

- When you are finished, click OK to close the dialog.

Selecting Spectrometer presets

If you have assigned your settings to the Preset buttons in the Settings dialog, you can quickly switch between different level scales and display modes, by clicking one of the Preset icons [1]-[5], or selecting the desired preset from the Options pop-up menu.

Exporting FFT data as ASCII text

When using the Spectrometer in off-line mode ("Monitor Edit cursor Position" or "Analyze audio election" mode) you can export the displayed FFT data as a text file, by selecting

"Export FFT data as ASCII" from the Options pop-up menu. The resulting text file can then be imported into applications that allow graph plotting from text files (Microsoft Excel, for example).

The Spectrometer can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

Spectroscope

11.2.7 Wavescope

The Wavescope meter displays a real-time waveform drawing of the audio signal being monitored.

It can be useful when recording or rendering a file if "Monitor File rendering" mode is active.

Making settings

You can adjust settings for the display via the Wavescope Settings dialog. This can be accessed via **Functions** > **Settings...** menu, or by using the icon. Here you can set various color options for the background, grid and waveform display, as well as setting the waveform rendering speed and vertical zoom. If "Clear waveform when reaching right of pane" is checked the waveform display is cleared each time the cursor reaches the right end of the display. If unchecked, the previous waveform is overwritten.

Tip: Wavescope is a useful meter for visualizing audio during recording. The Wavescope meter can be found in the **Meters** menu of the Audio Files and Audio Montage Workspaces. It can be used either as a floating window, or docked in the Workspace or the Control Window.

Related topics

Metering

11.2.8 Timecode

This shared tool window displays the current time of the playback head (or the cursor position if not playing back), in the timecode format currently selected in the Time Format dialog.

The current time can be in a range of standard SMPTE formats, CD or DVD specific formats, or a custom one - see Time Format dialog.

The most common use for this window is for working with video or film. You can use it to display the playback head position of the current Audio File in a SMPTE format to frame level resolution (i.e. hours:minutes:seconds:frames). You can also use it to display CD frames (minutes:seconds:frames) when you are authoring CD or DVDs.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Timecode**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Time Format dialog

CD Frame

SMPTE timecode

Authoring CDs and DVDs

11.2.9 Background task monitor

This specific tool window allows you to view all background rendering processes that are in progress.

You can adjust the priority with which they are processed, pause or cancel them. It is useful if you have a number of lengthy processes underway and want to free up some processing power to focus on editing. You can either lower the priority of a task so it does not use as much of the computer processor's capacity, or pause the task temporarily.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Background task monitor**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window by choosing **Utilities** > **Background task monitor**.

You can choose to automatically open the Background task monitor when a rendering process begins. To enable this option, select the checkbox via **Options** > **Global Preferences** > **Options** > **Make the Background Task Monitor visible when a task starts** . Since this window is a Shared_tool_window, the location where it opens may vary.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Control Window

Rendering

11.2.10 Log window

This shared tool window allows you to view log messages that have been issued by WaveLab.

When using WaveLab's scripting language for example, the *logWindow( )* function outputs messages to this window. There are a number of toggle buttons and these allow you to filter the types of messages displayed.

Note: When using the logWindow( ) method you need to have the "Show informal notes" button selected for the messages to be visible. If no log messages are displayed, make sure all the toggle buttons are selected - ! ⚠ .

You can access this window via **Workspace** > **Shared tool windows** > **Log**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window via **Utilities** > **Log**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Control Window

Specific tool windows

Scripting

11.3 Specific tool windows

Specific tool windows are windows that are specific to the current Workspace. They perform useful tasks in the active Workspace.

They can be docked and un-docked, and saved in your custom layouts.

Audio File Workspace tool windows:

File Browser

Spectrum editor

Basic Audio CD

Scripting window (Audio File Workspace)

Metadata window

Marker Window

Sample attributes

Error detection and correction

Master Project

Audio Montage Workspace tool windows:

Focused Clip window

Marker Window

Snapshots

File

File Browser

Zoom

CD

Clips

Notes

Effects

DVD-Audio

Scripting window (Audio Montage Workspace)

History

Navigator

Groups

Master Project

Podcast Workspace tool windows:

Master Project

File Browser

Batch Processor Workspace tool windows:

Master Project

File Browser

Batch Processing Plug-ins

Control Window tool windows:

Master Project

Related topics

About Tool Windows

11.3.1 File Browser

This dialog allows you to browse files directly from within WaveLab rather than using your operating system's file browser.

It provides you with all the standard browsing functions (such as List and Icon views)as well as additional controls to audition Audio Files and any marker defined regions.You can use it to open or insert an entire file or a specific region of a file by dragging and releasing the file in the location you wish to insert the file. In the Audio Montage Workspace you can also choose to only view certain types of WaveLab specific files. The File Browser can be very useful in speeding up the process of auditioning a long list of sound files. It can also be used to drag to windows such as Data CD/DVD, including dragging folders.

The playback functions

Play file Once you have selected a file in the File Browser list, you can audition it by clicking Play in the File Browser's toolbar. Click again to stop.

Auto-play You can also have files play back automatically as soon as they are selected. To do this, activate Auto-Play mode.

Tip: To audition a long list of Audio Files, enable Auto-play and use the UP and DOWN cursor keys to step through each Audio File in quick succession.

Opening and inserting a file or region

Once you have chosen a file you wish to open, double-click it to open the file in the current Workspace. You can also open it by dragging it onto an empty tabbed group, or onto the tab bar of a tabbed group.

You can also drag the file over a wave view to **insert** it at a given point. When you drag a file over a wave view, a beam is displayed so as to clearly display the insertion point.

When browsing Audio Files, the file browser also has a region panel. When you select an Audio File, its related regions are listed in this panel. You can then drag the region from the list, to insert only that part of the audio. Double-clicking on a region has the effect of opening the file in the Workspace and selecting that region. You can access this dialog via **Workspace** > **Specific tool windows** > **File Browser**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

11.3.2 Spectrum editor

This specific tool window allows you to select and perform operations on an Audio File via its audio spectrum.

It uses high quality linear-phase filters to process a spectrum selection both in the time domain and in the frequency domain. This can be useful for advanced audio restoration tasks like removing noise from a specific source in a recording, for example.

Making and editing selections

Spectrum editing can only be performed when a waveform is first displayed in Spectrum display mode and when a spectrum selection is defined. To activate the Spectrum editor, click on the Spectrum edit tool in the command bar (or via **Workspace** > **Specific tool windows** > **Spectrum Editor**). This displays the Spectrogram if this is not already the case (see The Spectrum display).

To select a region, point and drag to define a rectangle. The region selection defines a time and a specific frequency range. This allows you to edit and process audio both in the time

domain and in a specific frequency domain, as opposed to standard wave editing which always operates in the full frequency domain.

Notes on spectrum selections:

- When defining a region in a stereo file, a "mirrored" region is automatically created in the other channel. To define a region in only one channel, press **[Shift]** while drawing the region.
- Once you have a region selected, if you place your mouse cursor inside the region, a pop-up appears showing the currently set frequency range (Hz) and the time range (seconds/milliseconds) for the selection.
- A selected region can be moved in any direction by clicking and dragging the region with the cursor.
- If you press **[Shift]** whilst dragging a region, it only moves horizontally thus retaining your selected frequency range. If you press **[Ctrl]/[Command]** + **[Shift]** the region only moves vertically thus retaining the selected time range.
- You can resize a region by placing the mouse cursor at the edges of the region (so that a double-arrow appears) and then clicking and dragging.
- To remove a region that you have drawn, click anywhere else on the waveform and it disappears.
- If you have defined a region as a Source or Target, it can be selected at any time to create a new selection using the same dimensions.

Spectrum editor functions

The Spectrum Editor has the following parts:

- **Selection** - use this part to refine and define your selection(s). You can expand selections, shift selections around and define a source and target region for performing copy operations.
- **Operations** - use this part to perform copying, filtering and processing operations. For copying operations, you require both a source and a target region to have been defined using the Selection tab. You can choose which method to copy audio between the Source and Target regions. You can also choose to apply processing to a single spectrum region using a variety of different modes. See Spectrum Processing Modes for information on each mode type.
- **Master Section** - use this part to process and route selected spectral audio regions through the Master Section and its effects plug-ins. This allows for frequency-selective processing. A selected region's frequency spectrum can be routed to the Master Section where you can choose to process it separately from the non-selected frequency spectrum. The signal is split so that one part (region spectrum or non-selected spectrum) is sent to the plug-ins, while the other part can be mixed with this processed signal, after the Master Section output.

Using the Spectrum editor

You can use the Spectrum editor in two main operational modes:

Region copying and filtering (Operations tab)

This is mainly intended for audio restoration applied over brief time ranges. Selections can be copied, pasted, and filtered. This type of processing is mainly used to reduce, remove or replace unwanted sound artifacts in the audio material, and can be carried out with great precision. This could be useful for example, to replace part of a live recording that contains an unwanted noise (such as a mobile phone ring tone), with a copy of a similar region of the spectrum that only contains a "clean" signal. In general, the spectral Copy/Paste combination gives the best results, given that the source and destination regions are properly choosen.

Master Section processing (Master Section tab)

This allows you to process a specific frequency range via the Master Section. The selected or non-selected regions of the spectrum can be processed differently. You can also employ a number of filters (Bandpass/Low-pass/High-pass) to further refine the particular range of frequencies to be affected by any Master Section effects.

A selected spectrum region can either be:

- Processed separately by the Master Section plug-ins. The non-selected spectrum can either be bypassed or sent to the Master Section.
- Bypassed. This removes the selected spectrum region from the Audio File. The non-selected spectrum can either be routed to the Master Section input or the Master Section output.
- Sent to Master Section output. The non-selected spectrum can either be bypassed or sent to the Master Section input. If the latter is selected, it will be mixed with the selected spectrum region at the Master Section output.

You can access this window by choosing **Workspace** > **Specific tool windows** > **Spectrum Editor**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

The Spectrum display

Spectrum Processing Modes

Spectrogram options

Spectrometer

11.3.2.1 Spectrum Processing Modes

When applying processing to a spectrum region in the Spectrum Editor you can use a variety of different modes:

Damp

This is used to attenuate the level of a region. The Gain parameter specifies the level of the attenuation (it is also possible to set positive gain values, i.e. to boost the frequencies in the region). Three filter types are available to perform the damping:

- Bandpass filter: All frequencies in the region are equally attenuated.
- Low-pass filter: Higher frequencies in the region will be more attenuated.
- High-pass filter: Lower frequencies in the region will be more attenuated.

For the Low- and High-pass filters, a low Filter steepness setting is normally used (around 6 to 18dB).

Blur peaks

This filter operation analyzes the region to find the frequencies with the highest level. The level of these frequencies will be attenuated (or boosted) according to the set Gain value. If the gain is negative, these frequencies will be "blurred" and disappear in the mix as a whole. The purpose of the filter is to mask the loudest frequencies (e.g. it could be used to eliminate a sudden burst of pitched but unwanted sound in the audio material, like acoustic feedback).

This mode works best with the Infinite steepness setting, which is automatically selected when Blur peaks is used. You can, however, freely set any Filter steepness setting. The filter type options are not available when this mode is selected.

Dispersion

This is a special filter that "smears" the dynamics and pitch of the region, without changing the actual frequency content. This works best on lower frequencies, to mask the identity of a signal without changing the frequency spectrum.

This mode works best with the Infinite steepness setting, which is automatically selected when Dispersion is used. You can, however, freely set any Filter steepness setting. The filter type options are not available when this mode is selected. Dispersion can also be used to create special effects as it warps the sound in an unique way.

Fade Out

This can be used with any of the three filter types (Bandpass/Low-pass/High-pass). It gradually filters the frequencies in the region along the time axis (from nothing at the left edge of the region, to maximum allowed by the gain parameter at the right edge), useful for gradually removing frequencies from a region.

Fade In

This works like Fade out but inversed. It can also be used in conjunction with any of the three filter types (Bandpass/Low-pass/High-pass).

Fade out then in

This is a mix of the two previous options; the filtering effect gradually fades out until the middle of the region, then fades back in. It can also be used in conjunction with any of the three filter types (Bandpass/Low-pass/High-pass).

Fade in then out

The inverse of the above. It can also be used in conjunction with any of the three filter types (Bandpass/Low-pass/High-pass).

Related topics

The Spectrum display

Spectrogram options

Spectrum editor

11.3.3 Basic Audio CD

This specific tool window allows you to burn basic "Red Book compatible" audio CDs. You create your audio CD by adding Audio Files to create a list of tracks. Each track contains a reference to the external Audio File. This means that you can save your Basic Audio CD layout as its own session and continue editing individual tracks, for example. Once you are happy with your CD layout you can choose to check the cd for conformity to the Red Book standard, burn the CD or export it to the Audio Montage Workspace to edit further. You can also choose to consolidate the Audio Files in the CD into a single Audio File with track markers.

Adding tracks to your audio CD

To create tracks in your audio CD:

- **Drag Audio Files directly** - drag Audio Files directly from your computer's file browser into the window. You can also drag Audio Files from the Audio File Workspace by their tab or document icon. This creates CD track markers at the file edges, quantized on CD frame boundaries.

- **Drag an audio selection** - dragging a selection (rather than a whole file) into the basic audio CD creates a CD track with markers at the edges, but quantized on CD frame boundaries, not necessarily exactly on the exact position of the selection edges.

- **Browse for Audio Files** - select **File** > **Add track(s)...** from the Basic Audio CD menu. You can then select Audio Files to insert in your CD.

Once you have added your tracks, information about each track is displayed. Each track also has its marker information nested within it. To access this information click on the left most arrow icon to expand the track view.

Using the track list

Once you have added a number of tracks to your audio CD you can use the track list to:

- **Audition playback** - you can audition each track, with or without pre-roll, by clicking on the leftmost icons. Click on in the "audition" column to play the track from that track's start marker with a pre-roll. Click on the right hand button to playback the track exactly from the start marker position. You will notice a progress display appears at the bottom of the window to show you the playback position/time and track number. You can also start playback with or without a pre-roll from the tracks start or end markers using the same controls. If the markers are not visible click on the track expand arrow to reveal the track's markers.

- **Edit a track's start pause time** - for all tracks except the first one you can adjust the pause duration before the track begins playing. If the pause checkbox is not visible click on the track expand arrow to reveal the track's markers and pause time setting. Double click in the cell to edit the value.

- **Add ISRC code** - add an ISRC code for a track by Select-clicking on the field under the ISRC column.

- **Edit a track name** - by Select-clicking on the name in the list. This also renames the track's start marker labels.

- **Add a comment** - add a comment to each track by Select-clicking to edit its value.

- **Editing track times** - you cannot edit track start and end times directly in the track listing. Instead, open the tracks accompanying Audio File via **Edit** > **Edit Audio** and adjust the track start and track end marker locations in the main waveform display. The start and end times will update accordingly.

Completing your Audio CD

In addition to editing via the track window you can also set overall CD settings via the Basic Audio CD Settings dialog. You can use this dialog to add UPC/EAN codes and adjust pauses and silences in your CD. The Master Section is never used to process the signal to burn, unlike in the montage. This is why playback through the Master Section is an option that is off by default.

When you are happy with your CD layout or you wish to further refine it further you can:

- **Check CD conformity** - before burning your CD you can first check its Red Book conformity by choosing **Edit** > **Check CD conformity**.

- **Write audio CD** - choose to burn the CD using the Write Audio CD dialog.

- **Convert to Audio Montage** - convert the CD layout into an audio montage for further editing by choosing **Edit** > **Convert to Audio Montage**

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Authoring CDs and DVDs

Basic Audio CD Settings

Write Audio CD

Select-clicking

11.3.4 Scripting window (Audio File Workspace)

This window allows you to write and execute scripts in the Audio File Workspace.

The inbuilt text editor helps you when writing scripts by highlighting the different parts of the script with colors, making it more readable. A script can also be written in another text editor, and loaded via the File menu. To run a script, from within the script window choose **Functions** > **Execute script**.

For a general description of scripting, see Scripting. For an overview of the scripting language, see ECMAScript Reference.

You can access this dialog from the Audio File Workspace via **Workspace** > **Specific tool windows** > **Script**.

Note: To view trace or log messages when executing scripts that contain the logWindow() function, ensure that the Log window is visible and that it's warning filter buttons - ! ⚠ ⓘ are selected.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Scripting

ECMAScript Reference

Log window

11.3.5 Metadata window

This specific tool window displays any Meta-data tags belonging to the currently active file.

This usually consists of a set of tags that describe the audio contents. For example it might include the title of the track, the author, and the date it was created. This data will vary depending on the file type, and not all file types store this information so it may be blank. To edit the file's meta-data tags click anywhere in the window. This will open the corresponding file attributes dialog.

You can access this dialog in the Audio File Workspace via **Workspace** > **Specific tool windows** > **Meta-Data**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

11.3.6 Marker Window

The marker window is a specific tool window that allows you to create, edit and use markers while working on an audio waveform.

From the Functions menu you can also access useful functions to convert marker types, rename multiple markers and export the marker list as text.

The Marker list

The marker window contains a list of any markers placed in the currently active file along with their corresponding details. Click on a column header to sort the list in ascending or descending order based on the values in that column. You can also use the Filter menu to toggle which types of markers are displayed in the list.

From this list of markers you can use the fields, controls and menus to:

- **Create a new marker at the playback head position** - click on a marker icon (at the top of the window) or choose a marker type from the Insert menu to drop a marker at the current playhead location.
- **Create a pair of markers from a selection** - make a selection in the waveform and then click on a pair of marker icons (at the top of the window) or choose a "Create region" command from the Insert menu to drop a pair of markers at either side of the current selection.
- **Start playback** - It is possible to start playback at the selected marker position, with or without pre-roll, by clicking on the leftmost icons. Click on in the "audition" column to play the wave from that marker position with a pre-roll. Click on the right hand button to play the wave exactly from the marker position.
- **Change marker types** - You can change a marker type by (left) clicking on its icon and choosing another marker type from the pop-up list that appears.
- **Edit a marker's name/time/comments** - double click in a cell to edit a value (to align the marker to an exact time, for instance). You can edit the name of the marker, its time position and any comments.
- **Lock a marker's position** - select the checkbox to lock the marker. This prevents it from being accidentally dragged to a new position in the Waveform window.

For more information on using markers see Markers. For details on the different marker types and their uses see Marker types. Note that there are more marker features available in the Montage version compared to the Audio File version. See "Using markers in the Audio Montage Workspace" in Markers for more details.

You can access this specific tool window within the Audio Files and Audio Montage Workspaces via **Workspace** > **Specific tool windows** > **Markers**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Markers

Marker types

Convert marker type

Text format

11.3.7 Sample attributes

This dialog allows you to define settings for an audio sample before loading it into a hardware or software sampler.

The settings do not process the sample in any way, they just give the file properties that the receiving sampler can use. This includes information about the pitch of the sample (which can be detected automatically by WaveLab), the key range that the sample should span, and the velocity range to occupy. For WAV and AIFF files, this information is stored in the file's header.

By default, there are no sample attributes in an Audio File. Therefore you must explicitly create this information by clicking on the "Create" button first. This dialog is useful if your sampler can take advantage of these additional tags. If supported by your sampler, it can save you time by allowing you to both edit and define a samples properties from within WaveLab.

You can access this dialog via **Workspace** > **Specific tool windows** > **Sample Attributes...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

11.3.8 Error detection and correction

This specific tool window allows you to search for unwanted clicks and digital artifacts in an Audio File. Several detection and restoration methods are available. You can detect, mark and name, jump to, play back and remove individual audio errors.

The settings for error detection and correction are organized within the following groups of controls:

- **Detection tab** - choose the method used to detect any errors. Each different method has its own settings to determine when it will detect an error. Click Detection 1 and 2 will search for clicks in certain frequency ranges while Digital Click Detection looks for clicks that are caused by clicks typical to digital errors.
- **Correction tab** - specify the method used to correct any errors.
- **Search Area tab** - specify the range of audio you wish to search for errors.
- **Options tab** - this provides a range of preferences for playing back, viewing, and marking any errors found.
- **Browse & Correct panel** - step through your defined search region detecting each error in turn. You can choose to correct any errors or mark them for later correction. There are controls to skip between error markers and make fine adjustments to the

selection. You can also automatically detect and correct all marked errors in the search region.

Strategies to detect and correct errors

There are several strategies that you can employ when detecting and correcting errors:

- Define an audio selection where you have identified an error, then click "Correct", or "Mark for subsequent correction".
- Use the function "Detect next error" to let WaveLab automatically find the next error, and then click "Correct", or "Mark for subsequent correction".
- Use the function "Detect all errors" to let WaveLab automatically find all errors in the predefined range. Then browse the marked errors, removing or adjusting each audio range in turn, and click "Correct" to correct any specific errors, or you can use the option "Correct all".

You can access this dialog by choosing **Workspace** > **Specific tool windows** > **Error Correction**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Marker types

11.3.9 Master Project

This Specific tool window allows you to access and open all the files related to a project from one place: the Master Project window.

If you are working on a large project that consists of several montage files, Audio Files and editing notes (all the files belonging to an album project, for example), you may find it useful to open them from this window. You can organize the files in folders and sub-folders as you wish, with each file annotated with a note using the text editor provided.

You can place documents from any 'draggable' location into the Master Project (and of course, drag files from the Master Project back to external locations). Each file is listed with the pathname of the enclosing Folder and an editable Comment field. As in a standard file system, you can't put two files with the same name in the same folder.

The entire contents of a Master Project can also be archived to CD/DVD via a dedicated command. You can access this feature from the Master Project window via **File** > **Add to Data CD/DVD...**, which is available if the Master Project has been saved to file once (i.e. it is not untitled).

Only **one** Master Project can be opened at a time, but it can appear as a tool window **in each Workspace**. In other words, there is a single content that can be visualized independently in various windows. These windows may be docked or floated.

You can access this Specific tool window from any Workspace via **Workspace** > **Specific tool windows** > **Master Project**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Dragging operations

Recently used files

11.3.10 Focused Clip window

The Focused Clip window is an accordion control that contains some sophisticated editing commands. Each collapsible panel can be expanded by clicking on its title and allows you to access a range of tools for working with the selected or "focused" Clip.

Using the Focused Clip window

To use the Focused Clip window you first need to select a Clip. Once you have selected a Clip with your cursor its name will highlight in red. You can now choose one of the Focused Clip accordion's tabs. The accordion will open to reveal options for that particular pane's selection of editing shortcuts and tools. There are a number of panes including:

- **Edit** - provides you with a range of common playback shortcuts and editing tools for working with Clips.
- **Cue Point** - a cue point is a defined position marker that belongs to a Clip. It helps to align Clips together by making Clips snap to other Clips at the cue point position. Cue points are displayed as dotted vertical lines. This pane provides you with a number of shortcuts for creating and working with Cue Points.
- **Envelope** - allows you to access a number of controls for editing and working with a Clip's different envelopes.
- **Fade-in** - provides you with a range of tools for adjusting a Clip's Fade-in curve.
- **Fade-out** - provides you with a range of tools for adjusting a Clip's Fade-out curve.
- **Colors** - up to 20 colors may be assigned to help distinguish particular Clips visually. Select a Clip then click on a color. Custom colors can be defined with great subtlety from Options > Colors... Select the Custom color to be defined, then the parts to which it is applied, then define the color value using the color wheel (or type a set of RGB - Red, Green, Blue values). See Audio Montage Colors for more information.

When you choose a different Clip the properties in the Focused Clip window will update. So for example, if you select "Envelope" the corresponding controls will update to reflect the selected Clip's envelope information.

The Focused Clip window can be docked or floated. You can reveal or hide the focused Clip window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Focused clip**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Clip

Audio Streaming Settings

Customize Commands

Audio Montage Colors

11.3.11 Snapshots

This dialog enables you to save a number of "views" of your montage. You can then recall a particular view or snapshot at any time.

When you save a snapshot you capture the current zoom factor, the current cursor position, the current scroll position and the current time range displayed. You can name each snapshot and update an existing snapshot too. To recall a snapshot, just click on its title in the list. This restores all of its view settings. You can also choose to recall all or just one of its view properties by clicking the appropriate icon.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Snapshots**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

11.3.12 File

This specific tool window helps you to manage files that are used in the current Audio Montage.

It displays all current files that are being used by Clips in the current montage, along with their location, size and last modified date. In addition a number of file operations are available:

- add one or more Audio Files to a montage, this will open the Insert Audio Files window when one than one file is selected
- replace individual files in a montage (this will update any Clip references)
- rename a file (this updates all internal Clip references)

- select Clips that are referenced by a specific file
- open a file in the Audio File Workspace, and view the file in your computer's file browser.

This window is a useful tool to use when constructing and editing your montage.

You can access this dialog in the Audio Montage Workspace by choosing **Workspace** > **Specific tool windows** > **Files**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Insert Audio Files

11.3.13 Zoom

This specific tool window displays a magnified view of the selected track and allows you to accurately adjust the cross fade point for two adjacent Clips.

You can adjust the zoom level of the selected track, manually shift the Clip's position or allow WaveLab to find the best location to begin a cross fade between two Clips. You can adjust the amount that WaveLab will search for the best cross fade position to avoid phase cancellation issues, and also view the envelope points of the Clips. Mouse operations are identical to those in the main Clip view.

You can access this window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Zoom**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

11.3.14 CD

This specific tool window brings together all the functions for creating an audio CD or DVD-A within WaveLab.

It displays a list of CD tracks along with information about each track. You can edit each track and the CD's playback properties as well as access functions to check its conformity to Red Book standards, add and edit CD-text, add UPC/EAN codes, generate a CD report and burn the CD. There are also controls to preview skipping between each track and a number of other options including the ability to choose the "Audio in Pause" mode to determine whether gaps between track markers are replaced with silence (the normal default) or if they contain sound.

Adding tracks

To create an audio CD in WaveLab you must at least have a CD start and end marker pair. You can generate these automatically from audio Clips in the montage using the CD Wizard or by adding track markers manually.For more information on creating a CD or DVD-A see Authoring CDs and DVDs.

The Track list

Once you have created one or more tracks, the CD window displays a list of all tracks in the currently active montage along with their corresponding properties. From this list of CD tracks you can use the fields, controls and menus to:

- **Audition playback** - It is possible to audition playback at the selected track's start time, with or without pre-roll, by clicking on the leftmost icons. Click on the left hand button in the "audition" column to play the track from its start time with a pre-roll. Click on the right hand button to play the track exactly from its start time.
- **Edit a tracks's name** - You can add/edit a name for each track. Double click in a cell to edit. Note that to change the track name is to indirectly edit the marker name.
- **Edit a track's start/end time** - You can edit the start and end time for each track. This is the same as moving the track markers physically but can be done with more precision. Double-click in a cell to edit.
- **Pre-gap time** - This displays the gap between the CD track start marker and where the first track's audio actually begins. It can only be edited by physically moving the CD track start marker.
- **Post-gap time** - This displays the gap between the end of the audio of the CD's last track and the CD track end marker. It can only be edited by physically moving the CD track end marker.
- **Copy protection** - Select the checkbox to include a copy protection flag with the track. This may prevent it from later being imported by software that recognises this flag.
- **Emphasis** - This setting is used to indicate if the track was recorded with emphasis or not. Please note that this does not apply/remove emphasis from the audio, it is just an indicator as to how the file was created. Normally it is left off.
- **ISRC Code** - This allows you to enter an optional International Standard Recording Code. Double-click in a cell to add/edit a code.
- **CD-Text** - You can optionally enter CD-Text information for each track and the entire CD/DVD-A. Double clicking a cell will open the CD Text Editor.
- **Add/Edit a tracks's comments** - You can add/edit text comments for a track. Double click in a cell to edit. Note that this text comment is for reference only: nothing is burnt on CD.

Note that CD tracks can be reorderd by dragging them into new positons: this means Clips and markers are moved accordingly.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Authoring CDs and DVDs

Write Audio CD

CD Wizard

CD Text Editor

Audio CD Report

CD Pre-emphasis

Red Book CD-DA

CD Text

ISRC

UPC/EAN

11.3.15 Clips

This specific tool window contains a list of any Clips placed in the currently active montage file together with their corresponding details.

Click on a column header to sort the list in ascending or descending order based on the values in that column. Clicking on any row of the list will move the main window's view to show the selected Clip. To change a value, double click in a cell to begin editing (to align the Clip to an exact time, for instance). Clips may be dragged into new positions in the list.

From this list of Clips you can use the input fields and controls to:

- **Start playback** - It is possible to start playback at the selected Clip position, with or without pre-roll, by clicking on the leftmost icons. Click on in the "audition" column to play the montage from that Clip's position with a pre-roll. Click on the right hand button to playback the montage exactly from the start of the Clip.

- **Edit a Clips's name/start/end/length/comments** - edit the name of the Clip, its start/end time position and any comments.

- **Lock a Clips's position** - select the checkbox to lock the Clip. This prevents it from being accidentally edited or dragged to a new position.

- **Change Gain** - use this control to increase/decrease the gain of the Clip. Double click in the cell to edit the value.

- **Mute** - select this to mute the Clip. Its color changes in the track view and it is no longer audible during playback.

A wide range of Clip selection features, Clip functions, and Clip display options are available from the Clips menu. Use "What's This?" help for information on individual menu entries.

You can access this window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Clips**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Audio Montages

Clip

11.3.16 Notes

This specific tool window allows you to keep notes about the current Audio Montage session.

You can type into the window and use the standard HTML text editor controls to format your text, add images, hyperlinks and lists. These notes are saved along with the Audio Montage file.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Notes**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

11.3.17 Effects

This specific tool window allows you to add VST effect plug-ins, both to individual Clips and tracks. The difference between Clip effects and track effects are that Clip effects affect individual Clips only, whereas track effects affect all Clips on a track.

Each independent Clip and/or audio track in the Montage can be independently processed by up to ten VST effect plug-ins. Effects are configured either as Inserts, when all sound is processed by the effect, or as Send effects where the balance between the unprocessed sound and the effect send level can be adjusted or controlled by effect envelope curves (clip effects only). Note that not all effects are capable of "Send", and if this feature is not available, it is disabled.

The window only shows the effects of the focused Clip, or the focused track according to the active mode. Each plugin and track can be understood as being associated with a small "Master Section". A plug-in can be inserted and sorted, and you double-click on an effect to Edit it. When you close an effect window, its parameters are saved into the montage and it can be be undone if necessary. Use "What's This?" help on the column headers for more details.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Effects**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

11.3.18 DVD-Audio

This specific tool window allows you to author a DVD-Audio disk from a collection of audio montages.

You can add your finished audio montages into this window, check the DVD-Audio layout for conformity and output a DVD-Audio disk all from this window. **Note:** All included montages need to be in DVD-A mode (this can be set via Audio Montage properties).

Although DVD-A has a different specification from an Audio CD, its track concept makes is similair to an Audio CD - this is why creating a DVD-A is almost like creating a CD with WaveLab's Audio Montage. It is worth noting that throughout WaveLab, where "CD" is mentioned (for example, in messages), one can usually substitute "DVD-A".

To create a DVD-Audio disk, follow these basic steps:

- **Create an audio montage** - you need at least one audio montage in order to create a DVD-A. You can import any currently open ones or add them as *.mon files. Note that the Audio Montage must have DVD tracks defined with red (track) markers.

- **Add Montage(s)** - choose **File** > **Add Audio Montage(s)** from the DVD-Audio window to add one or more montages to your DVD-Audio. Each montage will then be displayed along with its details in the DVD-Audio window. You can drag the montages vertically to determine the track order. Montages can also be added by dragging montage files directly into the DVD_Audio track listing window.

- **Choose Options** - from the DVD-Audio window choose **Options** > **Options...** to open DVD-Audio options dialog. You can then define settings for your DVD-A including its name and volume information details, whether it will be played back on PAL or NTSC, and the duration and effects/transitions that included images will use. Images to be included can be laid out in a picture track in your audio montage or a single default image can be chosen. If a default image is used it will displayed at the beginning of each montage

- **Check and burn your DVD-A** - once you are finished laying out your DVD-A and choosing any options, you can check that it conforms to DVD-Audio specifications by choosing **Edit** > **Check DVD-Audio conformity...**. If everything is OK, choose **Edit** > **Burn DVD-Audio...** to open the DVD-Audio Creation dialog where you can then render out the disk as a collection of files ready for burning. Once these files are successfully outputted the Write Data CD/DVD dialog will open automatically allowing you to burn your DVD-Audio files to an optical DVD.

Note that WaveLab automatically creates some basic DVD menus to access the audio tracks from the DVD-Player.

For more information on authoring DVD-Audio disk see Authoring CDs and DVDs.

You can access this dialog in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **DVD-Audio**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

DVD-Audio Creation

DVD-Audio options

Write Data CD/DVD

11.3.19 Scripting window (Audio Montage Workspace)

This window allows you to write and execute scripts in the Audio Montage Workspace.

The inbuilt text editor helps you when writing scripts by highlighting the different parts of the script with colors, making it more readable. A script can also be written in another text editor, and loaded via the File menu. To run a script, from within the script window choose **Functions** > **Execute script**.

For a general description of scripting, see Scripting. For an overview of the scripting language, see ECMAScript Reference.

You can access this dialog from the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Script**.

Note: To view trace or log messages when executing scripts that contain the logWindow() function, ensure that the Log window is visible and that it's warning filter buttons - ! ⚠ ⓘ are selected.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Scripting

ECMAScript Reference

Log window

11.3.20 History

This specific tool window allows you to view a history of all your recent editing operations.

You can revert the current montage to a previous state by selecting the operation you want to return to and double clicking on it. You can then use the redo command (**Edit** > **Redo**) to

return to any state. Use the history menu options (**Edit** > **History** >) to undo all operations, redo all operations, or clear the history.

There are a couple of history options available in the Audio Montage preferences window. You can choose to empty the history each time you save a montage file and also to group similar types of sequential operations into a single operation. Both of these options enable you to use less memory when storing the operation history.

You can access this window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **History**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montage Preferences

11.3.21 Navigator

This specific tool window displays a representative view of the entire active montage and enables you to quickly navigate around it.

Each track is represented by a horizontal colored block. By default, stereo tracks are dark blue and monotracks are purple or pink (any custom colors you set for a track or via the Groups window are also displayed).

The visible Workspace is shown by a surrounding black rectangle. Dragging this rectangle scrolls the main montage window, enabling you to easily navigate to a location in your montage. You can also resize the rectangle vertically and horizontally to adjust the zoom setting of the active window. This window can be useful when you have a large montage with many Clips and tracks.

- Left click on a Clip block to zoom on it
- Right click anywhere to completely zoom out

You can access this window in the Audio Montage Workspace by choosing **Workspace** > **Specific tool windows** > **Navigator**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio Montages

Groups

11.3.22 Groups

This specific tool window displays a list of Groups within the current Audio Montage. Groups are a selection of Clips that you can quickly re-select via the Groups window. You can group

a number of Clips and nest Groups inside each other. It is also possible to disable individual Groups and color Groups so you can identify them easily.

Grouping Clips

To create a Group from a number of Clips:

1. Select the Clips you want to group.
2. Make sure you have the Groups window displayed.
3. Select "Group selected clips" from the drop down menu or click the Group icon.
4. Enter a name for the Group in the dialog that appears and click OK.

The new Group appears in the Group list. All Clips included in the Group will also have the name of the Group prepended to the Clip names in each track. Clicking on a Group in the Groups window will select all the Clips in your Montage so you can edit them together.

Adding Clips to an existing Group

If you already have a Group, and want to add one or several more Clips to it:

1. Select the Clips you want to add.
2. In the Groups view, select "Group selected clips", or click the corresponding icon.
3. In the dialog that appears, select the Group to which you want to add the Clips.
4. Click OK.

Removing Groups

1. Select the Group by clicking on its name in the Groups view list.
2. Pull down the Grouping menu and select "Remove selected group". The Group is removed (the Clips are not affected).

Using Groups

There are some important things to remember when using Groups:

- A Clip cannot be part of more than one Group. If you add a Clip to a Group, it is automatically removed from any other Group it is part of.
- It is possible to create nested Groups, by adding a Group to another Group.
- You can temporarily "turn off" a Group by deactivating the checkbox to the left of a Group in the list. When a Group is deactivated, you can move individual Clips in the Group, as if they were not grouped.
- You can select a specific color for a Group to make it easy to discern it in the track view.

- Double-click to rename a Group
- You can nest a Group inside another Group by dragging it
- The Render option is available to render all Groups as individual files (Render dialog)

You can access this window in the Audio Montage Workspace via **Workspace** > **Specific tool windows** > **Groups**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Render Montage window

Chapter 12

Audio Plug-ins

Steinberg created Virtual Studio Technology (VST) to allow effect plug-ins to be integrated with audio editors (such as WaveLab). VST uses Digital Signal Processing (DSP) to closely simulate in software the effects of familiar recording studio hardware. A vast number of plug-ins are available - from freeware to high-end commercial products - and can greatly extend the functionality of programs such as WaveLab.

The order of processing is significant. You can change the order in which effects are processed by moving the effects icons by dragging them between slots. See Dragging operations for more information.

WaveLab provides slots for up to ten plug-ins.

The more complex plug-ins provide a custom GUI, often displaying controls similar to the physical switches and knobs on audio hardware; other plug-ins rely on the host application for their UI.

The following plug-ins are supplied with WaveLab:

ASIO plug-ins

Audio Input

Using External effects

Legacy plug-ins

Under Windows, a set of plug-ins is provided for compatibility with audio projects that referenced these effects when using earlier versions of WaveLab. An Audio Montage which referenced these plug-ins would otherwise require cumbersome user intervention to open, for instance. Their use with new audio projects is not recommended and they are not documented.

Sonnox Ltd VST-3 plug-ins

Sonnox DeClicker

Sonnox DeNoiser

Sonnox DeBuzzer

Steinberg VST-3 plug-ins

Steinberg Autopan
Steinberg Chorus
Steinberg Compressor
Steinberg DeEsser
Steinberg Mono Delay
Steinberg Distortion
Steinberg Dual Filter
Steinberg Envelope Shaper
Steinberg Expander
Steinberg Gate
Steinberg GEQ-10/GEQ-30
Steinberg Limiter
Steinberg Maximizer
Steinberg Mix 6 to 2
Steinberg Mix 8 to 2
Steinberg Mono to Stereo
Steinberg Multiband Compressor
Steinberg Octaver
Steinberg Pingpong Delay
Steinberg PostFilter
Steinberg Roomworks
Steinberg Roomworks SE
Stereo Delay
Steinberg Stereo Enhancer
Steinberg Studio Chorus
Steinberg Studio EQ
Steinberg Test Generator
UV22HR Dithering
Steinberg Vintage Compressor
Steinberg VST Dynamics

Built-in plug-ins

Crystal Resampler

Leveler

Leveler Multi

Peak Master

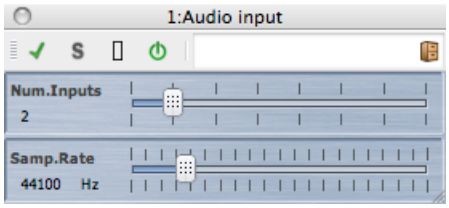
Silence

Stereo Expander

Dithering plug-ins

Internal Dithering

12.1 Audio Input



The Audio Input is a special Master Section plug-in that allows you to render a signal coming in to a sound card's input(s) together with any Master Section effects. This signal can be anything your sound card accepts, for example a feed from a mixer, a recorder or a microphone.

By default, this plug-in is located in the ASIO sub-menu of the Master Section effects. An ASIO driver must be used, and only one instance of this plug-in is allowed in the Master Section plug-in chain.

To use the External gear plug-in, proceed as follows:

1. Set the Audio Device to ASIO

You can access this dialog from Workspaces that support the recording of Audio Files by choosing **Options** > **Audio Streaming Settings...**. Set the Audio Device to "ASIO".

2. Assign the Device Input channels

From the Recording tab, select the channels to be used for device input and name them for use in WaveLab. Note that you only have to set the ASIO inputs – the ASIO outputs are not relevant for Audio Input. Press "OK" when you are done.

3. Set options for the Audio Input plug-in

Click in the top Master Section effect slot and select "Audio Input" from the ASIO plug-in sub-menu. The Audio Input control panel opens and you can set the following options:

- "Num.Inputs" Using the slider, set this to the number of inputs assigned in step 2.
- "Samp.Rate" Using the slider, set this to the sampling rate required (from 11025 to 19200Hz).

4. Monitor the Audio input

Press Play on the transport. Normal wave playback does not happen (the cursor does not move), but the Play button is lit and you can now monitor the input source. Pressing Stop ends input monitoring.

- If you change the settings in the control panel, press Stop and restart playback to apply them. Note that if you are using any non-multichannel plug-ins in the Master Section effect chain, these are limited to two channels. You also have to match the number of inputs in the ASIO Connections dialog to the number of inputs selected here - a warning message appears if they do not match.

- Click the Render button. A dialog appears allowing you to select a name, an audio format and a location for the file to be rendered. When you click OK, recording (or rendering) starts, recording the external input from the output of the Master Section, including all real-time processing. You can monitor the recording as it happens.

- Click Stop on the transport to stop the recording/rendering.

Related topics

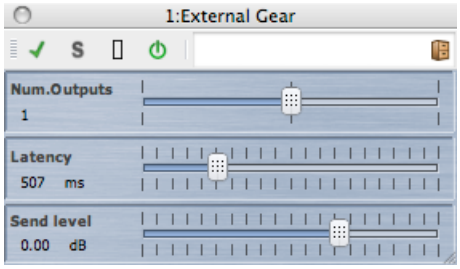
Master Section

Audio Streaming Settings

How do I add effects?

Presets

12.2 Using External effects



The External Gear Master Section plug-in allows you to process Audio Files using external hardware processors. One or more ASIO outputs are used to send the audio signal to your processor, and corresponding ASIO inputs are used to return the signal from the external processor.

By default, this plug-in is located in the ASIO sub-menu of the Master Section effects. An ASIO driver must be used, and only one instance of this plug-in is allowed in the Master Section plug-in chain.

To use the External Gear plug-in, proceed as follows:

1. Set the Audio Device to ASIO

You can access this dialog from Workspaces that support the playback of Audio Files by choosing **Options** > **Audio Streaming Settings...**. Set the Audio Device to "ASIO".

2. Assign the output and input channels

From the External gear plug-in tab, select the channels to be used for device output (to gear) and for device input (from gear). These should normally be different I/O channels than the ones you use for playback/recording. The number of available outputs in this plug-in equals the number of inputs (up to 8). Press "OK" when you are done.

3. Set options for the External Gear plug-in

Click in a Master Section effect slot and select "External Gear" from the ASIO plug-in sub-menu. The External Gear control panel opens and you can set the following options:

- "Num.Outputs" Normally the number of outputs is the same as the number of inputs (the "As Inputs" option). You can however use a mono in/stereo out configuration in which case, set this parameter to "2" with the slider.

- "Latency" External gear may introduce latency. WaveLab can automatically compensate for this if you select "Auto" (only active during rendering), or you can set this latency compensation yourself (up to 2000 milliseconds). The latency introduced by the ASIO driver is automatically taken into account by WaveLab.

- "Send level" The Send level should normally be set to 0 dB, but you can adjust the input level on the external effect, if necessary.

4. Process the signal

Now you can process a signal through the external processor, just as if it was a software plug-in effect. If you render a file using the External Gear plug-in, playback is not available during the rendering.

Related topics

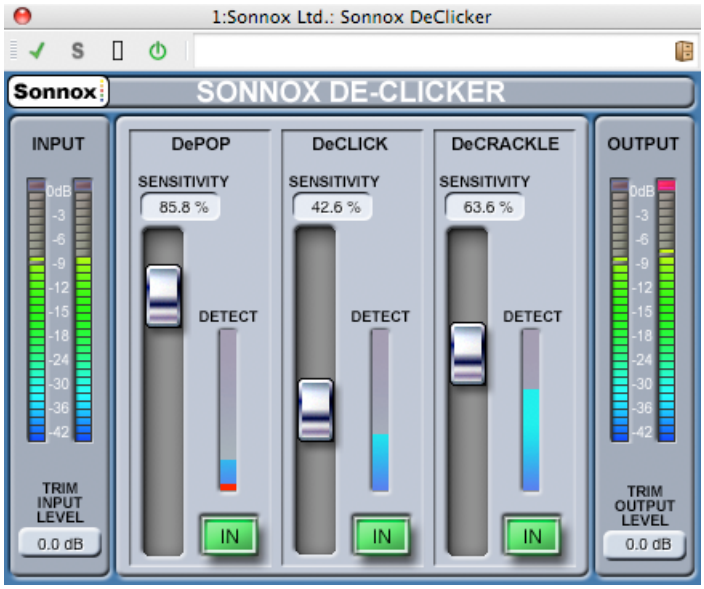
Master Section

Audio Streaming Settings

How do I add effects?

Presets

12.3 Sonnox DeClicker



Sonnox DeClicker is one of three plug-ins that form the Sonnox Restoration Toolkit. The tools are not just for restoring old material, but also for removing the clicks, pops, buzzes and background noise that can sometimes accidentally occur in new recordings.

A note on Touch Pad controls

Several controls in these plug-ins are Touch Pads. These controls give a clear display of the numeric value of a control, and respond to the following actions:

- Left-Click and drag up/down: Increase/decrease the value
- Right-Click and drag up/down: Fine adjustment of value
- Double Click: Directly edit value
- Shift + Left-Click and drag up/down: Fine adjustment of value
- Control + Left-Click (Windows), Cmd + Click (Mac): Set to default

Additionally, for all controls (including faders and knobs, not just Touch Pads)

- Control + Left-Click (Windows), Cmd + Click (Mac): Set to default

DeClicker screen controls

- **Sonnox Menu Options Button**
  Clicking this button brings down a menu of options and preferences. It allows choice of:
  the input/output meter clip light hold duration (permanent, 2s, 5s), and
  the splash screen to examine the version number and build date.

- **Input Level Meter**
  The Input Meter is designed to give exactly 1dB per LED for the top 18dB of dynamic range, and 2dB per LED thereafter. This gives a clear and intuitive impression of the working headroom.

- **Input Gain Trim Touch Pad (dB)**
  This allows you to adjust the input signal level by up to ±12dB.

- **Sensitivity Fader and Touch Pad (%) (Pop; Click; Crackle)**
  This controls the sensitivity of the detection circuits. Fully sensitive might allow the detection circuit to react to low level signals and possibly mis-classify programme as pops or clicks. Stronger pops and clicks will require a less sensitive setting.

- **In Button (Pop; Click; Crackle)**
  This button enables the pop, click or crackle removal processing. It is important to note that with this button "off" (i.e. the repair processing is disabled), the pop, click or crackle detection circuit is still enabled and the detect display will still show the degree of event detection.

- **Detect Meter Display (Pop; Click; Crackle)**
  The Detect meter display combines two indications. The main rising column indicates the sum of the energy of events that have been detected. With the "In" button de-selected (i.e. the repair circuit disabled), this column is colored orange/red. With the repair circuit active the column is colored blue. The bottom segment of the meter is an indication of each individual detected event.

- **Output Level Meter (dB)**
  The output meter is designed to give exactly 1dB per LED for the top 18dB of dynamic range, and 2dB per LED thereafter. There is a peak-hold feature that holds the highest peak, helping to give a better impression of the working dynamic range.

- **Output Gain Trim Touch Pad (dB)**
  This allows you to reduce the output level by up to 12dB. Dithering is applied after output gain control, so it may be necessary to reduce this value by a small amount to avoid clipping.

Quick-Start Guide for the De-Clicker

The De-Clicker is very simple and intuitive to use. It is generally wise to repair the larger and more energetic events first.

1. If there are large displacement events in the programme material, enable the DePop section and raise the sensitivity fader until the largest events are detected and repaired.
2. For clicks, enable that section and raise the sensitivity fader until they are detected and repaired.
3. Finally if there is crackle left, enable that section and raise its fader to remove the crackle.

Tip: There is necessarily some degree of overlap in the detection circuits of clicks and crackle. Decreasing the DeClick sensitivity might increase the apparent detection of crackle; and increasing the DeClick sensitivity might indicate less crackle. Best results are likely if the two controls are balanced.

Related topics

Sonnox DeBuzzer

Sonnox DeNoiser

Audio Plug-ins

Master Section

How do I add effects?

Presets

12.4 Sonnox DeNoiser

Sonnox DeNoiser is one of three plug-ins that form the Sonnox Restoration Toolkit. The tools are not just for restoring old material, but also for removing the clicks, pops, buzzes and background noise that can sometimes accidentally occur in new recordings.

A note on Touch Pad controls

Several controls in these plug-ins are Touch Pads. These controls give a clear display of the numerical value of a control, and respond to the following actions:

- Left-Click and drag up/down: Increase/decrease the value
- Right-Click and drag up/down: Fine adjustment of value
- Double Click: Directly edit value
- Shift + Left-Click and drag up/down: Fine adjustment of value
- Control + Left-Click (Windows) Cmd + Click (Mac): Set to default

For circular controls:

- the default action is Circular mode, but if the Alt key is held, the action is Linear mode.

Additionally, for all controls (including faders and knobs, not just Touch Pads)

- Control + Left-Click (Windows), Cmd + Click (Mac): Set to default

DeNoiser screen controls

- **Sonnox Menu Options Button**

  Clicking this button brings down a menu of options and preferences. It allows choice of:

  the input/output meter clip light hold duration (permanent, 2s, 5s), and

  the splash screen to view the version number and build date.

- **Graphical Display**

  The graphical display shows the real-time frequency/gain curve of the programme material. It is graduated from 0 to 20kHz and from 0 to -144dB. The yellow line is the calculated noise spectrum level, and in Auto mode will continually follow the noise in real time. Everything below this contour is assumed to be noise, and everything above the line is programme signal.

- **Input Level Meter**

  The Input Meter is designed to give exactly 1dB per LED for the top 18dB of dynamic range, and 2dB per LED thereafter. This gives a clear and intuitive impression of the working headroom.

- **Input Gain Trim Touch Pad (dB)**

  This allows you to adjust the input signal level by up to ±12dB.

- **Sensitivity Fader and Trim Touch Pad (dB)**

  The sensitivity fader defaults to 0.0dB, which is the midpoint of its travel. It will adjust the sensitivity of the noise detection circuit, and the visible effect of this is to move the yellow noise contour line up and down. The sensitivity level can be changed by up to ±18dB. To reduce the sensitivity and make the De-Noiser less reactive to the noise component, move the fader down. The noise contour will displace downwards, showing less noise component in the detection circuit. If the sensitivity is set too low, little noise reduction will occur. Conversely, to increase the sensitivity and make the De-Noiser more reactive to the noise component, move the fader up. The noise contour will displace upwards, showing more noise component in the detection circuit. The default setting is for the noise contour to lie just below the peaks of the signal. Making the detection circuit more sensitive to noise will decrease the signal component, possibly pushing the contour up towards the peaks of the signal. In this case, it is likely that processing artifacts will be heard, as the noise removal circuit will be acting on the signal component as well as the noise component.

- **Adapt Button**

  This button enables Adapt mode for the noise detection circuit. In this mode the noise fingerprint is continually calculated and updated. This mode is useful for material with a time-varying noise component.

- **Freeze Button**

  This button enables Freeze mode for the noise detection circuit. In this mode the noise fingerprint is calculated once (when the Freeze button is pressed). This mode is useful for material with a constant noise component, and would typically be sampled when the signal is absent and only the noise component is present.

- **In Button**

  This button enables the noise removal processing. It allows glitch-less comparisons with and without the noise reduction. It is important to note that with this button "off" (i.e. the noise removal processing is disabled), the noise detection circuit is still enabled and the graphical display will still show the real-time frequency display and the noise contour line.

- **HF Limit Knob and Touch Pad (Hz)**

  This displays and controls the frequency beyond which the attenuation is applied non-dynamically. Scrolling the frequency down from the default of 22kHz will show a red region in the frequency display that has a fixed attenuation. To the left of the HF Limit line the noise removal circuit behaves as normal. To the right the signal is attenuated by a fixed amount set by the attenuation fader. This mode is useful for band-limited programme material. A good example is a low bitrate encoded signal, which might be band limited to 12kHz. Due to the sharp discontinuity, the noise removal circuit can introduce audible artifacts around the band limit, and setting the HF Limit frequency slightly lower than the band limit will remove those artifacts.

- **Attenuation Fader and Touch Pad (dB)**

  The attenuation fader and touch pad determine the level of attenuation that the noise removal circuit will apply in the range 0 to -18 dB. Generally this should be set so that the noise reduction is pleasing. Excessive use of attenuation can degrade the signal unnecessarily.

- **Output Level Meter (dB)**

  The output meter is designed to give exactly 1dB per LED for the top 18dB of dynamic range, and 2dB per LED thereafter. There is a peak-hold feature that holds the highest peak, helping to give a better impression of the working dynamic range.

- **Output Gain Trim Touch Pad (dB)**

  This allows you to reduce the output level by up to 12dB. Dithering is applied after output gain control, so it may be necessary to reduce this value by a small amount to avoid clipping.

Quick-Start Guide for the De-Noiser

1. Start with Sensitivity and Attenuation controls at the default positions (0.0dB and -4.5dB respectively).
2. Select Auto mode if the noise varies in time; select freeze for a defined and static noise fingerprint.
3. Adjust the Sensitivity to find the correct balance between being too low (not enough noise is removed) and too high (too much signal is removed).

4. Adjust the Attenuation to find the most pleasing audio; too much attenuation can impair the audio, either by reducing brightness or by introducing low-level distortion.

Tip: You might be working with bandwidth-limited material, possibly as a result of samplerate conversion or lossy compression (for example limited at around 10kHz). If you experience distortion around the limit try reducing the HF Limit control. Adjust until it lies just to the lower frequency side of the limit (around 9.5kHz in our example).

Related topics

Sonnox DeBuzzer

Sonnox DeClicker

Audio Plug-ins

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How do I add effects?

Presets

12.5 Sonnox DeBuzzer

Sonnox DeBuzzer is one of three plug-ins that form the Sonnox Restoration Toolkit. The tools are not just for restoring old material, but also for removing the clicks, pops, buzzes and background noise that can sometimes accidentally occur in new recordings.

A note on Touch Pad controls

Several controls in these plug-ins are Touch Pads. These controls give a clear display of the numeric value of a control, and respond to the following actions:

- Left-Click and drag up/down: Increase/decrease the value

- Right-Click and drag up/down: Fine adjustment of value
- Double Click: Directly edit value
- Shift + Left-Click and drag up/down: Fine adjustment of value
- Control + Left-Click (Windows), Cmd + Click (Mac): Set to default

For circular controls:

- the default action is Circular mode, but if the Alt key is held, the action is Linear mode.

Additionally, for all controls (including faders and knobs, not just Touch Pads)

- Control + Left-Click (Windows), Cmd + Click (Mac): Set to default

DeBuzzer screen controls

- **Sonnox Menu Options Button**

  Clicking this button brings down a menu of options and preferences. It allows choice of:

  the input/output meter clip light hold duration (permanent, 2s, 5s),

  a default to 50Hz on start-up, and

  the splash screen to view the version number and build date.

- **Input Level Meter**

  The Input Meter is designed to give exactly 1dB per LED for the top 18dB of dynamic range, and 2dB per LED thereafter. This gives a clear and intuitive impression of the working headroom.

- **Input Gain Trim Touch Pad (dB)**

  This allows you to adjust the input signal level by up to ±12dB.

- **Frequency Knob and Touch Pad (Hz)**

  The De-Buzzer has an active frequency range for the buzz fundamental of between 20Hz and 440Hz. In Auto mode, this knob sets the frequency from which the buzz detection circuit will start to hunt for buzz components. In Freeze mode, this knob sets the exact frequency of the buzz fundamental. The knob is graduated around the circumference, and clicking on any labeled graduation will set the frequency to that graduation. The frequency can be scrolled from the touch pad.

- **Fine Adjust Button**

  This button enables fine tuning of the buzz frequency control. The graduations around the circumference of the frequency knob will re-draw to a finer scale, and scrolling the touch pad will enable very quick fine tuning of a hunt frequency. Scrolling past an end-stop will continue to scroll the frequency and the marked graduations will re-draw appropriately. Fine Adjust Mode forces Freeze, so that the selected frequency can be specified exactly, without the Auto circuitry hunting for a stronger fundamental. If entering Fine Mode from Auto, the Freeze button will flash and the plug-in will revert to Auto when Fine mode is exited.

- **Tone On Button**

  This button enables an audible tone generator, which can be used to aid location of the buzz fundamental. While the Tone button is on, a touch pad appears above the button and becomes a Tone level control. It defaults to -18dB, and has a range of -6dB to -96dB.

- **Sensitivity Knob and Touch Pad (%)**

  This controls the sensitivity of the buzz detection circuit. Fully sensitive might allow the detection circuit to lock to inaudible and possibly undesirable frequencies. Stronger buzzes, which typically would be removed first, will require a less sensitive setting.

- **Hum/Buzz Mode Button**

  This control switches between Hum mode and Buzz mode. In Hum mode the bandwidth limit for harmonic removal is 0 to 800Hz. In Buzz mode the bandwidth limit for harmonic removal is 0 to 4000Hz. Hum mode is less damaging, and should be used when possible.

- **Enable Button**

  This button enables the buzz removal processing. It allows glitch-less comparisons with and without the buzz removal. It is important to note that with this button "off" (i.e. the buzz removal processing is disabled), the buzz detection circuit is still enabled and the Detect display will still show the degree of buzz detection.

- **Reduction Display**

  The Reduction display indicates the level of audio that is being removed from the signal.

- **Attenuation Knob and Touch Pad (dB)**

  The attenuation knob and touch pad determine the level of attenuation that the buzz removal circuit will apply, up to a maximum of 96dB. Generally this should be set so that the buzz is just inaudible. Excessive use of attenuation can degrade the signal unnecessarily.

- **Auto Button**

  This button enables Auto mode for the buzz detection circuit. In this mode the buzz detection is continually calculated and a slow drift in the buzz fundamental frequency will be automatically followed. This mode is useful for material with a time-varying buzz component. In this mode the removal filters follow the Detected frequency.

- **Freeze Button**

  This button enables Freeze mode for the buzz detection circuit. In this mode the buzz fundamental is fixed to the frequency shown in the touch pad window. This mode is useful for material with fluctuating buzz level, but with a constant buzz frequency. In this instance, Auto mode would suffer when the buzz level drops and would typically re-hunt for a different buzz fundamental. In this mode the removal filters follow the Nominal frequency.

- **Detect Display**

  This display indicates the degree of detection that the buzz detection circuit has achieved.

- **Output Level Meter (dB)**

  The output meter is designed to give exactly 1dB per LED for the top 18dB of dynamic range, and 2dB per LED thereafter. There is a peak-hold feature that holds the highest peak, helping to give a better impression of the working dynamic range.

- **Output Gain Trim Touch Pad (dB)**

  This allows you to reduce the output level by up to 12dB. Dithering is applied after output gain control, so it may be necessary to reduce this value by a small amount to avoid clipping.

Quick-Start Guide for the De-Buzzer

1. Find the nominal frequency. Start with Sensitivity and Attenuation controls at the default positions (90% and -48dB respectively).
2. If you know the rough frequency of the nominal, select that frequency using either the knob or by typing into the touch pad.
3. In Auto mode, allow the detector time to drift towards the actual fundamental. The detect display will indicate confidence of hum detection. Auto mode should be used if the fundamental drifts over time.
4. Freeze mode should be used to select a specific frequency that might be variable in strength. Fine Adjust (which forces freeze mode) can be used to increase the resolution of selecting the fundamental.
5. If you are still having difficulty finding the fundamental, try using the Tone control.
6. The Hum mode removes harmonics up to 800 Hz. If you can hear harmonics that are higher in frequency, try selecting Buzz mode, which will remove harmonics up to 4000Hz. If there are no harmonics above 800Hz, be sure to use Hum mode to preserve as much original audio as possible.
7. Also in order to cause as little damage to the audio as possible, back off the attenuation until you can just hear the buzz, then increase it a little until the buzz is inaudible.
8. Now reduce the sensitivity until the buzz is just inaudible.

Tip: You can use the high resolution FFT in WaveLab to help identify the exact fundamental frequency.

Related topics

Sonnox DeClicker

Sonnox DeNoiser

Audio Plug-ins

Master Section

How do I add effects?

Presets

12.6 Steinberg Autopan

This is a simple autopan effect. It can use different waveforms to modulate the left-right stereo position (pan), using manual modulation speed settings.

The parameters are as follows:

Rate

Sets the auto-pan speed from 0.1 to 10, by rotating the knob by dragging, or using the mouse wheel.

Width

This sets the depth of the Autopan effect, that is, how far out to the left/right speaker the sound should move, from 0% to 100%.

Shape

Sets the modulation waveform. Sine produces a smooth sweep. Triangle creates a ramp (a sweep from one speaker to the other and then a quick jump back).

Related topics

Audio Plug-ins

Master Section

How do I add effects?

Presets

12.7 Steinberg Chorus

This is a single stage chorus effect. It works by doubling whatever is sent into it with a slightly detuned version.

The parameters are as follows:

- **Rate**

  The sweep rate can be set freely with the Rate knob, without sync to tempo.

- **Width**

  Determines the extent of the chorus effect. Higher settings produce a more pronounced effect.

- **Spatial**

  This sets the stereo width of the effect. Turn clockwise for a wider stereo effect.

- **Mix**

  Sets the level balance between the dry signal and the effect. If Chorus is used as a send effect, this should be set to maximum as you can control the dry/effect balance with the send.

- **Shape**

  This switch changes the shape of the modulating waveform, altering the character of the chorus sweep. Sine and triangle waveforms are available.

- **Delay**

  This parameter affects the frequency range of the modulation sweep, by adjusting the initial delay time.

- **Filter Lo/Hi**

  These parameters allow you to roll off low and high frequencies of the effect signal, respectively.

Related topics

Audio Plug-ins

Master Section

How do I add effects?

Presets

12.8 Steinberg Compressor

Compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both.

Compressor features separate controls for threshold, ratio, attack, hold, release and make-up gain parameters. Compressor features a separate display that graphically illustrates the compressor curve shaped according to the Threshold and Ratio parameter settings. Compressor also features a Gain Reduction meter that shows the amount of gain reduction in dB, Soft knee/Hard knee compression modes and a program-dependent Auto feature for the Release parameter.

The following parameters are available:

- **Threshold (-60 to 0dB)**

  Determines the level where Compressor "kicks in". Signal levels above the set threshold are affected, but signal levels below are not processed.

- **Ratio (1:1 to 8:1)**

  Sets the amount of gain reduction applied to signals over the set threshold. A ratio of 3:1 means that for every 3dB the input level increases, the output level will increase by only 1dB.

- **Soft Knee button**

  If this button is off, signals above the threshold are compressed instantly according to the set ratio (hard knee). When Soft Knee is activated, the onset of compression is more gradual, producing a less drastic result.

- **Make-up (0 to 24dB or Auto mode)**

  This parameter is used to compensate for output gain loss, caused by compression. If the Auto button is activated, the knob becomes dark and the output is automatically adjusted for gain loss.

- **Attack (0.1 to 100ms)**

  Determines how fast Compressor will respond to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.

- **Hold (0 to 5000ms)**

  Sets the time the applied compression will affect the signal after exceeding the threshold. Short hold times are useful for "DJ-style" ducking, while longer hold times are required for music ducking, e.g. when working on a documentary film.

- **Release (10 to 1000ms or Auto mode)**

  Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, Compressor will automatically find an optimal release setting that varies depending on the audio material.

- **Analysis (0 to 100) (Pure Peak to Pure RMS)**

  Determines whether the input signal is analyzed according to peak or RMS values (or a mixture of both). A value of 0 is pure peak and 100 pure RMS. RMS mode operates using the average power of the audio signal as a basis, whereas Peak mode operates more on peak levels. As a general guideline, RMS mode works better on material with few transients such as vocals, and Peak mode works better for percussive material with a lot of transient peaks.

- **Live button**

  When this button is activated, the "look ahead" feature of Compressor is disengaged. Look ahead produces more accurate processing, but adds a certain amount of latency as a trade-off. When Live mode is activated, there is no latency, which might be better for "live" processing.

Related topics

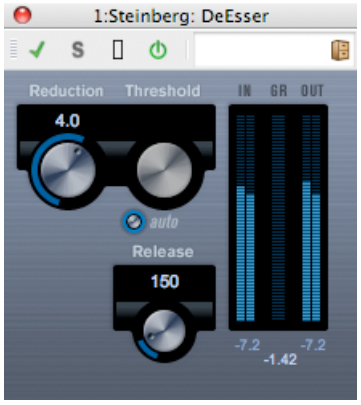
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.9 Steinberg DeEsser



A de-esser is used to reduce excessive sibilance, primarily for vocal recordings. Basically, it is a special type of compressor that is tuned to be sensitive to the frequencies produced

by the "s" sound, hence the name de-esser. Close proximity microphone placement and equalizing can lead to situations where the overall sound is just right, but there is a problem with sibilant (hissing) sounds.

The following parameters are available:

- **Reduction**

  Controls the intensity of the de-essing effect.

- **Threshold**

  When the Auto Threshold option is deactivated, you can use this control to set a threshold for the incoming signal level, above which the plug-in starts to reduce the sibilants.

- **Auto**

  The Auto Threshold function automatically and continually chooses an optimum threshold setting independent of the input signal. The Auto Threshold function does not work for low-level signals (< -30db peak level). To reduce the sibilants in such a file, set the threshold manually.

- **Release**

  Sets the amount of time it takes for the de-essing effect to return to zero when the signal drops below the threshold value.

- **Level meters**

  Indicate the dB values of the input (IN) and output (OUT) signals as well as the value by which the level of the sibilant (or s-frequency) is reduced (GR). The gain reduction meter shows values between 0dB (no reduction) and -20dB (the s-frequency level is lowered by 20dB).

Positioning the DeEsser in the signal chain

When recording a voice, the de-esser's position in the signal chain is usually located after the microphone preamp and before a compressor/limiter. This keeps the compressor/limiter from unnecessarily limiting the overall signal dynamics.

Related topics

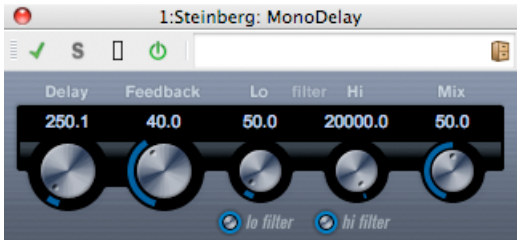
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.10 Steinberg Mono Delay



This is a mono delay effect using freely specified delay time settings.

The following parameters are available:

- **Delay**

  Sets the base note value for the delay from 0.1 to 5000 milliseconds.

- **Feedback**

  Sets the number of repeats for the delay.

- **Filter Lo**

  This filter affects the feedback loop of the effect signal and allows you to roll off low frequencies from 10Hz up to 800Hz. The button below the knob activates/deactivates the filter.

- **Filter Hi**

  This filter affects the feedback loop of the effect signal and allows you to roll off high frequencies from 20kHz down to 1.2kHz. The button below the knob activates/deactivates the filter.

- **Mix**

  Sets the level balance between the dry signal and the effect. If MonoDelay is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

Note that the delay can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the delay repeats are silenced. When the signal drops below the threshold, the delay repeats reappear.

Related topics

Audio Plug-ins

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12.11 Steinberg Distortion

Distortion will add crunch to your tracks.

The following parameters are available:

- **Boost**

  Increases the distortion amount.

- **Feedback**

  Feeds part of the output signal back to the effect input, increasing the distortion effect.

- **Tone**

  Lets you select a frequency range to which to apply the distortion effect.

- **Spatial**

  Changes the distortion characteristics of the left and right channel, thus creating a stereo effect.

- **Output**

  Raises or lowers the signal going out of the effect.

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12.12 Steinberg Dual Filter

The DualFilter effect filters out certain frequencies while allowing others to pass through.

The following parameters are available:

- **Position**

  Sets the filter cutoff frequency. If you set this to a negative value, DualFilter will act as a low-pass filter. Positive values cause DualFilter to act as a high-pass filter.

- **Resonance**

  Sets the sound characteristic of the filter. With higher values, a ringing sound is heard.

Related topics

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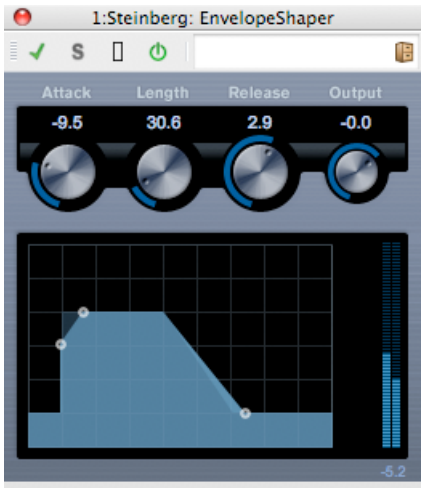
12.13 Steinberg Envelope Shaper



EnvelopeShaper can be used to cut or boost the gain of the Attack and Release phase of audio material. You can either use the knobs or drag the breakpoints in the graphical display to change parameter values. Be careful with levels when boosting the gain and if needed reduce the Output level to avoid clipping.

The following parameters are available:

- **Attack**

  (-20 to 20dB) Changes the gain of the Attack phase of the signal.

- **Length**

  (5 to 200ms) Determines the length of the Attack phase.

- **Release (-20 to 20dB)**

  Changes the gain of the Release phase of the signal.

- **Output**

  (-24 to 12dB) Sets the output level.

Related topics

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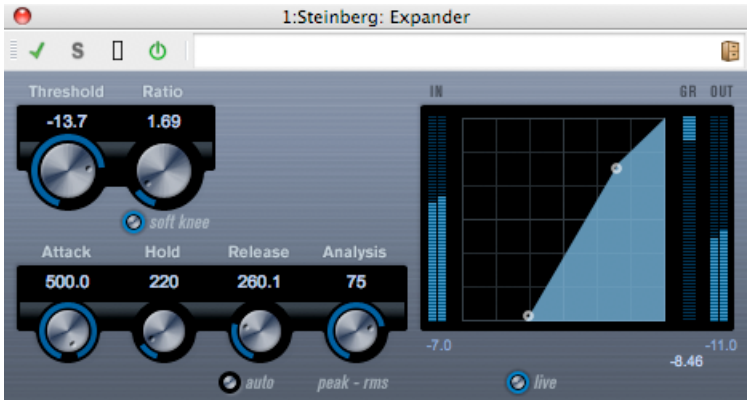
12.14 Steinberg Expander



Expander reduces the output level in relation to the input level for signals below the set threshold. This is useful when you want to enhance the dynamic range or reduce the noise in quiet passages. You can either use the knobs or drag the breakpoints in the graphical display to change the Threshold and the Ratio parameter values.

The following parameters are available:

- **Threshold (-60 to 0dB)**

  Determines the level where expansion "kicks in". Signal levels below the set threshold are affected, but signal levels above are not processed.

- **Ratio (1:1 to 8:1)**

  Determines the amount of gain boost applied to signals below the set threshold.

- **Soft Knee button**

  If this button is off, signals below the threshold are expanded instantly according to the set ratio ("hard knee"). When Soft Knee is activated, the onset of expansion is more gradual, producing a less drastic result.

- **Attack (0.1 to 100ms)**

  Determines how fast Expander responds to signals below the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.

- **Hold (0 to 2000ms)**

  Sets the time the applied expansion will affect the signal below the Threshold.

- **Release (10 to 1000ms or Auto mode)**

  Sets the amount of time it takes for the gain to return to its original level when the signal exceeds the threshold level. If the Auto button is activated, Expander will automatically find an optimal release setting that varies depending on the audio material.

- **Analysis (0 to 100) (Pure Peak to Pure RMS)**

  This parameter determines whether the input signal is analyzed according to peak or RMS values (or a mixture of both). A value of 0 is pure peak and 100 pure RMS. RMS mode operates using the average power of the audio signal as a basis, whereas Peak

mode operates more on peak levels. As a general guideline, RMS mode works better on material with few transients such as vocals, and Peak mode better for percussive material, with a lot of transient peaks.

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12.15 Steinberg Gate

Gating, or noise gating, silences audio signals below a set threshold level. As soon as the signal level exceeds the set threshold, the gate opens to let the signal through.

The following parameters are available:

- **Threshold (-60 to 0dB)**

  Determines the level where Gate is activated. Signal levels above the set threshold trigger the gate to open, and signal levels below the set threshold will close the gate.

- **State LED**

  Indicates whether the gate is open (LED lights up in green), closed (LED lights up in red) or something in between (LED lights up in yellow).

- **Filter buttons (LP, BP, and HP)**

  When the Side-Chain button (see below) is activated, you can use these buttons to set the filter type to either low-pass, band-pass, or high-pass.

- **Side-Chain button**

  This button (below the Center knob) activates the sidechain filter. The input signal can then be shaped according to set filter parameters. Internal side-chaining can be useful for tailoring how the Gate operates.

- **Center (50Hz to 20000Hz)**
  When the Side-Chain button is activated, this sets the center frequency of the filter.

- **Q-Factor (0.01 to 10000)**
  When the Side-Chain button is activated, this sets the resonance of the filter.

- **Monitor button**
  Allows you to monitor the filtered signal.

- **Attack (0.1 to 1000ms)**
  Sets the time it takes for the gate to open after being triggered. If the Live button (see below) is deactivated, it ensures that the gate will already be open when a signal above the threshold level is played back. Gate manages this by "looking ahead" in the audio material, checking for signals loud enough to pass the gate.

- **Hold (0 to 2000ms)**
  Determines how long the gate stays open after the signal drops below the threshold level.

- **Release (10 to 1000ms or Auto mode)**
  Sets the amount of time it takes for the gate to close (after the set hold time). If the Auto button is activated, Gate will find an optimal release setting, depending on the audio material.

- **Analysis (0 to 100) (Pure Peak to Pure RMS)**
  Determines whether the input signal is analyzed according to Peak or RMS values (or a mixture of both). A value of 0 is pure Peak and 100 pure RMS. RMS mode operates using the average power of the audio signal as a basis, whereas Peak mode operates more on peak levels. As a general guideline, RMS mode works better on material with few transients such as vocals, and Peak mode better for percussive material, with a lot of transient peaks.

- **Live button**
  When this button is activated, the "look ahead" feature of Gate is disengaged. Look ahead produces more accurate processing, but adds a certain amount of latency as a trade-off. When Live mode is activated, there is no latency, which might be better for "live" processing.

Note that the gate can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the gate opens.

Related topics

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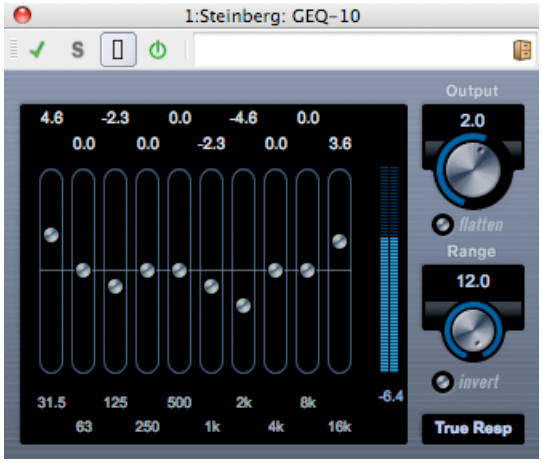
12.16 Steinberg GEQ-10/GEQ-30



(GEQ-10 pictured)

These two graphic equalizers are identical in every respect except for the number of available frequency bands (10 and 30 respectively). Each band can be cut or boosted by up to 12dB, allowing for fine control of the frequency response. In addition there are several preset modes available which can add "color" to the sound of the GEQ-10/GEQ-30.

You can draw response curves in the main display by click-dragging with the mouse. Note that you have to click on one of the sliders first before dragging across the display. You can also point and click to change individual frequency bands, or enter values numerically by clicking on a gain value at the top of the display.

At the bottom of the window the individual frequency bands are shown in Hz. At the top of the display the amount of cut/boost is shown in dB.

Apart from the frequency bands, the following parameters are available:

- **Output**

  Controls the overall gain of the equalizer.

- **Flatten button**

  Resets all the frequency bands to 0dB.

- **Range**

  Allows you to relatively adjust how much a set curve cuts or boosts the signal. If the Range parameter is turned fully clockwise, the range is +/-12dB.

- **Invert button**

  Inverts the current response curve.

- **Mode pop-up menu**

  The filter mode set here determines how the various frequency band controls interact to create the response curve, see also below.

About the filter modes

On the pop-up menu in the lower right corner there are several different EQ modes available. These modes can add color or character to the equalized output in various ways. Here follow brief descriptions of the filter modes:

- True Response – serial filters with accurate frequency response.
- Digi Standard – resonance of last band depends on sample rate.
- Variable Q – parallel filters where the resonance depends on the amount of gain. Musical sounding.
- Constant Q u – parallel filters where the resonance of the first and last bands depends on the sample rate (u=unsymmetrical).
- Constant Q s – parallel filters where the resonance is raised when boosting the gain and vice versa (s=symmetric).
- Resonant – serial filters where a gain increase of one band will lower the gain in adjacent bands.

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12.17 Steinberg Limiter

Limiter is designed to ensure that the output level never exceeds a set output level, to avoid clipping in following devices. Limiter can adjust and optimize the Release parameter automatically according to the audio material, or it can be set manually. Limiter also features separate meters for the input, output and the amount of limiting (middle meters).

The following parameters are available:

- **Input (-24 to +24dB)**
  Allows you to adjust the input gain.
- **Auto (-24 to +6dB)**
  Determines the maximum output level.
- **Release (0.1 to 1000ms or Auto mode)**
  Sets the amount of time it takes for the gain to return to its original level. If the Auto button is activated, Limiter will automatically find an optimal release setting that varies depending on the audio material.

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12.18 Steinberg Maximizer

Maximizer is used to raise the loudness of audio material without the risk of clipping. Optionally, there is a soft clip function that removes short peaks in the input signal and introduces a warm tube-like distortion to the signal.

The following parameters are available:

- **Output (-24 to +6dB)**
  Determines the maximum output level. Should normally be set to 0 (to avoid clipping).
- **Optimize (0 to 100)**
  Determines the loudness of the signal.
- **Soft Clip button**
  When this button is activated, Maximizer starts limiting (or clipping) the signal "softly", at the same time generating harmonics which add a warm, tube-like characteristic to the audio material.

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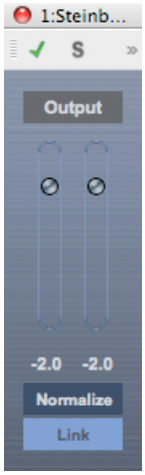
12.19 Steinberg Mix 6 to 2



Mix6To2 lets you quickly mix down your surround mix format to stereo. You can control the levels of up to six surround channels and decide for each channel up to which level it will be included in the resulting mix.

Mix6To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. The plug-in should be placed in one of the post-fader insert effect slots for the output bus.

For the Output bus the following parameters are available:

- **A Normalize button**

  If activated, the mixed output is normalized, i.e. the output level is automatically adjusted so that the loudest signal is as loud as possible without clipping.

- **A Link button**

  that links the two Output faders.

Related topics

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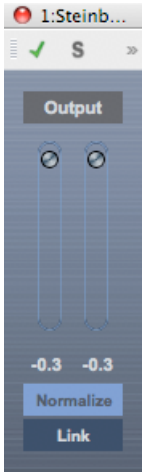
12.20 Steinberg Mix 8 to 2



Mix8To2 lets you quickly mix down your surround mix format to stereo. You can control the levels of up to eight surround channels and decide for each channel up to which level it will be included in the resulting mix.

Mix8To2 does not simulate a surround mix or add any psycho-acoustical artifacts to the resulting output – it is simply a mixer. The plug-in should be placed in one of the post-fader insert effect slots for the output bus.

For the Output bus the following parameters are available:

- **A Normalize button**

  If activated, the mixed output is normalized, i.e. the output level is automatically adjusted so that the loudest signal is as loud as possible without clipping.

- **A Link button**

  that links the two Output faders.

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12.21 Steinberg Mono to Stereo

This effect will turn a mono signal into a "pseudo-stereo" signal. The plug-in must be inserted on a stereo track playing a mono file.

The following parameters are available:

- **Width**

  Controls the width or depth of the stereo enhancement. Turn clockwise to increase the enhancement.

- **Delay**

  Increases the amount of differences between the left and right channels to further increase the stereo effect.

- **Color**

  Generates additional differences between the channels to increase the stereo effect.

- **Mono button**

  Switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when creating an artificial stereo image.

Related topics

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12.22 Steinberg Multiband Compressor

The MultibandCompressor allows a signal to be split into a maximum of four frequency bands, each with its own freely adjustable compressor characteristic. The signal is processed on the basis of the settings that you have made in the Frequency Band and Compressor sections. You can specify the level, bandwidth and compressor characteristics for each band by using the various controls.

The Frequency Band editor

The Frequency Band editor in the upper half of the panel is where you set the width of the frequency bands as well as their level after compression. Two value scales and a number of handles are available. The vertical value scale to the left shows the input gain level of each frequency band. The horizontal scale shows the available frequency range. The handles provided in the Frequency Band editor can be dragged with the mouse. You use them to set the corner frequency range and the input gain levels for each frequency bands.

- The handles at the sides are used to define the frequency range of the different frequency bands.
- By using the handles on top of each frequency band, you can cut or boost the input gain by +/- 15dB after compression, for that specific frequency band.

Bypassing frequency bands

Each frequency band can be by-passed using the "B" button in each compressor section.

Soloing frequency bands

A frequency band can be soloed using the "S" button in each compressor section. Only one band can be soloed at a time.

Using the Compressor section

By moving breakpoints or using the corresponding knobs, you can specify the Threshold and Ratio. The first breakpoint from which the line deviates from the straight diagonal will be the threshold point.

For each of the four bands the following compressor parameters are available:

- **Threshold (-60 to 0dB)**
  Determines the level where Compressor "kicks in". Signal levels above the set threshold are affected, but signal levels below are not processed.
- **Ratio (1000 to 8000) (1:1 to 8:1)**
  Determines the amount of gain reduction applied to signals over the set threshold. A ratio of 3000 (3:1) means that for every 3dB the input level increases, the output level increases by only 1dB.
- **Attack (0.1 to 100ms)**
  Determines how fast the compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) will pass through unprocessed.
- **Release (10 to 1000ms or Auto mode)**
  Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, the compressor will automatically find an optimal release setting that varies depending on the audio material.

The Output Control

The Output knob controls the total output level that the MultibandCompressor passes on to Wavelab. The range is from -24 to +24dB.

Related topics

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12.23 Steinberg Octaver

This plug-in can generate two additional voices that track the pitch of the input signal one octave and two octaves below the original pitch, respectively. Octaver is best used with monophonic signals.

The following parameters are available:

- **Direct**

  Adjusts the mix of the original signal and the generated voice(s). A value of 0 means only the generated and transposed signal is heard. By raising this value, more of the original signal is heard.

- **Octave 1**

  Adjusts the level of the generated signal one octave below the original pitch. Set to 0 means the voice is muted.

- **Octave 2**

  Adjusts the level of the generated signal two octaves below the original pitch. Set to 0 means the voice is muted.

Related topics

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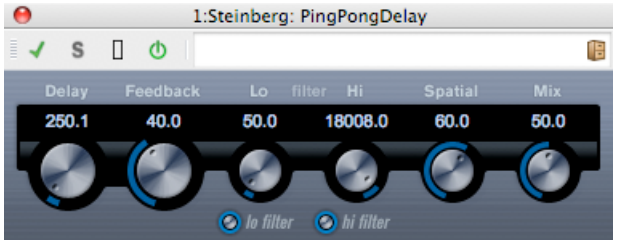
12.24 Steinberg Pingpong Delay



This is a stereo delay effect that alternates each delay repeat between the left and right channels.

The following parameters are available:

- **Delay**

  Sets the base note value for the delay from 0.1 to 5000 milliseconds.

- **Feedback**

  Sets the number of repeats for the delay.

- **Filter Lo**

  This filter affects the feedback loop and allows you to roll off low frequencies up to 800Hz. The button below the knob activates/deactivates the filter.

- **Filter Hi**

  This filter affects the feedback loop and allows you to roll off high frequencies from 20kHz down to 1.2kHz. The button below the knob activates/deactivates the filter.

- **Spatial**

  Sets the stereo width for the left/right repeats. Turn clockwise for a more pronounced stereo "ping-pong" effect.

- **Mix**

  Sets the level balance between the dry signal and the effect. If PingPongDelay is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

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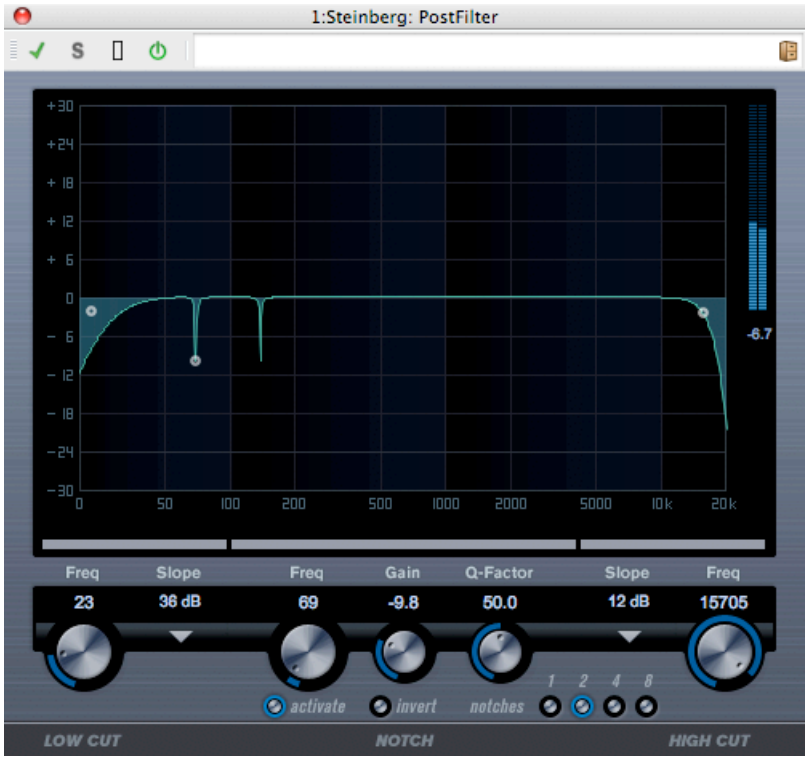
12.25 Steinberg PostFilter



The PostFilter is the filter plug-in to use if you are working on a post-production mix, but of course you can use it in music production, too, as an alternative to complex EQ configurations. It allows quick and easy filtering of unwanted frequencies, creating room for the important sounds in your mix.

The PostFilter plug-in combines a low-cut filter, a notch filter and a high-cut filter. You can either make settings by dragging the handles in the graphical display, or by adjusting one of the controls below the display section. Use the Preview buttons to compare the result of your filtering and the filtered frequencies.

The following parameters are available:

- **Level meter**

  The meter to the right of the EQ display shows the output level, giving you an indication of how the filtering affects the overall level of the edited event.

- **Low Cut Freq (20Hz to 1kHz, or Off)**

  Use this low-cut filter to eliminate low-frequency noise. The filter is off when the handle/knob is moved all the way to the left.

- **Low Cut Slope pop-up menu**

  Allows you to choose a slope value for the low-cut filter.

- **Low Cut Preview button**

  Use the Preview button (found between the Low Cut Freq button and the graphical display) to switch the filter to a complementary high-cut filter. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.

- **Notch Freq**

  Sets the frequency of the notch filter.

- **Notch Gain**

  Allows you to adjust the gain of the selected frequency. Use positive values to identify the frequencies that you want to filter out.

- **Notch Gain Invert button**

  Inverts the gain value of the notch filter. Use this button to filter out unwanted noise. While looking for the frequency to omit, it sometimes helps to boost it in the first place (set notch filter to positive gain). After you have found it, you can use the Invert button to cancel it out.

- **Notch Q-Factor**

  Sets the width of the notch filter.

- **Notch Preview button**

  Use the Preview button (found between the notch filter buttons and the graphical display) to create a bandpass filter with the peak filter's frequency and Q. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.

- **Notches buttons (1, 2, 4, 8)**

  These buttons add additional notch filters to filter out harmonics.

- **High Cut Freq (3Hz to 20kHz, or Off)**

  Use this high-cut filter to eliminate high-frequency noise. Filter is Off when the handle/-knob is moved all the way to the right.

- **High Cut Slope pop-up menu**

  Allows you to choose a slope value for the high-cut filter.

- **High Cut Preview button**

  Use the Preview button (found between the High Cut Freq button and the graphical display) to switch the filter to a complementary low-cut filter. This deactivates any other filters, allowing you to listen only to the frequencies you want to filter out.

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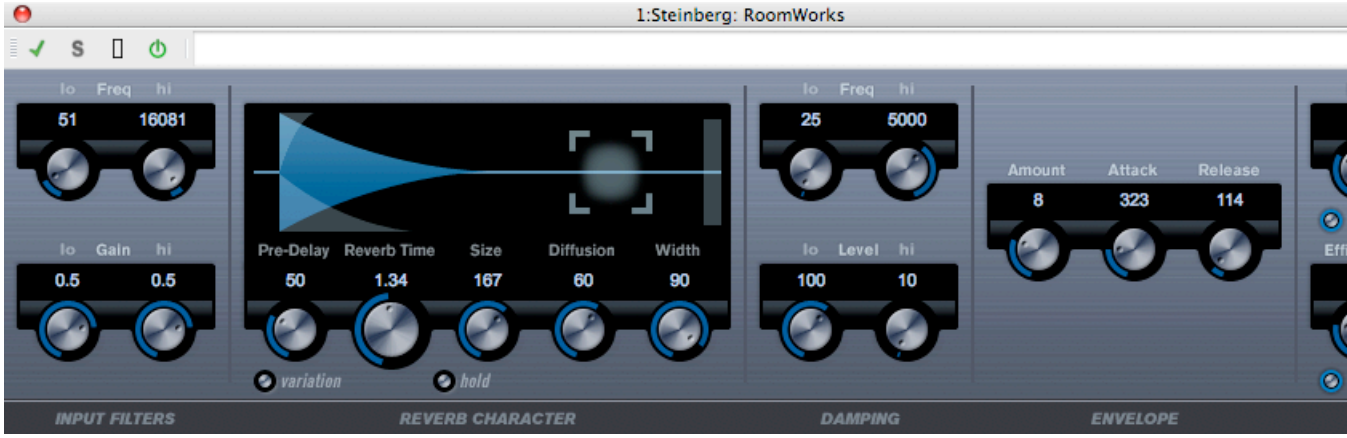
12.26 Steinberg Roomworks



RoomWorks is a highly adjustable reverb plug-in for creating realistic room ambiance and reverb effects in stereo and surround formats. The CPU usage is adjustable to fit the needs of any system. From short room reflections to cavern-sized reverb, this plug-in delivers high quality reverberation.

The following parameters are available:

Input Filters

- **Lo Freq**

  Determines the frequency at which the low-shelving filter takes effect. Both the high and low settings filter the input signal prior to reverb processing.

- **Hi Freq**

  Determines the frequency at which the high-shelving filter takes effect. Both the high and low settings filter the input signal prior to reverb processing.

- **Lo Gain**

  Controls the amount of boost or cut for the low-shelving filter.

- **Hi Gain**

  Controls the amount of boost or cut for the high-shelving filter.

Reverb Character

- **Pre-Delay**

  Controls how much time passes before the reverb is applied. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.

- **Reverb Time**

  Allows you to set the reverb time in seconds.

- **Size**

  Alters the delay times of early reflections to simulate larger or smaller spaces.

- **Diffusion**

  Affects the character of the reverb tail. Higher values lead to more diffusion and a smoother sound, while lower values lead to a clearer sound.

- **Width**

  Controls the width of the stereo image. 100% gives you full stereo reverb. At 0%, the reverb is all in mono.

- **Variation button**

  Pressing this button generates a new version of the same reverb program using altered reflection patterns. This is helpful when certain sounds are causing odd ringing or undesirable results. Creating a new variation will often solve these issues. There are 1000 possible variations.

- **Hold button**

  Pressing this button freezes the reverb buffer in an infinite loop (yellow circle around button). You can create some interesting pad sounds using this feature.

Damping

- **Lo Freq**

  Determines the frequency below which low-frequency damping will occur.

- **High Freq**

  Determines the frequency above which high-frequency damping will occur.

- **Low Level**

  Affects the decay time of low frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes low frequencies to decay quicker. Values above 100% cause low frequencies to decay more slowly than the mid-range frequencies.

- **High Level**

  Affects the decay time of high frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes high frequencies to decay quicker. Values above 100% cause high frequencies to decay more slowly than the mid-range frequencies.

Envelope

- **Amount**

  Determines how much the envelope attack and release controls affect the reverb itself. Lower values have a more subtle effect while higher values lead to a more drastic sound.

- **Attack**

  The envelope settings in RoomWorks control how the reverb will follow the dynamics of the input signal in a fashion similar to a noise gate or downward expander. Attack determines how long it takes for the reverb to reach full volume after a signal peak (in milliseconds). This is similar to a pre-delay but the reverb is ramping up instead of starting all at once.

- **Release**

  Determines how long after a signal peak the reverb can be heard before being cut off, similar to a gate's release time.

Output

- **Mix**

  Determines the balance of dry (unprocessed) and wet (processed) signal. When RoomWorks is used as an insert for an FX channel, you will most likely want to set this to 100% or use the Send button.

- **Wet only button**

  This button defeats the mix parameter, setting the effect to 100% wet or affected signal. This button should normally be pressed when RoomWorks is being used as a send effect for an FX or group channel.

- **Efficiency**

  Determines how much processing power is used for RoomWorks. The lower the value, the more CPU resources will be used, and the higher the quality of the reverb. Interesting effects can be created with very high Efficiency settings (>90%). Experiment for yourself.

- **Export button**

  Determines if during audio export RoomWorks will use the maximum CPU power for the highest quality reverb. During export you may wish to keep a higher efficiency setting to achieve a specific effect. If you want the highest quality reverb during export, make sure this button is activated.

- **Output meter**

  Indicates the level of the output signal.

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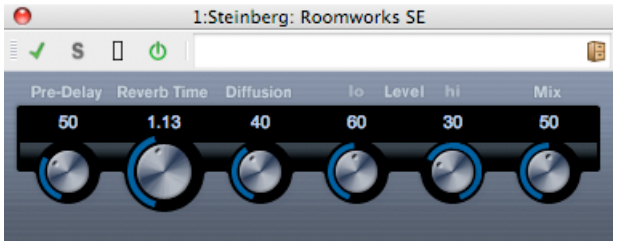
12.27 Steinberg Roomworks SE



RoomWorks SE is a "lite" version of the RoomWorks plug-in. This plug-in delivers high quality reverberation, but has fewer parameters and is less demanding of CPU than the full version.

The following parameters are available:

- **Pre-Delay**

  Controls how much time passes before the reverb is applied. This allows you to simulate larger spaces by increasing the time it takes for first reflections to reach the listener.

- **Reverb Time**

  Allows you to set the reverb time in seconds.

- **Diffusion**

  Affects the character of the reverb tail. Higher values lead to more diffusion and a smoother sound, while lower values lead to a clearer sound.

- **Hi Level**

  Affects the decay time of high frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes high frequencies to decay quicker. Values above 100% cause high frequencies to decay more slowly than the mid-range frequencies.

- **Lo Level**

  Affects the decay time of low frequencies. Normal room reverb decays quicker in the high- and low-frequency range than in the mid-range. Lowering the level percentage causes low frequencies to decay quicker. Values above 100% cause low frequencies to decay more slowly than the mid-range frequencies.

- **Mix Determines**

  the blend of dry (unprocessed) signal to wet (processed) signal. When using RoomWorks SE inserted in an FX channel, you will most likely want to set this to 100% or use the Send button.

Related topics

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Master Section

How do I add effects?

Presets

12.28 Stereo Delay

StereoDelay has two independent delay lines with freely specified delay time settings.

The following parameters are available:

- **Delay 1 & 2**
  Using these controls you specify the base note value for the delay time in milliseconds.
- **Sync button**
  The buttons below the Delay knobs are used to turn tempo sync on or off for the respective delay.
- **Feedback 1 & 2**
  The Feedback controls set the number of repeats for each delay.
- **Filter Lo 1 & 2**
  These filters affect the feedback loop and allow you to roll off low frequencies up to 800Hz. The buttons below the knobs activate/deactivate the filter.
- **Filter Hi 1 & 2**
  These filters affect the feedback loop and allow you to roll off high frequencies from 20kHz down to 1.2kHz. The buttons below the knobs activate/deactivate the filter.
- **Pan 1 & 2**
  These controls are used to set the stereo position for each delay.
- **Mix 1 & 2**
  Use these controls to set the level balance between the dry signal and the effect. If StereoDelay is used as a send effect, set them to the maximum value (100 %) as you can control the dry/effect balance with the send.

Note that the delay can also be controlled from another signal source via the side-chain input. When the side-chain signal exceeds the threshold, the delay repeats are silenced. When the signal drops below the threshold, the delay repeats reappear.

Related topics

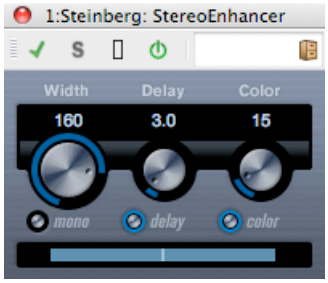
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.29 Steinberg Stereo Enhancer



This plug-in will expand the stereo width of (stereo) audio material. It cannot be used with mono files.

The following parameters are available:

- **Width**

  Controls the width of the stereo enhancement (i.e. the apparent stereo separation). Turn clockwise to increase the enhancement.

- **Delay**

  Controls the delay differences between left and right channels to further increase the stereo effect.

- **Color**

  Generates additional differences between the channels to increase the stereo enhancement.

- **Mono button**

  Switches the output to mono, to check for possible unwanted coloring of the sound which sometimes can occur when enhancing the stereo image.

Related topics

Audio Plug-ins

Master Section

How do I add effects?

Presets

12.30 Steinberg Studio Chorus

The StudioChorus plug-in is a two stage chorus effect which adds short delays to the signal and pitch modulates the delayed signals to produce a "doubling" effect. The two separate stages of chorus modulation are completely independent and are processed serially (cascaded).

For each stage the following parameters are available:

- **Rate**

  This is where you specify the base note value which may be set freely with the Rate knob.

- **Width**

  Determines the extent of the chorus effect. Higher settings produce a more pronounced effect.

- **Spatial**

  Sets the stereo width of the effect. Turn clockwise for a wider stereo effect.

- **Mix**

  Sets the level balance between the dry signal and the effect. If StudioChorus is used as a send effect, set this to the maximum value as you can control the dry/effect balance with the send.

- **Waveform Shape button**

  Allows you to select the modulation waveform, altering the character of the chorus sweep. A sine and a triangle waveform are available.

- **Delay**

  Affects the frequency range of the modulation sweep by adjusting the initial delay time.

- **Filter Lo/Hi**

  Allow you to roll off low and high frequencies of the effect signal.

Note that the modulation can also be controlled from another signal source via the side-chain input. When the sidechain signal exceeds the threshold, the modulation is controlled by the side-chain signal's envelope.

Related topics

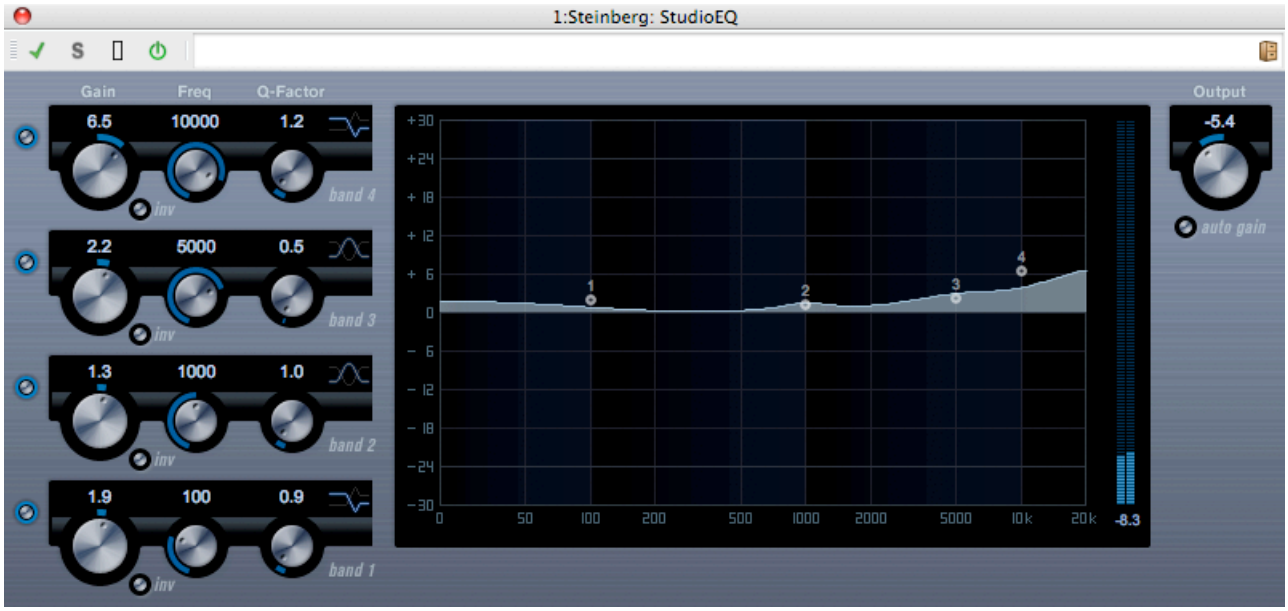
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.31 Steinberg Studio EQ



This is a high-quality 4-band parametric stereo equalizer with two fully parametric mid-range bands. The low and high bands can act as either shelving filters (three types), or Parameter Description as a Peak (band-pass) or Cut (low-pass/high-pass) filter.

Making settings

1. Click the corresponding On button on the left of the plug-in panel to activate any or all of the 4 equalizer bands (Low, Mid 1, Mid 2, and High). When a band is activated, the corresponding EQ point appears in the EQ curve display.
2. Set the parameters for an activated EQ band. This can be done in several ways:

- By using the knobs.
- By clicking on the numeric values and typing in new values.
- By using the mouse to drag points in the EQ curve display.

When using the mouse to change the parameter settings, you can use the following modifier keys: **-** When no modifier key is pressed and you drag an EQ point in the display, the Gain and Frequency parameters are adjusted simultaneously.

[Shift] Keep the **[Shift]** key pressed and drag the mouse to change the Q-factor of the corresponding EQ band.

[Alt]/[Option] Keep the **[Alt]/[Option]** key pressed and drag the mouse to change the frequency of the corresponding EQ band.

[Ctrl]/[Command] Keep the **[Ctrl]/[Command]** key pressed and drag the mouse to change the gain value of the corresponding EQ band.

The following parameters are available:

Band 1 Gain (-20 to +24dB)

Sets the amount of cut/boost for the low band.

Band 1 Inv button

Inverts the gain value of the filter. Use this button to filter out unwanted noise. While looking for the frequency to omit, it sometimes helps to boost it in the first place (set the filter to positive gain). After you have found it, you can use the Inv button to cancel it out.

Band 1 Freq (20 to 2000Hz)

Sets the frequency of the low band.

Band 1 Q-Factor (0.5 to 10)

Controls the width or resonance of the low band.

Band 1 Filter mode

For the low band, you can select between three types of shelving filters, a Peak (band-pass), and a Cut (lowpass/high-pass) filter. When Cut mode is selected, the Gain parameter is fixed. -Shelf I adds resonance in the opposite gain direction slightly above the set frequency. -Shelf II adds resonance in the gain direction at the set frequency. -Shelf III is a combination of Shelf I and II.

Band 2 Gain (-20 to +24dB)

Sets the amount of cut/boost for the mid 1 band.

Band 2 Inv button

Inverts the gain value of the filter (see the description of the Invert button for Band 1).

Band 2 Freq (20 to 20000Hz)

Sets the center frequency of the mid 1 band.

Band 2 Q-Factor (0.5 to 10)

Sets the width of the mid 1 band: the higher this value, the "narrower" the bandwidth.

Band 3 Gain (-20 to +24dB)

Sets the amount of cut/boost for the mid 2 band.

Band 3 Inv button

Inverts the gain value of the filter (see the description of the Invert button for Band 1).

Band 3 Freq (20 to 20000Hz)

Sets the center frequency of the mid 2 band.

Band 3 Q-Factor (0.5 to 10)

Sets the width of the mid 2 band: the higher this value, the "narrower" the bandwidth.

Band 4 Inv button

Inverts the gain value of the filter (see the description of the Invert button for Band 1).

Band 4 Gain (-20 to +24dB)

Sets the amount of cut/boost for the high band.

Band 4 Freq (200 to 20000Hz)

Sets the frequency of the high band.

Band 4 Q-Factor (0.5 to 10)

Controls the width or resonance of the high band.

Band 4 Filter mode

For the high band, you can select between three types of shelving filters, a Peak, and a Cut filter. When Cut mode is selected, the Gain parameter is fixed. -Shelf I adds resonance in the opposite gain direction slightly below the set frequency. -Shelf II adds resonance in the gain direction at the set frequency. -Shelf III is a combination of Shelf I and II.

Output (-24 to +24dB)

This knob on the top right of the plug-in panel allows you to adjust the overall output level.

Auto Gain button

When this button is activated, the gain is automatically adjusted, keeping the output level constant regardless of the EQ settings.

Related topics

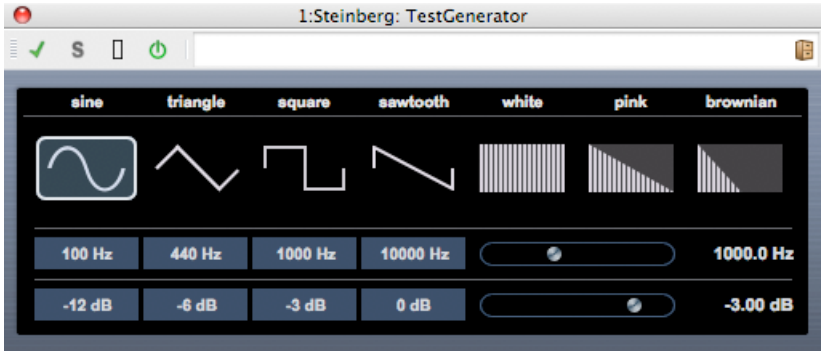
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.32 Steinberg Test Generator



This utility plug-in allows you to generate an audio signal, which can be recorded as an Audio File. The resulting file can then be used for a number of purposes:

- For testing the specifications of audio equipment.
- For measurements of various kinds, such as calibrating tape recorders.

- For testing signal processing methods.
- For educational purposes.

The TestGenerator is based on a waveform generator which can generate a number of basic waveforms such as sine and saw as well as various types of noise. Furthermore, you can set the frequency and amplitude of the generated signal. As soon as you add the TestGenerator as an effect on an audio track and activate it, a signal is generated. You can then activate recording as usual to record an Audio File according to the signal specifications:

Waveforms and noise section

Allows you to set the basis for the signal generated by the waveform generator. You can select between four basic waveforms (sine, triangle, square, and sawtooth) and three types of noise (white, pink, and brownian).

Frequency section

Allows you to set the frequency of the generated signal. You can select one of the preset values (100, 440, 1000, or 10000Hz), or use the slider to set a value between 1Hz and 20000Hz.

Gain section

Allows you to set the amplitude of the signal. The higher the value (up to 0dB), the stronger the signal. You can select one of the preset values (e.g. -20dB), or use the slider to set a value between -81 and 0dB.

Related topics

Audio Plug-ins

Master Section

How do I add effects?

Presets

12.33 UV22HR Dithering

The UV22HR is a high quality dithering plug-in, based on an advanced algorithm developed by Apogee. It allows extended information (from 32 bit floating point or 24 bit integer audio) to be encoded into a standard medium such as CD, MP3 or AAC without loss of sonic integrity. No pre-auditions or settings are necessary and audio detail is retained up to 30dB into the noise floor.

For an introduction to the concept of dithering, see Dithering.

The following parameters are available:

- **Bit Resolution**
  The UV22HR supports dithering to multiple resolutions:
- **8, 16, 20 or 24 bits**
  You select the desired resolution by clicking the corresponding button.
- **Hi**
  Try this first, it is the most "all-round" setting.
- **Lo**
  This applies a lower level of dither noise.
- **Auto black**
  When this is activated, the dither noise is gated (muted) during silent passages in the material.

Note that dithering should always be applied post-fader on an output bus.

Related topics

Audio Plug-ins

Master Section

How do I add effects?

Presets

Dithering

12.34 Steinberg Vintage Compressor

This is modeled after vintage type compressors. This compressor features separate controls for input and output gain, attack, and release. In addition, there is a Punch mode which preserves the attack phase of the signal and a program-dependent Auto feature for the Release parameter.

The available parameters work as follows:

Input (-24 to 48dB)

In combination with the Output setting, this parameter determines the compression amount. The higher the input gain setting and the lower the output gain setting, the more compression is applied.

Output (-48 to 24dB)

Sets the output gain.

Attack (0.1 to 100ms)

Determines how fast the compressor responds. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.

Punch button (On/Off)

When this is activated, the early attack phase of the signal is preserved, retaining the original "punch" in the audio material, even with short Attack settings.

Release (10 to 1000ms or Auto mode)

Sets the amount of time it takes for the gain to return to its original level. If the Auto button is activated, Vintage Compressor will automatically find an optimal release setting that varies depending on the audio material.

Related topics

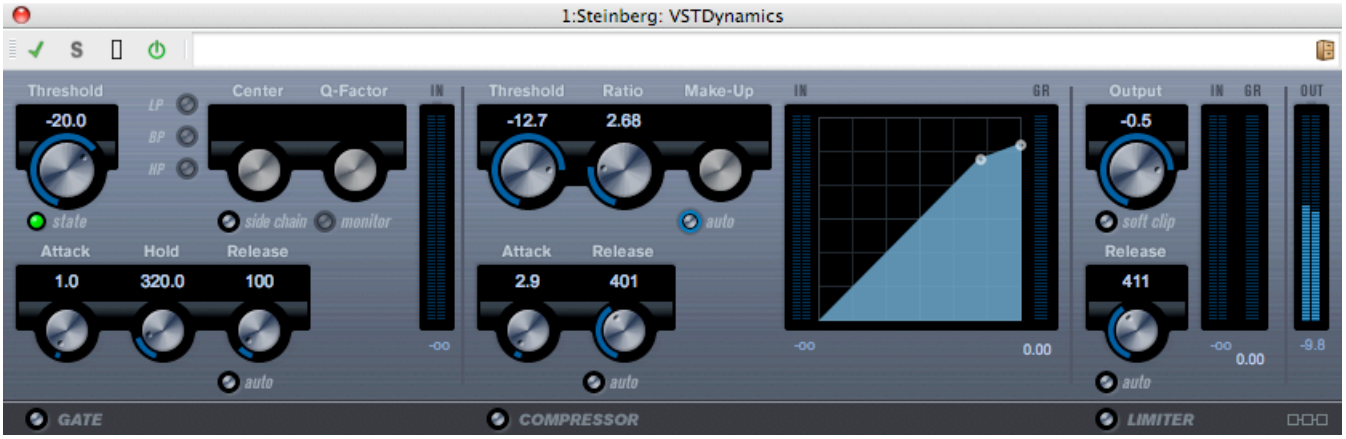
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.35 Steinberg VST Dynamics



VSTDynamics is an advanced dynamics processor. It combines three separate processors: Gate, Compressor and Limiter, covering a variety of dynamic processing functions. The window is divided into three sections, containing controls and meters for each processor.

Activating the individual processors You activate the individual processors using the buttons at the bottom of the plug-in panel.

The Gate section

Gating, or noise gating, is a method of dynamic processing that silences audio signals below a set threshold level. As soon as the signal level exceeds the set threshold, the gate opens to let the signal through. The Gate trigger input can also be filtered using an internal side-chain.

The following parameters are available:

Threshold (-60 to 0dB)

Determines the level where Gate is activated. Signal levels above the set threshold trigger the gate to open, and signal levels below the set threshold close the gate.

State LED

Indicates whether the gate is open (LED lights up in green), closed (LED lights up in red) or something in between (LED lights up in yellow).

Side-Chain button

This button activates the internal side-chain filter. You can use this to filter out parts of the signal that might otherwise trigger the gate in places you not want it to, or to boost frequencies you wish to accentuate, allowing for more control over the gate function.

LP (low-pass), BP (band-pass), HP (high-pass)

These buttons set the basic filter mode.

Center (50 to 22000Hz)

Sets the center frequency of the filter.

Q-Factor (0.001 to 10000)

Sets the resonance or width of the filter.

Monitor (On/Off)

Allows you to monitor the filtered signal.

Attack (0.1 to 100ms)

Sets the time it takes for the gate to open after being triggered.

Hold (0 to 2000ms)

Determines how long the gate stays open after the signal drops below the threshold level.

Release (10 to 1000ms or Auto mode)

Sets the amount of time it takes for the gate to close (after the set hold time). If the Auto button is activated, Gate will find an optimal release setting, depending on the audio material.

The Compressor section

The compressor reduces the dynamic range of the audio, making softer sounds louder or louder sounds softer, or both. It works like a standard compressor with separate controls for threshold, ratio, attack, release and make-up gain. The compressor features a separate display that graphically illustrates the compressor curve shaped according to the Threshold, Ratio and Make-Up Gain parameter settings. It also features Gain Reduction meters and a program-dependent Auto feature for the Release parameter.

The available parameters work as follows:

Threshold (-60 to 0dB)

Determines the level where the compressor "kicks in". Signal levels above the set threshold are affected, but signal levels below are not processed.

Ratio (1:1 to 8:1)

Determines the amount of gain reduction applied to signals above the set threshold. A ratio of 3:1 means that for every 3dB the input level increases, the output level increases by only 1dB.

Make-Up (0 to 24dB)

This parameter is used to compensate for output gain loss, caused by compression. When the Auto button is activated, gain loss is being compensated automatically.

Attack (0.1 to 100ms)

Determines how fast the compressor responds to signals above the set threshold. If the attack time is long, more of the early part of the signal (attack) passes through unprocessed.

Release (10 to 1000ms or Auto mode)

Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, the compressor will automatically find an optimal release setting that varies depending on the audio material.

Graphical display

Use the graphical display to graphically set the Threshold and Ratio values. To the left and right of the graphical display you will find two meters that show the amount of gain reduction in dB.

The Limiter section

The limiter is designed to ensure that the output level never exceeds a set threshold, to avoid clipping in following devices. Conventional limiters usually require very accurate setting up of the attack and release parameters to prevent the output level from going beyond the set threshold level. The limiter adjusts and optimizes these parameters automatically according to the audio material. You can also adjust the Release parameter manually.

The following parameters are available:

Output (-24 to +6dB)

Determines the maximum output level. Signal levels above the set threshold are affected, but signal levels below are left unaffected.

Soft Clip button

If this button is activated, the limiter acts differently. When the signal level exceeds -6dB, Soft Clip starts limiting (or clipping) the signal "softly", at the same time generating harmonics which add a warm, tube-like characteristic to the audio material.

Release (10 to 1000ms or Auto mode)

Sets the amount of time it takes for the gain to return to its original level when the signal drops below the threshold level. If the Auto button is activated, the limiter will automatically find an optimal release setting that varies depending on the audio material.

The Module Configuration button

Using the Module Configuration button in the bottom right corner of the plug-in panel, you can set the signal flow order for the three processors. Changing the order of the processors can produce different results, and the available options allow you to quickly compare what works best for a given situation. Simply click the Module Configuration button to change to a different configuration. There are three routing options:

- C-G-L (Compressor-Gate-Limit)
- G-C-L (Gate-Compressor-Limit)
- C-L-G (Compressor-Limit-Gate).

Related topics

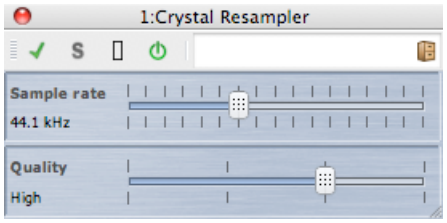
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.36 Crystal Resampler



This plug-in offers high quality sample rate conversion of Audio Files. For example, if your Audio File was recorded or saved at 96 or 48kHz and you wish to burn it to CD at 44.1kHz, this plug-in will perform the conversion ready for burning.

The following parameters are available:

- **Sample rate**

  The rate can be set in 17 steps from 6kHz to 384kHz. It includes the common sampling rates of 44.1, 48, 96 and 192kHz.

- **Quality**

  The quality requirements can be set in four steps - Preview (fast), Standard, High, or Ultra (slow).

Related topics

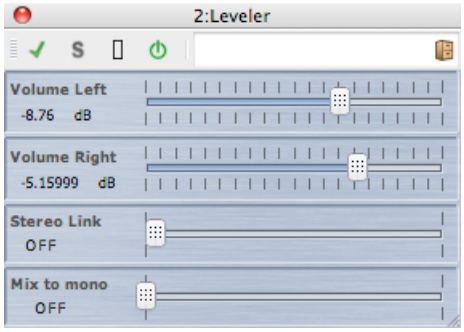
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.37 Leveler



Leveler is useful for correcting an imbalance or adjusting levels between stereo channels, or for mixing down to mono.

The parameters are as follows:

- **Volume Left, Right (-48dB to +12dB)**
  This governs how much of the signal is included in the left and/or right channel of the output bus.

- **Stereo Link (OFF or LINKED)**
  When set to LINKED, Volume Right delivers the same gain as that set for Volume Left.

- **Mix to Mono (OFF or ON)**
  When set to ON, a mono mix of the stereo channels is delivered to the output bus.

Related topics

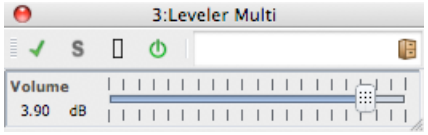
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.38 Leveler Multi



Leveler Multi takes multi-channel input and applies a fader equally to all channels.

The parameter is as follows:

- **Volume (-48dB to 12dB)**
  This governs how much gain is applied to the signal before the output bus.

Related topics

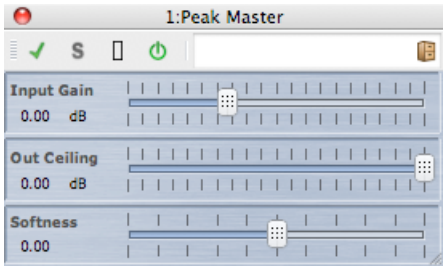
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.39 Peak Master



Peak master is a basic plug-in that minimizes peaks in your Audio File, allowing a louder mix without clipping. It is particularly useful in helping to tame very dynamic instruments.

The parameters are as follows:

- **Input Gain**
  Values range from -12dB to 24dB.

- **Out Ceiling**
  Values range from -18dB to 0dB.

- **Softness**
  Values range from -5 to +5.

Related topics

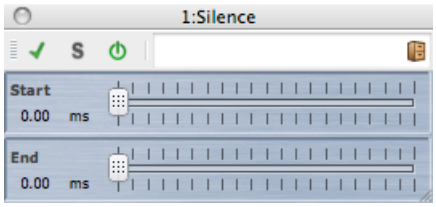
Audio Plug-ins

Master Section

How do I add effects?

Presets

12.40 Silence



The Silence built-in plug-in provides a simple way of injecting a precise period of silence into the start or end of an Audio File.

The parameters are as follows:

- **Start**

  Use the slider to add from 0 to 60,000ms of silence to the start of the file

- **End**

  Use the slider to add from 0 to 60,000ms of silence to the end of the file.

Related topics

Audio Plug-ins

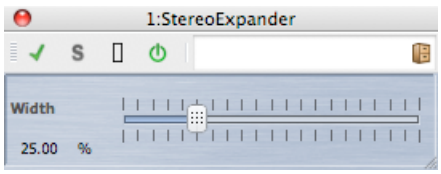
Master Section

How do I add effects?

Presets

Silence Generator

12.41 Stereo Expander



This plug-in is a stereo width enhancer that takes a stereo input signal and makes it sound "wider".

Stereo Expander gives better results from "real" stereo material (as opposed to mono channels panned to different positions in the stereo image).

Stereo Expander has the following parameter:

- **Width (0 to 100%)**

  Higher values result in a greater stereo width. Normally, you should set Width to values between 0% and 20%; higher values can be used for special effects.

Related topics

Audio Plug-ins

Master Section

How do I add effects?

Presets

12.42 Internal Dithering

Internal Dithering is a built-in plug-in that provides a simple way of adding a small amount of noise to the rendered signal to improve the apparent signal-to-noise ratio of the output.

For an introduction to the concept of dithering, see Dithering.

The following parameters are available:

- **Noise Type** Two types are available, in addition to No Noise
- **Noise Shaping** Three types are available, in addition to No Shaping
- **Bit resolution** of the quantized output: 8, 16, 20 or 24 bits are available.

Note that dithering is applied post-fader on the output bus.

Related topics

Audio Plug-ins

Master Section

How do I add effects?

Presets

Dithering

Chapter 13

Utilities

WaveLab's Utilities menu contains a number of useful tools accessible from all Workspaces. This page contains links to information about them.

CD/DVD utilities:

Import Audio CD

Burn Audio CD from DDP Image

Data CD/DVD

Batch utilities:

Batch Conversion

Batch Renaming

Utility windows:

Log window

Background task monitor

Timecode

13.1 Import Audio CD

This dialog allows you to import one or more tracks from an audio CD.

Once you have identified the source CD, you can select the track or tracks to import. You must specify a folder location and file format before the file or files can be saved to disk.

Actions tab Specify the optical disk drive from the drop-down list, and the disk reading speed. The title and track details are automatically completed. If no track names appear, click to refresh CD reading.

From this list you can select the tracks (or define the start and end points of a single track). The "Rename tracks" menu option automatically adds the track order number to the name in a range of different styles. You can select a track name for text editing by double-clicking.

In "Output", accept the default Folder location or choose another. Clicking produces the Audio File Format dialog for changing details of the Audio File's format.

Options tab This accesses a list of tickable options to refine how the tracks are read and output to file.

You can access this dialog via **Utilities** > **Import Audio CD tracks...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File Format Dialog

13.2 Burn Audio CD from DDP Image

This dialog allows you to burn an Audio CD from a DDP image. The dialog has controls for refreshing the list of attached devices, opening the CD/DVD drive and erasing a CD-RW or DVD-RW disk. You can choose to do a test burn of your CD, or write the CD from a range of write speeds.

You can access this dialog by choosing **Utilities** > **Burn Audio CD from DDP Image...**.

Alternatively you can import a DDP image in the Audio Montage Workspace via **File** > **Import** > **Audio DDP Image...** to convert it to a montage and then burn the montage.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

DDP files

Authoring CDs and DVDs

13.3 Data CD/DVD

This dialog allows you to arrange files and folders into a structure that you can later write as a "data" disk.

You can prepare a disk for a variety of media (CD-DA, DVD or Blu-ray Disk). The list of media devices displays the amount of data that will fit on each type of disk. As you add files, an indicator on the left (and a text display at the bottom) display how "full" your new disk is. You can enter a name for your disk and change the disk file structure before writing your data to an ISO image or writeable optical disk.

Note: Although you can prepare a disk in any format, the types of physical disk you can burn will be limited by the optical drive connected to your computer. If you do not have a

Blu-ray Disk writer connected, for example, you will not be able to write a Blu-ray Disk you may have prepared. However, you can still save it as an ISO Disk image and transfer it for remote burning.

To prepare a data disk, follow these steps:

Adding files

Add files and folders to the disk by:

- dragging them directly from your computer's file browser.
- dragging a file from any Workspace by either dragging it by its title bar tab or using the document button .
- choosing **File** > **Special** > **Add to Data CD/DVD** from any of the editing Workspaces to add the currently focused file.

Removing files

To remove files from the file structure:

- select the files/folders you wish to remove and click on the red cross icon. To remove a group of files, hold down **[Shift]** while selecting the files/folders to remove. Please note that this *does not* delete the files from the computer, only a reference to them.
- to remove all files/folders from the layout choose **Reset**.

Editing the file structure

To edit the arrangement of the files in your disk:

- drag files/folders to a different location in the file list pane. To move multiple files/folders hold down **[Shift]** while selecting the files/folders you wish to move.
- drag file/folders over a folder icon to move them into that directory.
- to add a new folder click on the folder icon.
- to edit the name of a file/folder, double click on its title.

Burning your disk

Once you are happy with your disk layout you can write it to a disk or disk image:

- select a media type from the list of media devices. This will determine the maximum size and type of medium you will be writing to, whether it be CD, DVD, Blu-ray Disk or an ISO image of any of these. You can change the media type at any stage.

- enter the name that you want your disk to display when it is mounted as a data volume.
- to write your disk, click write CD/DVD icon. This opens the write CD/DVD dialog which enables you to burn your data disk to an ISO image or to an optical disk using a connected CD/DVD/Blu-ray Disk writer.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Write Audio CD

Blu-ray Disk

ISO image

13.4 Batch Conversion

This dialog allows you to convert the file format of a group of Audio Files.

This could be useful for example, if you have a large number of Audio Files that you wish to encode as MP3 files, or a group of stereo files that you need to downsample and convert to mono. You can also take maximum advantage of your computer's architecture if you have a multiple processor cores, by spreading the processing load across them. These settings are selected in the Multitasking tab.

To use the Batch Converter tool:

1. Click on the plus sign icon to add files, (or just drag the files directly into the pane)
2. Define an output file location and file format
3. Choose any options or performance settings from the Options and Multi-tasking tabs
4. Click Start to begin (you can also choose an option to start the process automatically when dragging the file in).

You can access this dialog by choosing **Utilities** > **Batch conversion of Audio Files...**

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Audio File Format Dialog

13.5 Batch Renaming

This dialog appears in various locations in WaveLab and allows you to rename a batch of files, Clips or markers.

The Batch Renaming dialog offers a powerful set of facilities to convert, remove, format, replace and insert text into existing names. You can use simple options to match text, or you can build your own regular expressions, as required.

Batch renaming can be useful with large projects for example, so you can apply easily identifiable names to all referenced files, markers and Clips belonging to the project.

Batch renaming can be used for the following operations:

- to rename files
- to rename Clips in Audio Montages
- to rename markers in Audio Files and Audio Montages.

These functions share most of the same features, with some minor differences. The Batch Renaming dialog has three pages. The first page defines which files/clips/markers should be renamed; the second dialog page defines how the renaming should be executed. The third page shows you a preview of the names that will be produced, before you commit to the changes.

You can access this dialog in various places throughout WaveLab:

- from any Workspace: via **Utilities** > **Batch file renaming...**
- from the Marker window: via **Functions** > **Batch marker renaming...**
- from the Clips window in the Audio Montage Workspace: via**Functions** > **Batch clip renaming...**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Batch Conversion

Marker Window

About Regular Expressions

Regular Expressions

13.6 Log window

This shared tool window allows you to view log messages that have been issued by WaveLab.

When using WaveLab's scripting language for example, the *logWindow( )* function outputs messages to this window. There are a number of toggle buttons and these allow you to filter the types of messages displayed.

Note: When using the logWindow( ) method you need to have the "Show informal notes" button selected for the messages to be visible. If no log messages are displayed, make sure all the toggle buttons are selected - ! .

You can access this window via **Workspace** > **Shared tool windows** > **Log**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window via **Utilities** > **Log**.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Control Window

Specific tool windows

Scripting

13.7 Background task monitor

This specific tool window allows you to view all background rendering processes that are in progress.

You can adjust the priority with which they are processed, pause or cancel them. It is useful if you have a number of lengthy processes underway and want to free up some processing power to focus on editing. You can either lower the priority of a task so it does not use as much of the computer processor's capacity, or pause the task temporarily.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Background task monitor**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace. You can also access this window in the Control Window by choosing **Utilities** > **Background task monitor**.

You can choose to automatically open the Background task monitor when a rendering process begins. To enable this option, select the checkbox via **Options** > **Global Preferences** > **Options** > **Make the Background Task Monitor visible when a task starts** . Since this window is a Shared_tool_window, the location where it opens may vary.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Specific tool windows

Control Window

Rendering

13.8 Timecode

This shared tool window displays the current time of the playback head (or the cursor position if not playing back), in the timecode format currently selected in the Time Format dialog.

The current time can be in a range of standard SMPTE formats, CD or DVD specific formats, or a custom one - see Time Format dialog.

The most common use for this window is for working with video or film. You can use it to display the playback head position of the current Audio File in a SMPTE format to frame level resolution (i.e. hours:minutes:seconds:frames). You can also use it to display CD frames (minutes:seconds:frames) when you are authoring CD or DVDs.

You can access this window by choosing **Workspace** > **Shared tool windows** > **Timecode**. It is available in the Control Window, Audio File Workspace and Audio Montage Workspace.

For an explanation of each parameter and interface feature, click on , or the 'What's this?' question mark icon. For more information see Getting Help

Related topics

Time Format dialog

CD Frame

SMPTE timecode

Authoring CDs and DVDs

Chapter 14

How do I...

The following is a list of common tasks and brief instructions on how to achieve them using WaveLab.

... import songs from an audio CD into WaveLab?

... use VST Plug-ins?

... edit an existing Audio File?

... compile and edit several Audio Files on multiple tracks?

... apply processing to an Audio File?

... convert an Audio File into another audio format?

... play back with real-time effects?

... record new Audio File?

... convert the sample rate of an Audio File?

... create a Podcast?

... open an aac, m4a or m4p Audio File from Apple iTunes?

14.1 ... import songs from an audio CD into WaveLab?

From the Audio File Workspace, select **Utilities** > **Import Audio CD tracks...**

This opens the Import Audio CD dialog window where you can specify the tracks to import or the specific time ranges to import for each track. You can also specify where the output Audio Files are stored and the format in which to save them.

See Import Audio CD for more information.

14.2 ... use VST Plug-ins?

You use the Master Section to apply effects to a file (or a selection), using the built in and/or third party VST plug-ins.

See the Master Section for more details on using VST plug-ins.

The VST plug-ins are described individually in Audio Plug-ins.

14.3 ... edit an existing Audio File?

You can open Audio Files from within the Audio File Workspace by:

- opening the file directly using the **File** > **Open** command
- using WaveLab's File Browser to locate your file and open it (you may need to change the file type filter so that the file type matches the Audio File)
- dragging a file directly from the operating system's file browser

When you open the Audio File, it appears in a Wave edit window for you to begin editing. In the Audio Montage Workspace you can only drag Audio Files into an existing montage; you can't open an Audio File directly.

14.4 ... compile and edit several Audio Files on multiple tracks?

To accomplish this you need to create an Audio Montage.

First create a new Audio Montage file from **File** > **New** in the Audio Montage Workspace, or if you are working within the Audio File Workspace from **Edit** > **Create Audio Montage from Audio File...**

To add more Audio Files you can use drag them directly from:

- the File Browser
- the Operating System's file browser
- the document button in the status bar from within the Audio File Workspace. See Status Bar for more information.
- the document tab.

To add further tracks to your montage, right-click on the **[1]** button on the left side of the window and choose "Add stereo track/Add mono track".

See Audio Montages for more information.

14.5 ... apply processing to an Audio File?

The Audio File Workspace Process menu contains a number of options for manipulating Audio Files, including fade-in, fade-out and normalize. Use these functions to process a complete Audio File (or a selected region) that you have open in the main editing window.

You can also use the "Process in place" rendering function in the Master Section. This applies effects to a file (or a selection) using the built in and/or third party VST plug-ins. You can use the unlimited "Undo" function to reverse any bad editing decisions.

See Render Wave window for more information.

Power users may wish to use the facilities available in the Batch Processor Workspace. This allows complex processing to be applied to a list of Audio Files. See Batch Processing for more information.

14.6 ... convert an Audio File into another audio format?

There are two ways:

- From the Audio File Workspace you can either use the "Save as" or the "Save Special" commands in the File menu. See Save Audio File dialog and Audio File Format Dialog for more information.
- You can also use the Render function within the Master Section. See Render Wave window for more information.

Power users may wish to use the facilities available in the Batch Processor Workspace. This allows complex processing to be applied to a list of Audio Files. See Batch Processing for more information.

14.7 ... play back with real-time effects?

You can play back Audio Files with real-time effects using the Master Section.

Plug-in effects (supplied with WaveLab or available from third parties) are inserted into the audio signal path via the Master Section effects slots.

See Master Section and Audio Plug-ins for more information.

14.8 ... record new Audio File?

Use the Record button on the Transport toolbar, or select **Record** from the Transport menu.

This opens the Record dialog where you can set up a new Audio File and begin to record. Once recorded, the new file appears in a Wave window where you can edit it or add it to a Montage using **Edit** > **Create Montage from Audio File...**

See Recording Dialog for more information.

14.9 ... convert the sample rate of an Audio File?

There are two ways to do this.

- From within the Audio File Workspace choose **Process** > **Convert Sample Rate...**
- You can also convert the sample rate in real-time using a resampler plug-in such as Crystal Resampler, which is included in WaveLab. See Crystal Resampler.

See Render Wave window for more information on applying VST effects.

Power users may wish to use the facilities available in the Batch Processor Workspace. This allows complex processing to be applied to a list of Audio Files. See Batch Processing for more information.

14.10 ... create a Podcast?

To create a Podcast you use the Podcast Workspace.

First you need to create a new Podcast, there are several ways to do this.

- From within the Podcast Workspace choose **File** > **New**, or
- From the Audio File Workspace add Audio Files to a podcast by choosing **File** > **Special** > **Add to Podcast...**, or
- Select **Edit** > **Create Podcast from Audio File...**

The Podcast Workspace is where you assemble, define and publish your Podcast to the Internet.

See Podcasts for more information.

14.11 ... open an aac, m4a or m4p Audio File from Apple iTunes?

Apple OS X and iTunes uses a number of proprietary audio codecs. These can be found natively on Apple OS X but on Windows platforms you need to install QuickTime (this is normally installed with iTunes on a PC).

QuickTime provides WaveLab with the necessary codecs to open these file formats.

Other file formats that are available using QuickTime include .caf, .m4v, .qt, .dv and others.

Chapter 15

Glossary

A	B	C	D	E	F	G	H	I	J
K	L	M	N	O	P	Q	R	S	T
U	V	W	X	Y	Z				

A

ADPCM AES31 AIFF files Apple Lossless files ALAW files Aliasing Amplitude ASIO Attack Audio montage

B

Bandwidth Bit depth Blu-ray Disk

C

CD Frame CD Pre-emphasis CD Text Chorus Clip Clipping Compression Cross fade

D

DAW DC Offset DDP files Decibel (dB) Dithering DVD-A

E

ECMAScript Ensoniq Paris files Equalization

F

FFT Focused Clip Formant FTP Site

G

No entries available.

H

Headroom

I

ISO image ISRC

J

No entries available.

K

No entries available.

L

Loops Loudness

M

Markers Media Catalog Number MIDI MIDI Channels Mixing Mobile phone Audio Files MP2 files MP3 files

N

Non-destructive editing Normalize NTSC

O

Ogg Vorbis files OSQ files

P

PAL/SECAM Pan Peak level Plug-ins Pre-roll and Post-roll Preset system Processor cores

Q

Quantization

R

Raw PCM files Red Book CD-DA Regular Expressions RF64 files RMS

S

Sample rate SMPTE timecode Sound Designer II files Spectrogram Sun/Java files

T

Tempo Text/Excel files

U

ULAW files UPC/EAN

V

No entries available.

W

WAV files WAV64 files Waveform Wet/Dry WMA files

X

No entries available.

Y

No entries available.

Z

Zero crossing

15.1 ADPCM

ADPCM – Microsoft/Dialogic is an audio file format commonly used for games and telephony applications that offers a lower bit rate than linear PCM and thus requires less storage space/bandwidth.

The file extension is .vox or .adpcm

Glossary contents

15.2 AES31

The **AES31 Standard** is a non-proprietary audio file format developed by the Audio Engineering Society Standards Committee (AESSC). Its aim is to allow a project to be moved to another audio editing system with a minimum of fuss, and to reproduce the project (including Edit Decision Lists and other assets) as accurately as possible.

AES31 does not have a 2GB file-size limit and can be read by PC and Mac platforms. The standard uses a form of universal resource locator for accessing files on any platform or network, together with a sample-accurate Audio Decision List: in/out points of the component audio fies are defined in terms of hours:minutes:seconds:frames *and* sample count.

The file extension is .adl

Glossary contents

15.3 AIFF files

AIFF (Audio Interchange File Format) is the default uncompressed audio file format used on the Macintosh.

AIFF was developed by Apple Inc. from IFF (Interchange File Format) which was in turn developed by Electronic Arts. The audio data in AIFF is uncompressed PCM (Pulse Code Modulation). A compressed variant, AIFF-C or AIFC, supports a number of compression codecs. 8-, 16-, 20- or 24-bit resolutions are supported.

Typical filename extensions are .aiff, .aif, .aifc, .snd.

Glossary contents

15.4 Apple Lossless files

ALAC (Apple Lossless Audio Codec) is an audio codec developed by Apple Inc. for lossless data compression of digital audio. Also known as Apple Lossless and Apple Lossless Encoder (ALE), ALAC audio data is stored within the MP4 container and uses the mathematical technique of linear prediction to achieve a file size reduction between 60% to 40%, depending on the audio content.

The Apple Lossless Encoder is part of Quicktime.

Typical filename extensions are .m4a and .alac.

Glossary contents

15.5 ALAW files

ALAW is an audio encoding and compression technique for telephony, using 8-bit resolution. The EU telephone system uses A-law encoding for digitization.

Typical file extensions are .alaw, .vox

Glossary contents

15.6 Aliasing

Aliasing is a type of distortion caused during the analog-to-digital conversion process. If the sampling rate is not at least double the frequency of the analog signal, spurious harmonics and other artifacts may be created.

Careful filtering before the sampling stage can reduce this aliasing to a minimum.

Glossary contents

15.7 Amplitude

Amplitude is the height of a waveform above or below the zero line.

Glossary contents

15.8 ASIO

Audio Stream Input/Output (ASIO) is a computer device driver protocol for digital audio specified by Steinberg, providing a low-latency and high fidelity interface between a software application and a computer's soundcard.

Glossary contents

15.9 Attack

Attack is the initial part of the sound. An attack is said to be fast when the maximum amplitude is reached very quickly (with percussive sounds from drums, guitar or piano, for example). Orchestral strings generally have slow attack, where the volume increases more slowly.

Glossary contents

15.10 Audio montage

An Audio Montage is a compilation of audio Clips composited together. In WaveLab the Audio Montage Workspace allows you to compile and edit multiple Clips (references to Audio Files on disk) on one or more tracks.

The resulting Audio Montage can include both track and Clip-based effects, volume and pan automation, wide-ranging fade and cross fade functions.

Multichannel support makes it possible to create surround mixes that can be burned to DVD-Audio compatible disks.

Glossary contents

15.11 Bandwidth

Bandwidth is the group of frequencies (expressed in Hertz or in musical octaves) that pass through a device with a loss of less than 3dB.

Glossary contents

15.12 Bit depth

Bit depth is the number of bits used to represent a single sample. 8-, 16- and 24-bits are common bit depths. Larger bit depths represent the sample more accurately and are less noisy, but take up more space on the disk.

Bit Meter

Glossary contents

15.13 Blu-ray Disk

Blu-ray Disk (BD or Blu-ray) is an optical disk storage medium designed to replace the standard DVD format. The disk has the same physical dimensions as standard CDs and DVDs, but the standard data storage capacity is 25GB (single layered), and 50GB (dual layered disk).

The name "Blu-ray" derives from the 405 nanometer blue-violet laser used to read the disk (standard DVDs use a 650nm red laser, CDs use 780nm). The shorter wavelength allows for five or 10 times more data storage than a DVD although a recent development has pushed the storage capacity to 500GB on a single disc by using 20 layers. In addition to optical improvements, Blu-ray Disks feature improvements in data encoding that further increase their capacity.

Data CD/DVD

Glossary contents

15.14 CD Frame

In a Red Book CD-DA, the time format is commonly measured in minutes, seconds and frames (mm:ss:ff), where one "**frame**" corresponds to one sector, or 1/75th of a second of stereo sound. In editing and audio extraction, the frame is the smallest addressable time interval for an audio CD, so that track start and end positions can only be defined in steps of 1/75 second duration.

Red Book CD-DA

Glossary contents

15.15 CD Pre-emphasis

CD pre-emphasis refers to process designed to increase, within a band of frequencies, the magnitude of some (usually higher) frequencies compared to the magnitude of other (usually lower) frequencies in order to improve the overall signal-to-noise ratio. Pre-emphasis aims to minimize the adverse effects of attenuation distortion or the saturation of recording media in subsequent parts of the system. When carried out properly, pre-emphasis enables a received signal to more closely resemble the original or desired signal and produces fewer bit errors.

Pre-emphasis is commonly used in telecommunications, digital audio recording, record cutting and in FM broadcasting transmissions. The presence of pre-emphasis on a track is sometimes indicated by a tick in the column on the **Import Audio CD** dialog.

Import Audio CD

Glossary contents

15.16 CD Text

CD Text is an extension of the Red Book Compact Disc specifications standard for audio CDs. The standard allows disc and track information (album name, song name and artist, for example) to be embedded on a standards-compliant audio CD.

CD-Text information is usually stored in the lead-in area of the CD (where there is roughly 5KB of space available). Support for CD-Text is common, but not all CD players can display it.

Red Book CD-DA

Glossary contents

15.17 Chorus

Chorus is an time-based effect produced by combining the original signal with a number of moving delays and pitch shifted copies, often panned across a stereo field.

Steinberg Chorus

Glossary contents

15.18 Clip

Clip A Clip contains a reference to a source Audio File on your hard disk, as well as start and end positions in the file (allowing Clips to play back smaller sections of their source Audio Files). Any number of Clips can reference the same source file.

Note that a Clip is a reference to the original source file and does not contain audio data.

Focused Clip versus Select Clips

It is important to note that there is a difference between a "focused" Clip and a "selected" Clip. Some editing functions can only be executed on an individual Clip or "Focused Clip", while others can be executed on multiple Clips or "Selected Clips".

- *Focused Clip* - there can only ever be one or no Focused Clip at a time. The Focused Clip's name is displayed in red. It's necessary to identify the Clip as focused because there are certain functions that can only be executed on a single Clip. You use the Focused Clip window to perform these functions.
- *Selected Clip* - there can be many selected Clips. This allows you to edit multiple Clips at the same time using functions such as copy, delete, move, etc. Selected Clips have a different background color.

Glossary contents

15.19 Clipping

Clipping is a form of audio distortion that occurs when a signal is truncated because it exceeds the capabilities of the amplifier circuit. On an oscilloscope, the audio peaks appear sliced off.

To avoid clipping, reduce the system gain in or before the gain stage in which the clipping occurs.

Steinberg Limiter

Glossary contents

15.20 Compression

Compression This dynamic process is used to smooth out any large transient peaks in an audio signal that would otherwise cause distortion through clipping or otherwise overloading the system. Levels are progressively attenuated as they approach maximum levels.

Steinberg Compressor

Steinberg Multiband Compressor

Steinberg Vintage Compressor

Glossary contents

15.21 Cross fade

Cross fade is mixing two audio signals by fading one out at the same time as fading the other in.

Glossary contents

15.22 DAW

DAW A Digital Audio Workstation is a software and hardware system dedicated to recording, editing and playing back digital audio tracks on hard disk.

Because of the computational demands of audio editing and mastering, DAWs are often based on very highly specified PC or Macintosh computers, equipped with high-quality sound cards, large amounts of RAM, fast (multi-)processors and large capacity high-performance hard disks. In addition, they may also feature a noise suppressing enclosure and a passive cooling system.

Glossary contents

15.23 DC Offset

DC offset occurs when hardware (such as a sound card) adds DC current to an audio signal. The recorded waveform is not centered around the baseline (i.e. negative infinity). WaveLab can correct a DC offset, which would otherwise artificially reduce the headroom available during normalization processes.

DC Remover

Glossary contents

15.24 DDP files

DDP (Disk Description Protocol) is an industry format for submitting CD and DVDs to replication facilities.

DVDs are usually submitted in DDP 2.0 format; this is also referred to as "Plant Direct" (Sonic Solutions) and "DDPi" (Universal Music). A DDP file set for CD consists of: DDPID (Mandatory), DDPMS (Mandatory), PQ_DESCR (Optional) and one or more Image.dat file(s) (Mandatory).

CD-RWs, formatted as "Audio CDs" are really only suitable for content distribution, not disk replication or publishing. The DDP format offers the much more robust CIRC error correction

and so protects both the audio data and its associated metadata.

The DDP image is a data file, and when burned to optical media, is a data disc. As such, it has the full measure of CIRC error-correction to allow error-free copying of files on your computer and across networks (including the internet). It's also why you often need a DVD-R to contain a DDP image of a CD, because the data redundancy is more robust in data form than in streaming audio form.

Sending DDP files is "best practice" for disk replicators, who will upload the image file to their network and burn a glass master directly from the image using specialist hardware/software. CIRC error-correction will assure that the data matches the original, or it will stop the process.

WaveLab can read DDP files by choosing **File** > **Import** > **Audio DDP Image...**. Glossary contents

15.25 Decibel (dB)

Decibel (dB) is a logarithmic unit of measurement that expresses the size of a physical quantity (usually sound pressure level, power or intensity) relative to a reference level.

Decibels are often used in matters of sound because the ear perceives loudness on a logarithmic scale. The equation dB = 20 x log(V1/V2) is often used by sound engineers comparing two values. Applying a gain of 3dB doubles the amplitude of a sound, and a gain of -3dB halves the amplitude, and this is confirmed perceptually.

The decibel does not have a unit, since it expresses a ratio of two quantities with the same unit of measurement.

Glossary contents

15.26 Dithering

Dithering is the technique of adding small quantities of noise to a signal to reduce the audibility of low level distortion in a digital recording. A small amount of random noise is added to the analog signal before the sampling stage, reducing the effect of "quantization errors".

Note that dithering should always be applied after the output bus fader stage.

UV22HR Dithering

Quantization

Glossary contents

15.27 DVD-A

DVD-Audio (DVD-A) is a digital format for delivering high-fidelity audio content on a DVD. Audio on a DVD-A disk can be stored in many different bit-depth, sampling rate and channel combinations - from 16-, 20- or 24-bit depth, at 44.1 to 96kHz sampling rates, in mono, stereo and various surround channel combinations, including 5.1 channel surround. The sampling rate may be as high as 192kHz for stereo channels, and different bit-depth/sampling

rate and channel combinations can be used on a single disk.

DVD-Audio

Glossary contents

15.28 ECMAScript

WaveLab's scripting language is based on the **ECMAScript** scripting language, as defined in "Standard ECMA-262 - ECMAScript Language Specification".

Microsoft's JScript, Netscape's JavaScript and Adobe's Actionscript are also based on the freely-available ECMAScript standard.

ECMAScript Reference

Glossary contents

15.29 Ensoniq Paris files

Ensoniq Paris is an Audio File format used by the old Ensoniq Paris™ system. It has 16-bit and 24-bit resolution.

The file extension is .paf.

Glossary contents

15.30 Equalization

Equalization (EQ) is a process by which certain frequency bands in an Audio File are increased or lowered in level to compensate for recording or playback inadequacies.

Steinberg GEQ-10/GEQ-30

Glossary contents

15.31 FFT

FFT (Fast Fourier Transform) analysis is a mathematical method to convert a waveform from the Time Domain to the Frequency Domain.

FFT uses various mathematical shortcuts to carry out a spectral analysis, trading computational speed for limitations in the number of samples and frequency bands used in the analysis.

Glossary contents

15.32 Focused Clip

There can be no more than one Focused Clip. See Clip

Glossary contents

15.33 Formant

Formants are the distinguishing frequency components of a sound.

Glossary contents

15.34 FTP Site

File Transfer Protocol (FTP) is a network protocol used to copy a file from one host to another over a TCP/IP-based network such as the Internet. An **FTP site** may require user-based password authentication or it may allow anonymous user access. Podcasts are published to FTP sites.

Publish menu

FTP site

Glossary contents

15.35 Headroom

Headroom is the amount by which the signal-handling capabilities of an audio system exceeds a designated level - the Permitted Maximum Level (PML). Headroom can be thought of as a safety zone allowing transient audio peaks to exceed the PML without exceeding the signal capabilities of an audio system.

DC Remover

Glossary contents

15.36 ISO image

An **ISO image** is an archive file or "disk image" of an optical disk in a format defined by the International Organization for Standardization (ISO).

"ISO image" derives from the ISO 9660 file system used with CD-ROM media. ISO image files are supported by WaveLab and typically have a file extension of .iso.

Glossary contents

15.37 ISRC

ISRC - the "International Standard Recording Code" - is a code for identifying sound recordings (and music video recordings) on CDs intended for commercial distribution. WaveLab allows you to specify an ISRC code for each audio track.

The code contains the following elements:

- Country Code (2 ASCII characters)
- Registrant Code (typically, a record label - 3 ASCII characters or digits)
- Recording Year (2 digits or ASCII characters)
- Serial Number (unique number identifying the recording - 5 digits or ASCII characters).

The groups of characters are often presented with hyphens to make them easier to read, but hyphens are not part of the code.

Authoring CDs and DVDs

Burn Audio CD from DDP Image

UPC/EAN

Import Audio CD

Glossary contents

15.38 Loops

Loops are short audio Clips that create a repeating beat or pattern.

Loop Tone Uniformizer

Loop Tweaker

Glossary contents

15.39 Loudness

Loudness is a subjective measure, the perceptual correlate of amplitude of sound. There are objective measures of sound strength, such as SPL (sound pressure level in dB), but Loudness is affected by other factors, principally frequency (the sensitivity of the human ear changes as a function of frequency) and duration.

The perception of Loudness varies from one person to another, so it cannot be measured with a single metric.

Loudness distribution

Loudness normalizer

Glossary contents

15.40 Markers

Markers are symbols identifying a reference points in an Audio File, like tab markers in word processing. There are several marker types in WaveLab, which may be used for quick navigation or more specialist uses. Some markers must be used in pairs.

Markers

Marker types

Glossary contents

15.41 Media Catalog Number

The **MCN (Media Catalog Number)** is a 13-digit code for an optical disk intended for commercial distribution - there is one such code per disk. See UPC/EAN.

You can read the Media Catalog Number (EAN) from a CD by choosing **Utilities** > **Import Audio CD tracks...** > **Functions** > **CD Info...**.

Import Audio CD Glossary contents

15.42 MIDI

MIDI (Musical Instrument Digital Interface) is an industry standard that allows devices such as synthesizers and computers to communicate with each other through a system of channels and triggers.

Glossary contents

15.43 MIDI Channels

MIDI provides 16 channels for sending data. When processing MIDI triggers, WaveLab needs to know which **MIDI channel** to monitor to receive a trigger.

Glossary contents

15.44 Mixing

Mixing is the process of blending multiple sound files into a single file, with control over the relative levels of each incoming file,

Glossary contents

15.45 Mobile phone Audio Files

3GP and 3G2 file formats are multimedia container formats designed to minimize storage and bandwidth requirements for use with mobile phones.

- **3GP** (3GPP file format) is defined by the Third Generation Partnership Project (3GPP) for GSM-based (UMTS) mobile phones.
- **3G2** (3GPP2 file format) is a multimedia container format defined by the 3GPP2 for CDMA2000-based mobile phones.

Tyoical file extensions are .3gp, .3g2

Glossary contents

15.46 MP2 files

MP2 (MPEG-1, audio layer 2) is an audio encoding format defined by ISO/IEC 11172-3, alongside MPEG-1 and MPEG-3, using lossy audio compression. The three audio "layers" (MP1, MP2 and MP3) are different perceptual encoding techniques.

MP2 remains an important format for broadcast audio and is part of DAB digital radio and DVB digital television standards. It is also the audio format used in HDV camcorders. MP2 files are sometimes referred to "Musicam files".

Typical filename extensions are .mp2, .mpa, .mpg, .mus, .m32, .m44 and .m48.

MP3 files

MP2 encoding

Glossary contents

15.47 MP3 files

MP3 (MPEG-1, audio layer 3) is a patented digital audio encoding format using lossy audio compression, designed by the Moving Picture Experts Group as part of its MPEG-1 standard, and derived from MP2. MP3 is popular for PC and internet applications - data compression ratios of 10:1 are typical.

Note that when you open an MPEG compressed file in WaveLab, the file is converted to a temporary wave file that is much larger than the original compressed file, so make sure that you have enough free space on your hard disk when opening MPEG compressed files. On saving, the temporary wave file is converted back to MP3, so from a user perspective the file handling is transparent apart from the size difference mentioned above.

The filename extension is .mp3

MP2 files

Audio File Format Dialog

Glossary contents

15.48 Non-destructive editing

Non-destructive editing When you delete or change a part of an Audio File in a non-destructive system, the audio is not deleted or permanently changed - instead a set of pointers keeps track of all the edits, so these can be readily reversed. WaveLab provides comprehensive facilities for **Non-destructive editing**.

Glossary contents

15.49 Normalize

To **Normalize** is to raise the volume so that the sample with the highest level is set to the maximum user-defined level. This ensures that you are using all of the dynamic range available to you.

Steinberg Limiter

Steinberg Maximizer

Glossary contents

15.50 NTSC

NTSC (from the US "National Television System Committee") is the analog television system used in North America, parts of South America, Japan and some Pacific territories.

NTSC has 29.97 interlaced frames of video per second - a field refresh frequency of 59.94Hz (actually 60Hz/1.001). Each frame consists of a total of 525 scanlines, 486 of which are visible.

The digital terrestrial television (DTT) replacement for NTSC is ATSC ("Advanced Television Standards Committee").

PAL/SECAM

Glossary contents

15.51 Ogg Vorbis files

Ogg Vorbis is a lossy compressed audio format and codec that produces files about the same size as MP3 files but with objectively higher fidelity and no patent restrictions.

Vorbis is intended for sample rates from 8kHz telephony to 192kHz digital masters and a range of channel representations that includes monaural, stereo, surround 5.1, ambisonic, or up to 255 discrete channels. Vorbis is open source (free) software maintained by the Xiph.Org Foundation.

The file extension is .ogg.

Audio File Format Dialog

Glossary contents

15.52 OSQ files

Original Sound Quality (OSQ) is WaveLab's proprietary lossless compressed audio format. By saving files in this format, you can save considerable disk space without compromising audio quality.

The filename extension is .osq

Glossary contents

15.53 PAL/SECAM

PAL is short for "Phase Alternate Line"; **SECAM** is short for Séquentiel couleur à mémoire ("Sequential Color with Memory"). They are related analog television encoding systems used

in broadcast television systems in large parts of the world. NTSC is a another analog system used in North America and parts of South America.

In digital applications PAL video mode is usually referred to as "576i" (i.e. a vertical resolution of 576 scan lines, interlaced), in analogue contexts it is often quoted as "625 lines". SECAM Digital video uses its own colour space, so (in the digital domain) the minor colour space differences that used to exist between PAL and SECAM are no longer present.

In digital video applications such as DVDs and digital broadcasting, PAL/SECAM video is based on 576 frame lines with two interlaced video fields per frame (i.e. 50 fields per second, yielding 25 frames per second). Audio is pulse-code modulated.

NTSC

Glossary contents

15.54 Pan

Pan is short for panoramic. Pan is perceptually "placing" a mono or stereo sound source between two or more speakers. We sense stereo position from a difference in loudness when the sound strikes each ear, taking into account the time delay, spectrum, reverberation and other cues.

Steinberg Autopan

Pan Normalizer

Glossary contents

15.55 Peak level

Peak level is a measure of maximum sound level. Peak Level Meters should go as high as possible without ever reaching 0dB! The Peak Level meters display the peak levels of each channel, graphically and numerically.

By default, the meter segments and numerical peak values are displayed in green for low levels, yellow for levels between −6dB and −2dB, and red for levels above −2dB.

If the level is set too low, noise levels will be high relative to the recorded sound; if peak level exceeds 0dB by a significant amount, clipping effects can occur.

Peak Master

Glossary contents

15.56 Plug-ins

A vast number of specialist audio software **Plug-ins** are available, ranging from freeware to expensive high-end commercial products, and these can greatly extend the functionality of WaveLab.

Steinberg created Virtual Studio Technology (VST) to allow effect plug-ins to be integrated with audio editors (such as WaveLab). VST uses Digital Signal Processing (DSP) to emulate in software the effects of familiar recording studio hardware.

Audio Plug-ins

Glossary contents

15.57 Pre-roll and Post-roll

Pre-roll is the time elapsed before an event occurs; **Post-roll** is the time after an event. Pre-roll can give a brief context if you are auditioning a Clip.

Glossary contents

15.58 Preset system

WaveLab provides a sophisticated **Preset system**, which allows you to store a range of individual or bulk settings and recall them for later use.

Presets

Glossary contents

15.59 Processor cores

Processor cores Modern computers are often designed with a multi-core processor, with two or more "cores" (i.e. independent processors) in a single physical package. A dual-core processor contains two cores, and a quad-core processor contains four cores, with the aim of improving computational performance.

The performance gains of multi-core processors depend on whether the software has been implemented to take advantage of this architecture.In WaveLab, the multi-core approach allows you to simultaneously encode as many files as there are CPU cores, for example, leading to significant performance increases. Batch Conversion

Glossary contents

15.60 Quantization

Quantization is the process by which measurements are rounded to discrete values. In digital audio, the continuously varying voltages of an analog signal are quantized to discrete digital values, represented in binary numbers. In 8-bit audio, sample values range from -128 to 127; in 24-bit audio, sample values range from -8288608 to 8288607.

In analog-to-digital conversion and compression, differences between the actual analog value and the quantized digital value is called "quantization error". This error is due either to rounding or truncation and can give rise to "quantization noise" which may be reduced by using "dithering" techniques.

Dithering

Glossary contents

15.61 Raw PCM files

Raw PCM files is an Audio File format that includes no information about bit resolution or sample rate. If you open a file in this format, WaveLab prompts you to specify the bit resolution and sample rate. If this is not done correctly, the file will not play back as intended.

Note that the "$$$" file type is WaveLab's own temporary file format. If you experience a computer crash you can try opening any stray "$$$" files on your hard disk(s) to restore them.

Tyical file extensions are .raw, .bin, .pcm, .$$$

Glossary contents

15.62 Red Book CD-DA

Red Book is the standard for audio CDs (Compact Disc Digital Audio, or CD-DA). Other "Rainbow Books" in the series, known by the color of their covers, define the formats of CD-ROMs, VCD, SACD and other compact disc formats. DVD and Blu-ray formats are not defined in a Rainbow Book.

The first edition of the Red Book was released in 1980 by Philips and Sony and ratified as IEC 60908. The standard is not freely circulated - from 2009 the IEC 60908 document has been available as a PDF download for US$260. The Red Book specifies a compliant CD-DA in great detail, including the physical properties of the CD, the digital audio encoding (2-channel, signed 16-bit Linear Pulse Code Modulation, sampled at 44.1kHz), the modulation system, the CIRC error-correction scheme and subcode channels. Data are stored in sectors of 2352 bytes ("frames"), read at 75 frames/s. The bit rate is 1411.2kbits/s. Other considerations are that the maximum number of tracks is 99, the minimum duration of a track is 4 seconds (including a 2-second pause), the maximum playing time is 79.8 minutes, and the ISRC should be included.

WaveLab provides Red Book-compliant CD mastering.

CD Frame

ISRC

Glossary contents

15.63 Regular Expressions

A **Regular Expression** (regex for short) is a formula composed of characters that have a special meaning (called "operators"). A simple example is in using "*.doc" to search for all text documents in a file manager. In this case, the asterisk (star) character is an operator that matches any string before the ".doc" in a filename. In addition to WaveLab, many software applications, text editors (like EditPad Pro) and programming languages support regular expressions.

The characters ". [ ] ( ) { } ? * + - | ! / ^ $" are operators in WaveLab's regular expression engine. You can submit regular expressions in a WaveLab text box wherever you see the 💡 icon. Clicking on the light bulb icon gives a fly-out menu listing special characters and short cuts, to help you build a precise and well-formed expression.

There are many helpful web-based tutorials that explore the topic of regex.

About Regular Expressions

Glossary contents

15.64 RF64 files

RF64 is an audio file format. You have an option (set in **Options** > **Audio File editing preferences** > **Files** tab) to support the RF64 file format. If this is activated, the standard Wave file format switches automatically to the RF64 file format as soon as the file size exceeds 2GB, without any performance loss or interruption. This is especially useful when recording very long sessions as there is no need to worry about file size limit (apart from disk space availability).

An RF64 file uses the ".wav" extension but it can only be opened with an application that supports the RF64 standard.

WAV files

Glossary contents

15.65 RMS

RMS (Root Mean Square) can be used as a measurement of intensity of sound (or AC voltage, etc.) over a period of time. The RMS measurement calculates a practical average value of all the points on a waveform as a single number over a variable period that could be 1 ms, 20 ms, or 1 hour.

RMS is often a good basis for determining an "equal level" when equalizing Clip volumes, producing more natural results than using Peak levels, since RMS offers a more realistic measure of the perceived "loudness" of a sound.

Glossary contents

15.66 Sample rate

The **Sample Rate**, or sampling frequency, is the number of samples per second used to store a sound. High sample rates are better able to approximate the original continuous waveform, but require more storage space.

Glossary contents

15.67 SMPTE timecode

The Society of Motion Picture and Television Engineers devised a code to synchronize time between devices, the **SMPTE time code**. The time code is composed of hours:minutes:seconds:frames.

Frames are fractions of a second based on the frame rate (which can take the values 24, 25, 29.97 and 30 frames per second).

Timecode

Time Format dialog

Glossary contents

15.68 Sound Designer II files

Sound Designer II is an audio file format used by Digidesign applications (such as Pro Tools). 8-, 16- or 24-bit resolutions are supported.

The filename extension is .sd2

Glossary contents

15.69 Spectrogram

A **Spectrogram** is an image that shows how the spectral density of a signal varies over time.

The Spectrum display

Glossary contents

15.70 Sun/Java files

Sun/Java is an Audio File format used on Sun and NeXT computers. Files in this format can often be found on the Internet. 8-, 16- or 24-bit resolutions are supported.

Typical file extensions are .snd, .au

Glossary contents

15.71 Tempo

Tempo is the rhythmic rate of a musical piece, usually given in BPM (beats per minute).

Glossary contents

15.72 Text/Excel files

Text/Excel is a text representation of a waveform. By saving an Audio File as a text file and then opening it in a spreadsheet application such as Excel, you can view it in textual, decimal form and edit the sample values. When a text file representing a waveform is opened in WaveLab it is decoded and opened as an Audio File.

Note that these files are not compressed in any way, so they can get very large! Therefore, avoid creating and opening extremely large .txt files. Also note that when using 32 bit float files, the .txt format is not entirely lossless because it is not possible to express a binary floating point value in textual decimal form without some loss of precision.

The file extension is .txt

Audio File Format Dialog

Glossary contents

15.73 ULAW files

ULAW is an audio encoding and compression technique supported by Windows and Web phones, using 8-bit resolution. The U.S. telephone system uses U-law encoding for digitization.

Typical file extensions are .ulaw, .vox

Glossary contents

15.74 UPC/EAN

UPC/EAN code - the "Universal Product Code/European Article Number", a catalog number for an item (such as a CD) intended for commercial distribution. On a CD, the code is also called the "Media Catalog Number" - there is one such code per disk.

UPC is a 12-digit barcode widely used in the United States and Canada; EAN-13 is a 13-digit barcoding standard (12 + a checksum digit) defined by the GS1 standards organization. "EAN" is now renamed as "International Article Number", but the abbreviation has been retained.

You can read the Media Catalog Number (EAN) from a CD by choosing **Utilities** > **Import Audio CD tracks...** > **Functions** > **CD Info...**.

Authoring CDs and DVDs

Import Audio CD

Glossary contents

15.75 WAV files

WAV (Waveform Audio File Format) is an audio file format. Although a WAV file can hold compressed audio, the most common WAV format contains uncompressed audio in the linear pulse code modulation (LPCM) format.

Developed by Microsoft and IBM, WAV is closely related to AIFF and is compatible with Windows, Macintosh and Linux operating systems. Uncompressed WAV files are quite bulky, but they have a simple file structure which can be processed efficiently in audio editing and produce archive files of high quality. 8-, 16-, 20-, 24-bit and 32-bit (float) resolutions are supported.

The file extension is .wav

WAV64 files

Audio File Format Dialog

Glossary contents

15.76 WAV64 files

Wave 64 is an audio file format developed by Sony that is very similar to the standard Wave format, but with an important difference. Standard Wave files are limited in size to 2GB (stereo), but Wave 64 allows virtually any length (actually the limit is 512GB).

The file extension is .w64.

WAV files

Glossary contents

15.77 Waveform

A **Waveform** is the visual representation of wave-like phenomena (such as sound or light). The amplitude or sound pressure over time usually results in a smooth waveform.

In WaveLab, a waveform is generally represented with time on the horizontal (x-)axis, with the start of the waveform being the leftmost point. The vertical (y-)axis is the amplitude of the wave.

Glossary contents

15.78 Wet/Dry

Wet/Dry is a term that describes the balance between the effect sound (wet) and the original, unprocessed audio signal (dry). in a "dry" mix, the original signal is more pronounced; with a "wet" mix, the effect sound is more dominant.

Glossary contents

15.79 WMA files

WMA (Windows Media Audio) is a proprietary audio data compression technology developed by Microsoft. WMA can refer to an Audio File format or a range of audio codecs. Like AAC (Advanced Audio Coding) and Ogg Vorbis, WMA is intended to address perceived deficiencies in the MP3 standard.

It is also possible to import/export audio in WMA surround format, provided that you have Windows Media Player 9 (or later) installed on your system.

Typical file extensions are .wma, .asf

Audio File Format Dialog

Glossary contents

15.80 Zero crossing

A **Zero crossing** is a point where the wave crosses the zero level axis, the point where the wave is considered to have "zero level".

If you cut out a portion of a wave and paste it in somewhere else, chances are there will be a discontinuity where the two waves are joined. This discontinuity will result in a transient in the wave when it is played back, which is perceived as a "click" or "bump" in the sound. To avoid this you need to make any editing splices at a zero crossing point.

Glossary contents