VERSION 11

SONY
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Sound Forge™ Pro

User Manual

What's new in version 11?

- Improved recording workflow. *For more information, see Recording on page 143.*
- Added a Loudness Meters tool and loudness logging. *For more information, see Loudness Meters on page 140.*

  The Statistics dialog now includes loudness data. *For more information, see Viewing selection statistics on page 68.*

  Added **Use True Peaks** and **Enable DC Blocking Filter** controls to the Detect Clipping dialog for measuring loudness. *For more information, see Detecting and marking clipping on page 120.*
- Improved support for metadata in Broadcast Wave Format file. *For more information, see Broadcast Wave window (Ctrl+Alt+M, 4) on page 55.*
- Added support for editing files in SpectraLayers Pro 2.0. *For more information, see Editing with SpectraLayers Pro on page 305.*
- Improved Plug-In Chain window now allows floating plug-in windows. *For more information, see Using the Plug-In Chain on page 209.*
- Improved selection dragging: you no longer need to drag up before dragging a selection.
- Fade in and fade out curves now default to a linear curve in the processing and Mix dialogs.
- Added **Remember last-used Save As folder** to the General tab in the Preferences dialog. *For more information, see General tab on page 331.*
- Added plain text file option for saving and opening a file's regions list and playlist/cutlist. *For more information, see Saving a regions/playlist file on page 126 and Importing a regions/playlist file on page 126.*
- You can now rearrange maximized data window tabs by dragging the tabs to a new location.
- Added automatic resampling during playback for unsupported sample rates. *For more information, see Editing file properties on page 105.*
- Added support for splitting events at region boundaries. *For more information, see Splitting events at region boundaries on page 174.*
- Added support for moving markers, regions, and envelope points with events. The **Options** > **Paste Markers/Regions** command is now **Options** > **Lock to Selection** > **Markers/Regions** and **Options** > **Lock to Selection** > **Envelope Points**.
- Added support for ripple editing in event-editing mode. Choose **Options** > **Event** > **Auto Ripple** to toggle automatic ripple editing for downstream events. *For more information, see Auto ripple events on page 175.*
- Improved audio playback and recording device routing in the Preferences > Audio tab. *For more information, see Audio tab on page 342.*

Welcome

After Sound Forge® Pro software is installed and you start it for the first time, the registration wizard is displayed. This wizard offers easy steps that allow you to register the software online with Sony Creative Software Inc. Alternatively, you can register online at *http://www.sonycreativesoftware.com/reg/software* at any time.

Registration assistance

Registration assistance is available online at *http://www.sonycreativesoftware.com/reg/faq* or by fax at (608) 250-1745.

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For a detailed list of customer service options, we encourage you to visit *http://www.sonycreativesoftware.com/support/custserv.asp*. Use the following numbers for telephone support during normal weekday business hours:

Telephone/Fax	Country
(800) 577-6642 (toll-free)	US, Canada, and Virgin Islands
(608) 203-7620	for all other countries
(608) 250-1745 (Fax)	All countries

Technical support

For a detailed list of technical support options, we encourage you to visit *http://www.sonycreativesoftware.com/support/default.asp*. To listen to your support options, please call (608) 256-5555.

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Legal notices

FLAC/Ogg File Formats

Gracenote

LPEC

Microsoft DirectX programming interface

Microsoft Windows Media 9

PNG file format

Steinberg Media Technologies AG.

Tagged Image File Format (TIFF)

Targa file format

Thomson Fraunhofer MP3

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Chapter 1

Introduction

Introducing Sound Forge Pro software

Thank you for purchasing Sound Forge® Pro software and for your continued support of the Sony Creative Software Inc. family of products. The software provides you with the powerful features you have come to expect, as well as a number of new features designed to make digital audio editing quick and easy.

Sample files

Throughout the manual, you will find references to six sample audio files. The manual directs you to use these files as you experiment with different Sound Forge features. These files are installed in the same folder as the application:

- Drumhit.pca
- Fill.pca
- Loop.pca
- Musicbed.pca
- Saxriff.pca
- Voiceover.pca

The files are saved in Perfect Clarity Audio® (PCA) format, a Sony Creative Software Inc. proprietary lossless audio compression format.

Technical support

The Web site at *http://www.sonycreativesoftware.com/support/default.asp* has technical support, reference information, program updates, tips and tricks, user forums, and a knowledge base.

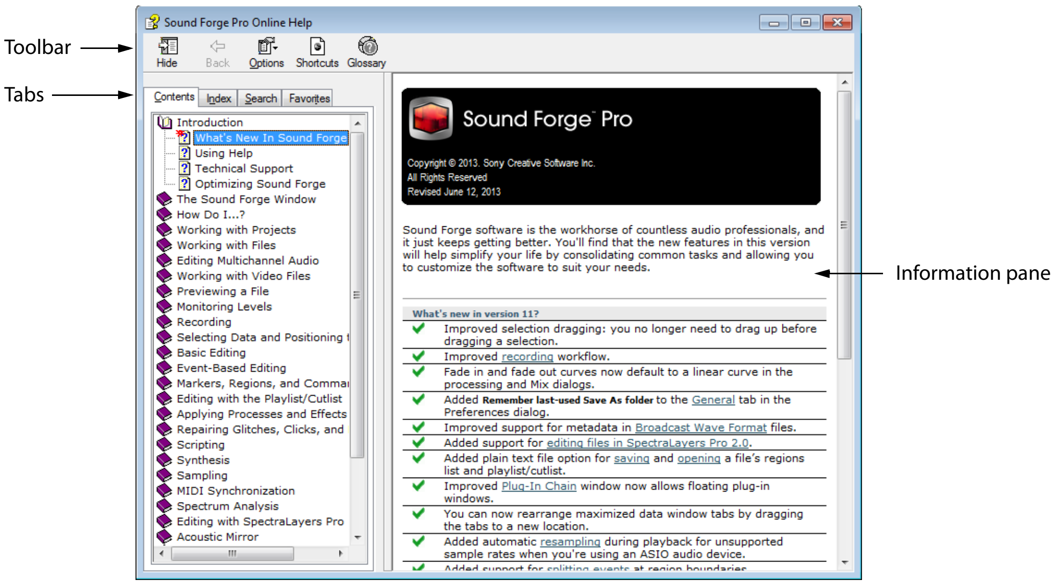
Getting help

You can access two varieties of help within Sound Forge:

- Online help
- Interactive tutorials

Online help

To access online help, choose **Contents and Index** from the **Help** menu or press F1.



The online help window has four tabs that you can use to find the information you need.

Tab	Description
Contents	Provides a list of available help topics. Click a closed book ( ) to open the pages, and then click on a topic page ( ).
Index	Provides a complete listing of the help topics available. Scroll through the list of available topics or type a word in the **Type in the keyword to find** box to quickly locate topics related to that word. Select the topic and click the **Display** button to view it.
Search	Allows you to enter a keyword and display all of the topics in the online help that contain the keyword you have entered. Type a keyword in the **Type in the word(s) to search for** box and click the **List Topics** button. Select the topic from the list and click the **Display** button to view it.
Favorites	Allows you to keep topics that you revisit often in a separate folder. To add a topic to your favorites, click the **Add** button on the **Favorites** tab.

Interactive tutorials

You can learn more about many of the features in Sound Forge by using the interactive tutorials installed with the software.

You can access the tutorials at any time by choosing **Interactive Tutorials** from the **Help** menu.

Help on the Web

Additional Sound Forge information is available on the Sony Creative Software Inc. Web site. From the **Help** menu, choose **Sony on the Web**, and choose the desired location from the submenu. The software starts your system's Web browser and attempts to connect to the appropriate page on the Sony Web site.

Chapter 2

Learning the Sound Forge Pro Workspace

This chapter provides a detailed overview of Sound Forge® Pro toolbars and controls.

Using the mouse

The following table defines the mouse-related terms used throughout this manual.

Mouse Term	Description
Pointing	Moving the mouse pointer over an item.
Clicking	Pointing to an item and quickly pressing and releasing the left mouse button. If there is no left or right specification, left-clicking is implied.
Right-clicking	Pointing to an item and quickly pressing and releasing the right mouse button. Right-clicking is frequently used to display shortcut menus.
Double-clicking	Identical to clicking, but instead of pressing and releasing the mouse button once, it is done twice in quick succession. Double-clicking always indicates the left mouse button.
Triple-clicking	Identical to clicking, but instead of pressing and releasing the mouse button once, it is done three times in quick succession. Triple-clicking always indicates the left mouse button.
Toggle-clicking	Clicking the right mouse button while holding down the left mouse button. This is used to toggle options and is a shortcut for drag-and-drop editing and using the Magnify tool.
Shift-clicking	Holding down the Shift key while clicking the mouse. Shift-clicking is typically used to skip dialogs and quickly repeat operations.
Ctrl-clicking	Holding down the Ctrl key while clicking the mouse. Ctrl-clicking is used to modify the operation of a normal click.
Dragging	Holding down the left mouse button while moving the mouse pointer and releasing the mouse at the desired location. Dragging is used to quickly move sections of data between windows, as well as to adjust sliders, scrollbars, and faders.
Slow-dragging	Holding down the right and left mouse buttons while adjusting sliders and faders increases the resolution of the movement. This is useful when making fractional adjustments to parameters.

Tip: *After you are familiar with Sound Forge basics, you may want to use mouse and keyboard shortcuts. For more information, see Shortcuts on page 349.*

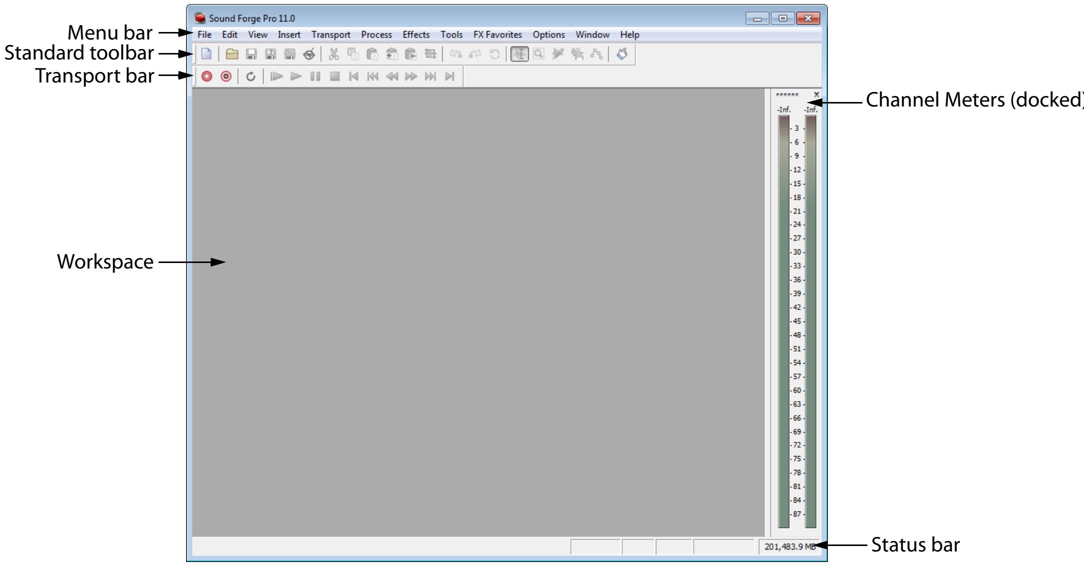
Using the mouse wheel

The following table describes the available mouse wheel functionality you can use to navigate audio files.

Mouse Functionality	Description
Wheel Up	Zoom in horizontally
Wheel Down	Zoom out horizontally
Ctrl+Wheel Up	Zoom in vertically
Ctrl+Wheel Down	Zoom out vertically
Shift+Wheel Up	Scroll left (in tenths of screen width)
Shift+Wheel Down	Scroll right (in tenths of screen width)
Ctrl+Shift+Wheel Up	Move cursor left or move current selection point left (if there is a selection)
Ctrl+Shift+Wheel Down	Move cursor right or move current selection point right (if there is a selection)

The main window

When you start the application, the main window is displayed. The main window's workspace is where you perform all audio editing.



Main window components

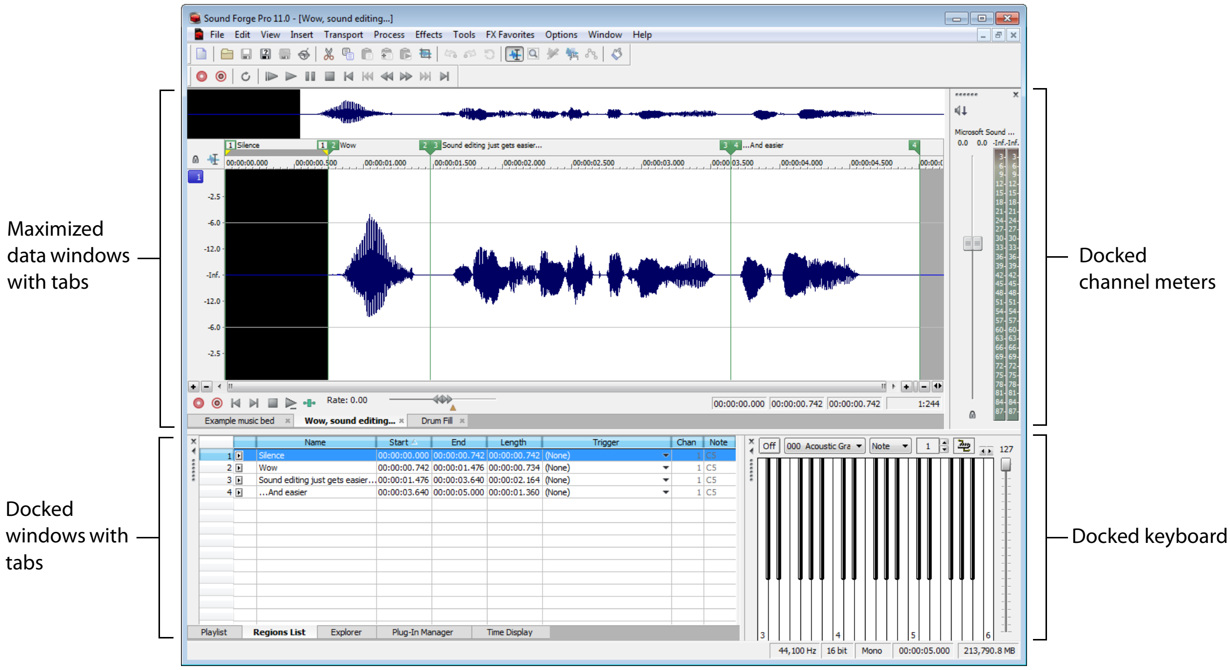
The following table describes the major components of the main window.

Component	Description
Menu bar	Displays the menu headings for the available functions.
Standard toolbar	Provides quick access to some of the most common tasks in the application. *For more information, see Standard toolbar on page 40.*
Transport bar	Provides quick access to basic audio transport functions. *For more information, see Transport toolbar on page 41.*
Status bar	Help and processing information is displayed on the left side. The boxes on the right side display the playback sample rate, bit depth, channel configuration, length of the active data window, CD time remaining, and total free storage space. **Note:** *The* **CD Time Remaining** *box is displayed only when CD tracks exist in the active data window.* With the exceptions of the **CD Time Remaining** and **Free Storage** boxes, you can edit these boxes by double-clicking or right-clicking them. When no data windows are open, only the **Free Storage** box contains a value. *For more information, see Editing file properties on page 105.*
Workspace	This is the area located behind the data windows. Audio selections dragged to the workspace automatically become new data windows. Windows such as the Regions List and Playlist can be docked along the edges of the workspace or in floating window docks.
Channel Meters	Displays the level of the output audio signal. These meters can be toggled on/off by choosing **Channel Meters** from the **View** menu. Right-clicking the channel meters displays a shortcut menu that allows you to precisely configure the appearance of the meters.

Floating and docking windows

Your workspace can become cluttered quickly if you have several windows and toolbars visible.

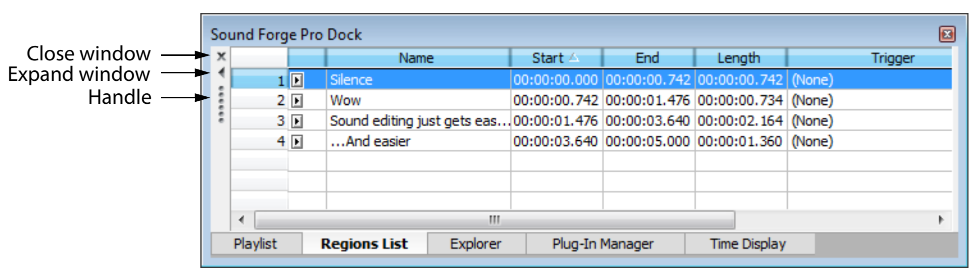
You can create multiple window docks to organize your Sound Forge windows. These docks can be anchored to the perimeter of the Sound Forge workspace, or they can float over the workspace or on a secondary monitor.



- To dock a window, drag it to a floating dock or to any edge of the Sound Forge workspace.
- To undock a window, click the handle and drag it out of the docking area or floating dock.
- To prevent a window from docking when you drag it, hold the Ctrl key.

> **Note:** *When the* **Allow floating windows to dock** *check box on the* **General** *tab of the Preferences dialog is cleared, windows will not dock unless you hold the Ctrl key. When the check box is selected, you can prevent a window from docking by holding the Ctrl key. For more information, see General tab on page 331.*

- To expand a docked window so it fills the docking area, click the **Maximize** button ( ). Click again to restore the window to its previous size.
- To remove a window from the docking area or a floating dock, click the **Close** button ( ).



You can dock several windows in the same area of the screen, and the windows will be layered. Click a window's tab to bring it to the top.

Hiding the window docking areas

You can double-click the separator between the workspace and window docking area to hide the docking area. You can also use the following shortcut keys to manage the workspace:

Note: *These shortcuts do not apply to floating docks.*

Shortcut key	Description
F11	Show/hide windows docked at bottom of workspace.
Shift+F11	Show/hide windows docked on left/right sides of workspace.
Ctrl+F11	Show/hide all docked windows.

Explorer window (Alt+1)

The Explorer window is used to find, preview, and open media files. From the **View** menu, choose **Explorer** to show or hide the Explorer window. *For more information, see Using the Explorer window on page 63.*

File Properties window (Alt+2)

The File Properties window is used to view or edit information saved in the active file. From the **View** menu, choose **File Properties** to show or hide the File Properties window. *For more information, see Editing file properties on page 105.*

Video Preview window (Alt+3)

The Video Preview window shows the video frame at the current cursor or play position. From the **View** menu, choose **Video Preview** to show or hide the Video Preview window. *For more information, see Previewing files with video on page 290.*

Time Display window (Alt+4)

The Time Display window displays the current cursor or play position. From the **View** menu, choose **Time Display** to show or hide the Time Display window. *For more information, see Customizing the Time Display window on page 329.*

Channel Meters window (Alt+5)

Sound Forge software provides peak and VU/PPM (peak program) meters that you can use to monitor your audio levels. From the **View** menu, choose **Channel Meters** to show or hide the channel meters. *For more information, see Channel Meters on page 135.*

Loudness Meters window (Alt+6)

The Loudness Meters tool provides data about an audio file's momentary loudness, short-term loudness, integrated (overall) loudness, and loudness range. You can use these values when mastering for broadcast to ensure compliance with loudness standards (such as the CALM Act). *For more information, see Loudness Meters on page 140.*

Hardware Meters window (Alt+7)

The Hardware Meters window allows you to monitor hardware outputs and adjust preview levels. From the **View** menu, choose **Hardware Meters** to show or hide the Hardware Meters window. *For more information, see Using the hardware meters on page 117.*

Undo/Redo History window (Alt+8)

The Undo/Redo History window allows you to see all of your edit operations. From the **View** menu, choose **Undo/Redo History** to show or hide the Undo/Redo History window. *For more information, see Using the Undo/Redo History window on page 85.*

Spectrum Analysis window (Alt+9)

The Spectrum Analysis window allows you to examine the fundamental frequency and overtones present in a recording. From the **View** menu, choose **Spectrum Analysis** to show or hide the Spectrum Analysis window. *For more information, see* *Using Spectrum Analysis on page 295*.

Plug-In Chain window (Ctrl+Alt+0)

The Plug-In Chain window allows you to link up to 32 DirectX and VST plug-ins into a single processing chain. From the **View** menu, choose **Plug-In Chain** to show or hide the Plug-In Chain window. *For more information, see* *Using the Plug-In Chain on page 209*.

Plug-In Manager window (Ctrl+Alt+1)

The Plug-In Manager window displays your plug-ins in a tree view like Windows Explorer. From the **View** menu, choose **Plug-In Manager** to show or hide the Plug-In Manager. *For more information, see* *Using the Plug-In Manager on page 206*.

Keyboard window (Ctrl+Alt+2)

The Keyboard window allows you to control internal or external synthesizers and samplers from Sound Forge software. From the **View** menu, choose **Keyboard** to show or hide the Keyboard window. *For more information, see* *Using the MIDI keyboard on page 267*.

Script Editor window (Ctrl+Alt+3)

The Script Editor window can be used to open, create, edit or run scripts. From the **View** menu, choose **Script Editor** to show or hide the Script Editor window. *For more information, see* *Using the Script Editor window on page 249*.

Loop Tuner window (Ctrl+Alt+4)

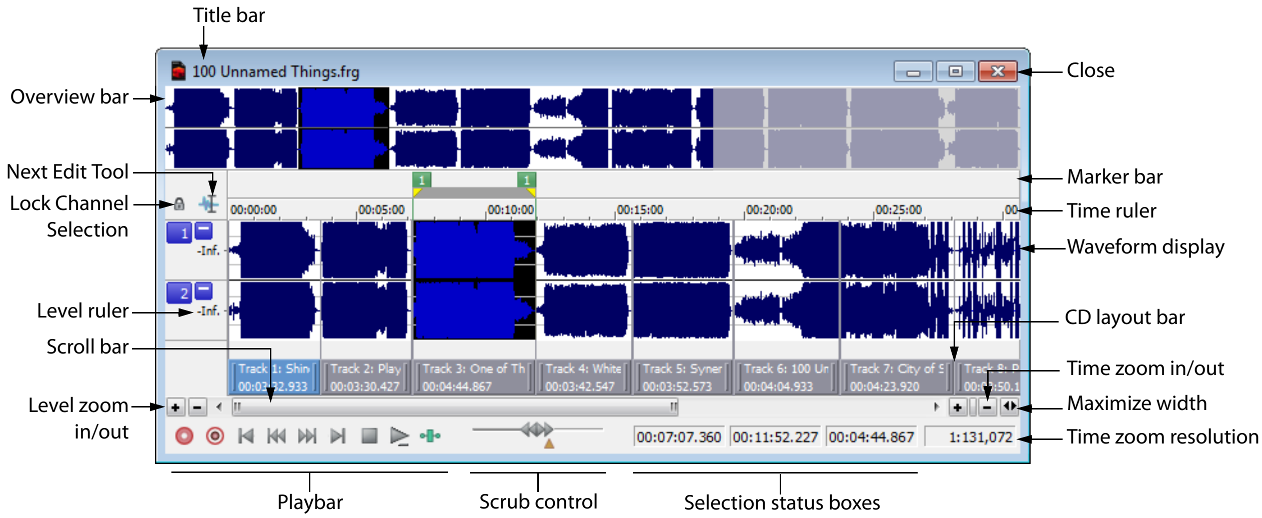
The Loop Tuner window can be used to adjust the starting and ending points of a loop to create smooth transitions. From the **View** menu, choose **Loop Tuner** to show or hide the Loop Tuner window. *For more information, see* *Editing loops on page 279*.

Record Options window (Ctrl+Alt+5)

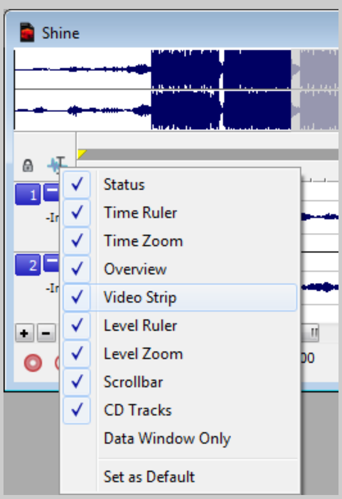
You can use the Record Options window to configure various options for recording in Sound Forge Pro. *For more information, see* *Recording options on page 150*.

Data windows

Each sound file is opened in a data window. Each data window shows you a graphical representation of the waveform and other information about the file.



Tip: *You can quickly toggle the display of portions of a data window (decorations) by right-clicking the* **Next Edit Tool** *button ( ) and choosing a command from the shortcut menu or choosing* **Options** > **Data Window**:



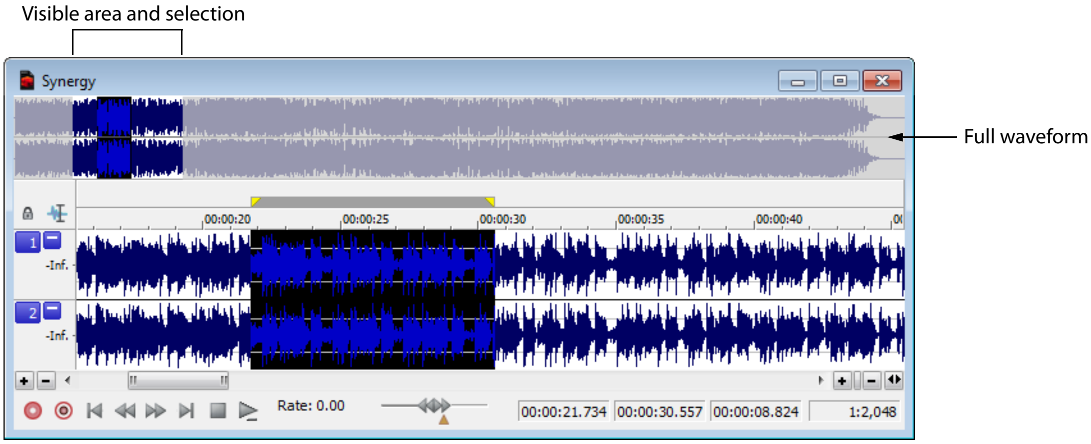
Title bar

Displays the sound file's title. If no title is specified on the Summary Information window, the file name will be displayed.

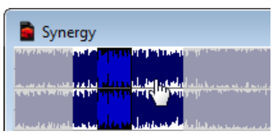
Double-click to maximize and restore the window.

Overview bar

Allows for quick navigation and playback of any part of the file:

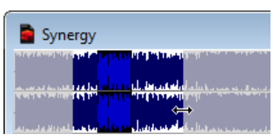


- The full waveform is displayed in the overview bar.
- The unshaded portion of the waveform display represents the portion of the waveform shown in the data window. You can drag this portion to navigate the waveform.
- The current selection is also represented in the overview bar.
- Click in the overview bar to move the cursor.
- Double-click to center the cursor in the waveform display.
- Right-click in the overview bar to toggle playback of the file from the cursor position in the data window.
- To navigate the waveform, you can drag the unshaded portion of the waveform display:



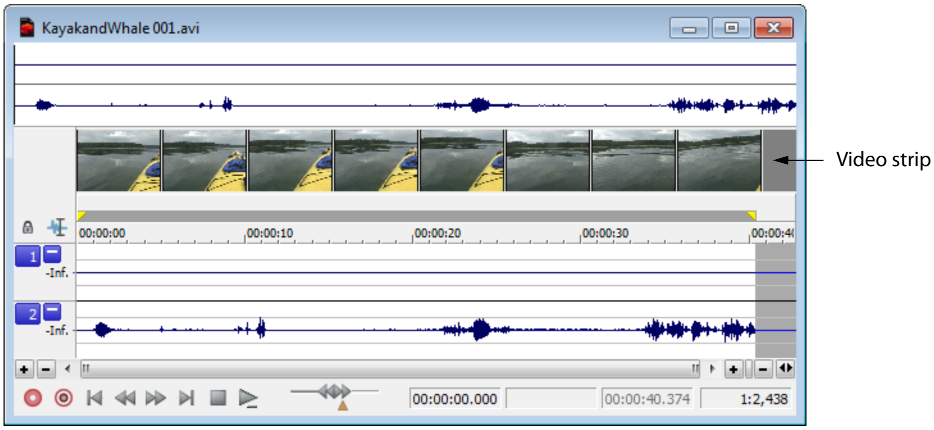
Tip: *Hold Ctrl and drag the unshaded portion of the waveform display to scrub with the audio event locator. For more information, see* *Scrubbing with the audio event locator on page 93*.

- To zoom horizontally, you can drag the ends of the unshaded portion of the waveform display:



Video strip

When you open a file that contains a video stream, Sound Forge displays a video strip above the audio waveform to help you navigate the file. *For more information, see Working with Video on page 289.*



Time ruler

Shows the current location in the data window as well as ruler tags.

- Right-click to display the time ruler shortcut menu.
- Drag to scroll the data window.

Next Edit Tool button

Click to toggle through the Edit, Magnify, Pencil, Event, and Envelope tools.

> **Note:** *The Pencil tool is available only at magnification levels below the* **Pencil tool maximum zoom ratio** *setting on the Editing tab in the Preferences dialog.* *For more information, see Editing tab on page 335.*

Minimize channel height

Click the **Minimize** button ( ) to reduce the height of individual channels, or click the **Restore** button ( ) to restore their height.

Hold Shift while clicking a **Minimize** button to minimize all channels except for the one you clicked.

Level ruler

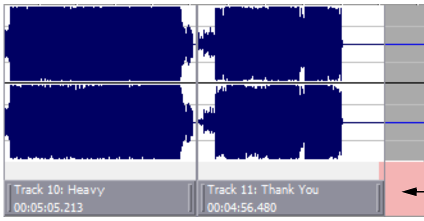
Shows the amplitude of the waveform.

- Right-click to display the level ruler shortcut menu, which allows you to change the zoom level and labels.
- Drag to shift the view up or down when zoomed in vertically.

CD layout bar

The CD layout bar displays information about the tracks you've created for a disc-at-once CD. Each CD track shows the track's number and length.

Red indicators are drawn at the right end of the CD layout bar to represent the end of the disc (if the disc length is known).



End-of-disc indicator

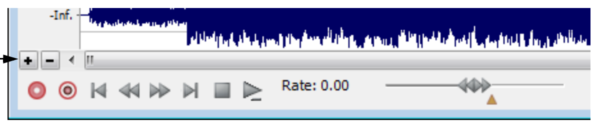
You can use the CD layout bar to perform many of the track-editing functions from the Track List window.

For more information, see Moving tracks on the CD layout bar on page 315.

Level zoom

To zoom in and out vertically by small increments, click the **Level Zoom In/Out** buttons, or click and drag the area between the buttons to zoom quickly.

Level Zoom In/
Out buttons



For more information, see Zooming and magnifying on page 93.

Playbar

Use the playbar transport buttons to control playback:

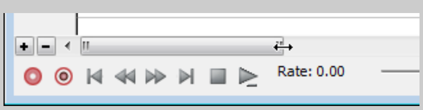
Button	Description
Arm	Opens the wave device and loads all recording buffers in order to minimize the amount of time between clicking the **Record** button ( ) and when recording starts. When **Create new window** is selected in the **Mode** drop-down list in the Record Options window, the **Arm** and **Record** buttons are enabled even when no data windows are open. When **Normal** or **Create regions** is selected, the **Arm** and **Record** buttons are not available until you create a data window or open a file. *For more information, see Recording options on page 150.*
Record	Starts and stops recording. For more information, please see one of the following sections: • *Creating a new recording on page 143* • *Recording into an existing sound file on page 146* • *Recording audio automatically on page 148*
Go to Start	Moves the cursor to the start of the file.
Go to Previous Track	Moves the cursor to the previous disc-at-once track or index. Hold Ctrl while clicking to skip index markers, or hold Shift to extend a selection. **Tip:** *This button is displayed only if disc-at-once tracks are present in your data window.*
Go to Next Track	Moves the cursor to the next disc-at-once track or index. Hold Ctrl while clicking to skip index markers, or hold Shift to extend a selection. **Tip:** *This button is displayed only if disc-at-once tracks are present in your data window.*
Go to End	Moves the cursor to the end of the file.

Button	Description
Stop	Stops playback and returns the cursor to its position prior to playback.
Play Normal	Plays the file in Normal mode. • If there is no selection, playback occurs from the cursor to end of file. • If there is a selection, playback occurs from the beginning of the selection to the end of the selection. • Effects from the Plug-In Chain are previewed in real time when you play back the file. • To bypass the plug-in chain, bypass the chain in the Plug-In Chain window. *For more information, see Previewing the effects chain on page 212.*
Play as Sample	Click to set playback to Sample mode. When you click the **Play** button ( ) while in Sample mode, playback will adhere to the following rules: • If the file contains loops, the loops will repeat as many times as specified on the Edit Sample dialog. Use this to listen to a sound file as it would sound when played by a sampler. • If the file does not contain any loops, the file will be played once from beginning to end. **Note:** *This button is not displayed unless a sample loop has been defined in your file.*
Play as Cutlist	Click to begin playback from the cursor position, skipping any cutlist regions. **Note:** *This button is not displayed unless you have playlist/cutlist regions defined and have set your playlist/cutlist to Cutlist mode. To use Cutlist mode, choose* **Playlist/Cutlist** *from the Edit menu, and then choose* **Treat as Cutlist** *from the submenu.*

Scroll bar

Within the scroll bar, the box represents the portion of the waveform shown in the waveform display. Drag to scroll the sound file forward and backward in time to see parts of the file not currently visible in the waveform display.

Tip: *You can drag the end of the scroll box to zoom in or out:*



Scrub control

Drag the scrub control ( ) at the bottom of a data window to shuttle forward or backward from the cursor position to find an edit point.

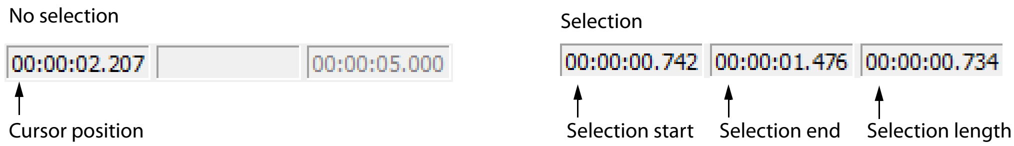
Tip: *Hover over the scrub control and roll the mouse wheel forward or backward.*

You can drag the Normal Rate indicator ( ) below the scrub control to adjust playback speed (or double-click the label to type a playback rate).

For more information, see Scrubbing on page 92.

Selection status bar

Shows the beginning, end, and length of a selection. If no selection has been made, only the cursor position is displayed:



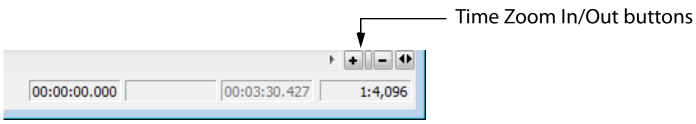
- Double-click the **Selection Start** box to edit the current value. Press Tab or Enter to move the cursor to the new position.
- Double-click the **Selection Start** or **Selection Length** box to edit beginning or ending of the selection. Press Tab or Enter to update the selection.

> **Tip:** *To update the* **Selection Start**, **Selection End**, *or* **Selection Length** *values quickly, you can type + or - and a numeric value. For example, to extend the right end of a selection one second, double-click the* **Selection End** *box and type +1. To move the left end of a selection one minute to the left, type -1:00.*

- Right-click to display the Status Format shortcut menu, which allows you to choose a time format.

Time zoom

To zoom in and out horizontally by small increments, click the **Time Zoom In/Out** buttons, or drag the area between the buttons to zoom quickly.



For more information, see Zooming and magnifying on page 93.

Time zoom resolution

Indicates the number of samples of data represented by each point on the screen horizontally. This determines the length of time shown in the waveform display. With a small resolution value (1:1, 1:2, 1:4, ...), a shorter length of time is displayed.

For more information, see Zooming and magnifying on page 93.

Maximize width

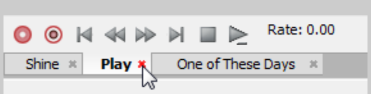
Click to stretch the width of the data window to fit within the Sound Forge workspace.

> **Tip:** *Press Ctrl+Enter.*

Close data window

Click the **Close** button to close a data window.

> **Tip:** *If you have maximized your data windows, click the* **Close** *button in a data window tab to close that data window.*



Arranging data windows

You can use the commands on the Window menu to arrange data windows in the Sound Forge workspace.

Tip: *Press Ctrl+Tab to switch forward through the open windows, or press Ctrl+Shift+Tab to switch backward through the open windows.*

Command	Description
New Window	Creates a new data window.
Cascade	Arranges all open data windows so they overlap with the title bar of each window remaining visible.
Tile Horizontally	Arranges all open data windows top to bottom with no overlapping. This command affects only non-minimized windows.
Tile Vertically	Arranges all open data windows left to right with no overlapping. This command affects only non-minimized windows.
Arrange Icons	Arranges minimized data windows at the bottom of the workspace.
Maximize All	Maximizes all open data windows. Tabs for each data window also appear if the **Tabs for maximized data windows** setting is set to **Top** or **Bottom**. *For more information, see Display tab on page 333.*
Minimize All	Minimizes all open data windows.
Restore All	Restores all minimized windows to their previous window size and position.
Close All	Closes all open data windows.
Window List	Displays a list of all open data windows. Choose a window from the menu to switch focus to that data window.

Toolbars

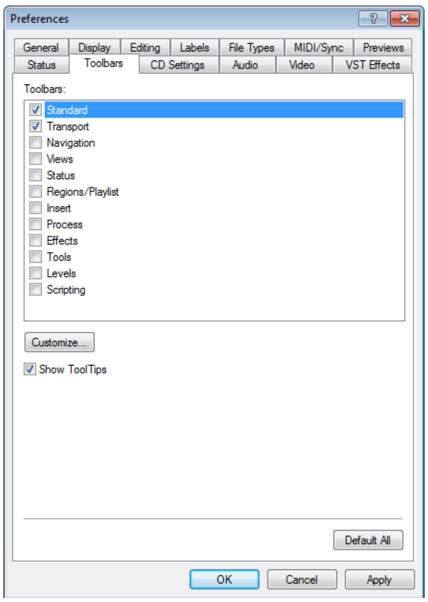
Sound Forge toolbars contain buttons used to quickly perform many of the program's commands and functions. Toolbars can be dragged throughout the workspace, docked, resized, hidden, and customized.

You can use the **Toolbars** tab in the Preferences dialog to specify which toolbars you want to display. Perform either of the following actions to display this tab:

- From the **Options** menu, choose **Preferences**. When the Preferences dialog is displayed, click the **Toolbars** tab.
- From the **View** menu, choose **Toolbars**.

Displaying a toolbar

1. From the **View** menu, choose **Toolbars**. The Preferences dialog is displayed with a list of available toolbars.
2. To display a toolbar, select the corresponding check box and click **OK**.

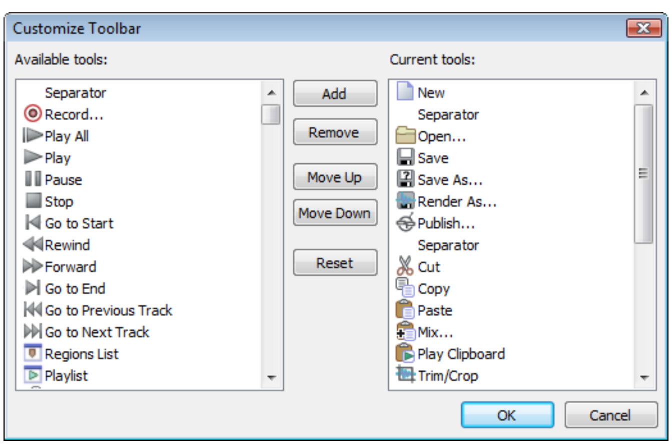


Select a check box to display a toolbar.

Click **Customize** to add, remove, or rearrange buttons on a toolbar.

Customizing a toolbar

1. From the **View** menu, choose **Toolbars**. The Preferences dialog is displayed with a list of available toolbars.
2. Select the check box for a toolbar and click **Customize**. The Customize Toolbar dialog is displayed.



3. Use the controls in the Customize Toolbar dialog to add, remove, or rearrange the buttons on the selected toolbar. Click **Reset** to restore the toolbar to its default setting.
4. Click **OK**.

Docking a toolbar

When you drag a floating toolbar to any edge of the main screen, the toolbar docks on that edge.

Floating a toolbar

When you drag a docked toolbar away from an edge, the toolbar becomes a floating toolbar.

Standard toolbar

The Standard toolbar is displayed by default when you start the application. The buttons on this toolbar provide quick access to many common commands.

New Creates a new data window. *For more information, see Creating data windows on page 71.*

Open Displays the Open dialog. *For more information, see Using the Open dialog on page 62.*

Save Saves the current audio data. *For more information, see Saving a file on page 72.*

Save As Saves the current file with a new name or format. *For more information, see Using the Save As/Render As dialog on page 72.*

Render As Renders the current project file to a media file. *For more information, see Using the Save As/Render As dialog on page 72.*

Publish Opens the Publish Setup wizard so you can upload your media file to the Web. *For more information, see Publishing files to the Web on page 89.*

Cut Removes selected audio data and places it on the clipboard. This command has no effect if there is no selection. *For more information, see Cutting on page 81.*

Copy Copies selected audio data to the clipboard. This command has no effect if there is no selection. *For more information, see Copying on page 79.*

Paste Inserts a copy of the clipboard data at the current insertion point. If there is a selection, this command replaces the selected data with the clipboard data. *For more information, see Pasting on page 80.*

Mix Mixes a copy of the clipboard data with the current audio file. The mix start point is either the cursor point or the start or end of the selection in the destination data window. *For more information, see Mixing on page 83.*

Play Clipboard Plays the audio on the clipboard. *For more information, see Previewing clipboard contents on page 79.*

Trim/Crop Removes all data from the file that is not currently selected. This command has no effect if there is no selected data. This command does not copy data to the clipboard. *For more information, see Trimming/Cropping on page 83.*

Undo Reverses the last edit operation. *For more information, see Using Undo and Redo on page 85.*

Redo Reverts the previously undone edit operation. *For more information, see Using Undo and Redo on page 85.*

Repeat Repeats the last operation. This command can be used with most processing functions. The previous operation's parameters are repeated. To specify new parameters, hold Shift and click this button. *For more information, see Repeating an operation on page 159.*

Edit Tool Selects the Edit tool.

Magnify Tool Selects the Magnify tool. *For more information, see Using the Magnify tool on page 96.*

Pencil Tool Selects the Pencil tool. *For more information, see Repairing audio glitches manually with the Pencil tool on page 165.*

Event Tool Selects the Event tool. *For more information, see Using the Event Tool on page 171.*

Envelope Tool Selects the Envelope tool. *For more information, see Adjusting envelopes on page 225.*

Interactive Tutorials Opens the Interactive Tutorials window where you can select tutorials and learn about the features in Sound Forge.

Transport toolbar

The Transport toolbar also displays by default and contains basic audio transport buttons.

Record Click to display the Record dialog. *For more information, see* *Recording on page 143*.

Loop Playback Plays the selected data in a continuous mode.

> **Note:** *If there is no selection, the entire sound file is played in an endless loop.*

Play All Click to play the entire file from beginning to end, regardless of cursor position, selection, or playlist.

Playback for musical instrument files behaves slightly differently than playback in a normal data window.

- If no samples are selected, click **Play All** to play all samples in the data window.
- If you have samples selected, click **Play All** to play all selected samples in the data window.

Play Click to play back the file in the current playback mode.

Playback for musical instrument files behaves slightly differently than playback in a normal data window.

- If no samples are selected, click **Play** to play all samples from the cursor position to the end of the data window.
- If you have samples selected, click **Play** to play all selected samples from the cursor position to the end of the data window.

> **Tip:** *Select the* **Spacebar and F12 Play/Pause instead of Play/Stop** *check box in the General Preferences tab if you want the F12 and spacebar keyboard shortcuts to toggle between Play and Pause mode. In this mode, the cursor will maintain its position.*

Pause Pauses playback and maintains the cursor at its current position.

Stop Stops playback and returns the cursor to its prior position.

Go to Start Moves the cursor to the start of the file.

Go to Previous Track Moves the cursor to the previous disc-at-once track or index. Hold Ctrl while clicking to skip index markers, or hold Shift to extend a selection.

> **Note:** *This button is available only if disc-at-once tracks are present in your data window.*

Rewind Moves the cursor backward in the current file.

Forward Moves the cursor forward in the current file.

Go to Next Track Moves the cursor to the next disc-at-once track or index. Hold Ctrl while clicking to skip index markers, or press Shift to extend a selection.

> **Note:** *This button is available only if disc-at-once tracks are present in your data window.*

Go to End Moves the cursor to the end of the file.

Navigation toolbar

The Navigation toolbar contains buttons used to navigate within the current data window.

Zoom In Full Magnifies the selected area to a 24:1 ratio.	**Cursor to Selection End** Moves the cursor to the end of the selection.
Zoom Normal Resets the audio data to its original magnification.	**Center Sustaining Start** Moves the cursor to the beginning of the sustaining loop.
Zoom Selection Maximizes the selection vertically and horizontally.	**Center Sustaining End** Moves the cursor to the end of the sustaining loop.
Custom Zoom 1 Sets the audio data to a custom time magnification level.	**Center Release Start** Moves the cursor to the beginning of the release loop.
Custom Zoom 2 Sets the audio data to a custom time magnification level.	**Center Release End** Moves the cursor to the end of the release loop.
Mark In Marks the "in" point of a new selection. *For more information, see Selecting audio during playback on page 99.*	**Double Selection** Doubles the size of the current selection.
Mark Out Marks the "out" point of a new selection. *For more information, see Selecting audio during playback on page 99.*	**Halve Selection** Divides the current selection in half.
Go To Displays the Go To dialog and allows you to quickly move the cursor to a specific point in a file. *For more information, see Setting the cursor position on page 91.*	**Shift Selection Left** Shifts the current selection to the left so the current start point becomes the end point.
Cursor Center Centers the display with the cursor displayed in the center of the data window.	**Shift Selection Right** Shifts the current selection to the right so the current end point becomes the start point.
Cursor to Selection Start Moves the cursor to the beginning of the selection.	

Tempo box

The Navigation toolbar also contains a **Tempo** box that appears to the right of the toolbar buttons. This box calculates and displays the tempo of the current selection as is if the selection represents a complete measure.

Views toolbar

The Views toolbar contains buttons used to store and retrieve data window views.

Toggles views 1-8 between setting and restoring.	2 Stores and recalls specific selection views.

Status toolbar

The Status toolbar contains buttons used to specify a file's status format and control snapping functions.

Samples Changes the status format to Samples.	**SMPTE EBU** Changes the status format to SMPTE EBU (25 fps).
Time Changes the status format to Time.	**SMPTE Non-Drop** Changes the status format to SMPTE Non-Drop (29.97 fps, Video).
Seconds Changes the status format to Seconds.	**SMPTE Drop** Changes the status format to SMPTE Drop (29.97 fps, Video).
Time & Frames Changes the status format to Time & Frames.	**SMPTE 30** Changes the status format to SMPTE 30 (30 fps, Audio).
Absolute Frames Changes the status format to Absolute Frames.	**Audio CD Time** Changes the status format to Audio CD Time.
Measures & Beats Changes the status format to Measures & Beats.	**Edit Tempo** Calculates the musical tempo (beats per minute) based upon the current selection.
SMPTE Film Sync (24 fps) Changes the status format to SMPTE Film Sync (24 fps).	

Regions/Playlist toolbar

The Regions/Playlist toolbar contains the **Regions List** and **Playlist** buttons as well as buttons corresponding to synchronization commands and status displays.

Regions List Displays the Regions List. *For more information, see Using the Regions List on page 129.*

Playlist Displays the playlist. *For more information, see Using the Playlist on page 130.*

Trigger from MIDI Timecode Configures the software to be triggered by MIDI commands received through the MIDI input port. The MIDI input port is specified on the MIDI/Sync tab in the Preferences dialog. *For more information, see Triggering from MIDI timecode on page 271.*

Generate MIDI Timecode Configures the software to send MIDI timecode through the MIDI output port. The MIDI output port is specified on the MIDI/Sync tab of the Preferences dialog.

Pre-Queue for MIDI Timecode Opens the wave device and preloads data for the next region to be played from the playlist.

Playlist Position display

Displays the current playback position of an audio file being played from the playlist. Right-clicking this box displays a shortcut menu that allows you to specify a new format.

Sync Status display

Allows you to monitor the status of incoming/outgoing MIDI commands.

Insert toolbar

The Insert toolbar contains buttons corresponding to all commands located in the **Insert** menu.

Insert Marker Inserts a marker at the cursor location. *For more information, see Using markers on page 119.*

Insert Region Inserts region tags at the beginning and end of the current selection. *For more information, see Using regions on page 121.*

Insert Sample Loop Inserts sustaining loop tags at the beginning and end of the current selection. *For more information, see Creating a sustaining loop on page 277.*

Insert Command Inserts a command marker at the cursor location. *For more information, see Using commands on page 127.*

Insert CD Track Inserts a CD track using the current selection as the track length. *For more information, see Creating and editing tracks for disc-at-once CDs on page 310.*

Insert CD Index Inserts a CD index marker at the cursor location. *For more information, see Creating and editing tracks for disc-at-once CDs on page 310.*

Insert Volume Envelope Adds a volume envelope to the active data window. *For more information, see Adding a volume or panning envelope on page 223.*

Insert Pan Envelope Adds a panning envelope to the active data window. *For more information, see Adding a volume or panning envelope on page 223.*

Insert Silence Inserts user-configurable silence into audio files. *For more information, see Inserting silence on page 159.*

DTMF/MF Tones Synthesis Generates dial tones used by telephone companies. *For more information, see Generating DTMF/MF tones on page 166.*

FM Synthesis Uses frequency modulation and additive synthesis to create complex sounds from simple waveforms. *For more information, see Generating audio with frequency modulation on page 167.*

Simple Synthesis Generates a simple waveform of a given shape, pitch, and length. *For more information, see Generating simple waveforms on page 169.*

Process toolbar

The Process toolbar contains buttons corresponding to all commands located in the **Process** menu.

Auto Trim/Crop Removes silence and automatically fades in/out the end-points of each phrase. *For more information, see Auto Trim/Crop on page 185.*

Bit-Depth Converter Converts a file to a different bit depth. *For more information, see Bit-Depth Converter on page 186.*

iZotope MBIT+ Dither Converts a file to a different bit depth and applies dithering. *For more information, see iZotope MBIT+ Dither on page 188.*

Channel Converter Converts between mono and multichannel formats. Can also intermix the channels of a file to create panning effects. *For more information, see Channel Converter on page 189.*

DC Offset Changes the baseline of an audio file. *For more information, see DC Offset on page 191.*

Graphic EQ Opens the XFX Graphic EQ. *For more information, click the* **Help** *button (?) in the process dialog.*

Paragraphic EQ Opens the XFX Paragraphic EQ. *For more information, click the* **Help** *button (?) in the process dialog.*

Parametric EQ Opens the XFX Parametric EQ. *For more information, click the* **Help** *button (?) in the process dialog.*

Graphic Fade Creates user-configurable fades. *For more information, see Fade - Graphic Fade on page 192.*

Fade In Fades-in the selection. *For more information, see Fade - Fade In on page 194.*

Fade Out Fades-out the selection. *For more information, see Fade - Fade Out on page 194.*

Invert/Flip Inverts (or flips) the polarity of the current selection. *For more information, see Invert/Flip on page 194.*

Mute Mutes the current selection. *For more information, see Mute on page 195.*

Normalize Normalizes the loudness of an audio file. *For more information, see Normalize on page 196.*

Pan/Expand Creates custom pans, expands, and mixes. *For more information, see Pan/Expand on page 198.*

Resample Creates a copy of the audio file with a new sample rate. *For more information, see Resample on page 200.*

iZotope 64-Bit SRC Changes the sample rate of an existing file. *For more information, see iZotope 64-Bit SRC on page 202.*

Reverse Reverses the current selection. *For more information, see Reverse on page 203.*

Rotate Audio Moves the current selection to the opposite end of the file. *For more information, see Rotating audio on page 286.*

Smooth/Enhance Opens the XFX Smooth/Enhance tool. *For more information, click the* **Help** *button (?) in the process dialog.*

Time Stretch Opens the XFX Time Stretch tool. *For more information, click the* **Help** *button (?) in the process dialog.*

élastique Timestretch Opens the élastique Timestretch tool. *For more information, click the* **Help** *button (?) in the process dialog.*

Volume Adjusts the volume of an audio file. *For more information, see Volume on page 204.*

Effects toolbar

The Effects toolbar contains buttons corresponding to all Sound Forge built-in XFX™ plug-ins.

Acoustic Mirror Adds environmental coloration to your existing recordings. *For more information, see What are the Acoustic Mirror effects? on page 233.*

Amplitude Modulation Applies a sinusoidal or square-shaped periodic gain to the input signal. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Chorus Simulates multiple audio sources from a single sound. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Multi-Tap Delay Creates a delay with up to eight delay-taps spaced anywhere within 2.5 seconds of the original sound. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Simple Delay Adds a delayed copy of the audio signal to the file. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Distortion Simulates the overloading of an amplifier. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Graphic Dynamics Applies compression, expansion, and limiting to affect the dynamic range of an audio file. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Multi-Band Dynamics Allows compression and limiting to be placed on up to four different frequency bands. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Envelope Forces the amplitude envelope of a waveform to match a specified envelope shape. *For more information, see Envelope on page 227.*

Flange/Wah-Wah Mixes a modulated delay signal with the original signal. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Gapper/Snipper Removes/inserts sections of silence at regular intervals to create unusual effects. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Noise Gate Removes signals below a set amplitude threshold. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Pitch Bend Creates a modified sound envelope that corresponds to increasing or decreasing the pitch of a sound file over time. *For more information, see Bend on page 229.*

Pitch Shift Changes the pitch of a selection with or without preserving the duration of the file. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Resonant Filter Restricts the range of a sound using low-pass, band-pass, or high-pass filtering, and then boosts and adds oscillation to the resonant frequency. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Reverb Simulates the acoustics of different environments. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Vibrato Creates periodic pitch modulation in an audio file. *For more information, click the* **Help** *button ( ) in the effect dialog.*

Wave Hammer Acts as a classic compressor and volume maximizer. *For more information, see What is the Wave Hammer plug-in? on page 245.*

Tools toolbar

The Tools toolbar contains buttons corresponding to commands in the **Tools** menu.

Extract Audio from CD Extracts audio from CD and opens for editing. *For more information, see Extracting audio from CDs on page 76.*

Burn Track-at-Once CD Burns the selected audio track to CD. *For more information, see Burning track-at-once (TAO) CDs on page 308.*

Burn Disc-at-Once CD Burns a disc-at-once CD using the current CD layout. *For more information, see Burning disc-at-once (DAO) CDs on page 310.*

Auto Region Creates regions in an audio file according to rapid sound attacks or a specified time interval. *For more information, see Creating regions based on fast attacks on page 123 and Creating regions based on a musical time interval on page 124.*

Extract Regions Extracts all file regions and saves them as individual files. *For more information, see Extracting regions to new files on page 124.*

Detect Clipping Performs clip detection on the current file or selection. *For more information, see Detecting and marking clipping on page 120.*

Find Searches for clicks and pops, volume levels, or silent breaks in an audio signal. *For more information, see Finding and repairing audio glitches on page 163.*

Interpolate Replaces selected audio with interpolated audio data based on the selection's beginning and end samples. *For more information, see Interpolating new audio on page 164.*

Replace Replaces selected audio data with previous adjacent data. *For more information, see Replacing audio with preceding data on page 165.*

Copy Other Channel Replaces selected audio with a corresponding selection from the opposite channel. *For more information, see Copying the other channel on page 164.*

Noise Reduction Analyzes and removes background noise such as tape hiss, electrical hum, and machinery rumble.

For more information, click the **Help** *button ( ) in the plug-in dialog.*

Click and Crackle Removal Detects and removes all clicks and pops. *For more information, click the* **Help** *button ( ) in the plug-in dialog.*

Clipped Peak Restoration Rounds the tops of clipped peaks and applies peak limiting to the area immediately surrounding the audio clip. *For more information, click the* **Help** *button ( ) in the plug-in dialog.*

Audio Restoration Removes clicks and background noise associated with vinyl records. *For more information, click the* **Help** *button ( ) in the plug-in dialog.*

Batch Converter Allows you to modify and manipulate multiple audio files without having to process each file individually. *For more information, see Using the Batch Converter on page 255.*

Crossfade Loop Mixes audio occurring before the loop start point into the end of the loop to smooth transitions. *For more information, see Crossfading loops on page 282.*

Sampler Allows you to transfer samples to/from the Sound Forge application. *For more information, see Sampling on page 259.*

Statistics Displays statistics corresponding to the current file or selection. *For more information, see Viewing selection statistics on page 68.*

Preset Manager Backs up and transfers user-configured presets from effects, processes, and plug-ins. *For more information, see Using the Preset Manager on page 226.*

Edit in SpectraLayers Pro Click to open the active data window in SpectraLayers Pro.

When you're done editing, close SpectraLayers Pro. You'll be prompted to export your changes back to Sound Forge Pro. Click **Yes**, and the Sound Forge Pro data window is updated to reflect any changes.

For more information, see Editing with SpectraLayers Pro on page 305.

Send to SpectraLayers Pro Click to open the active data window as a layer in SpectraLayers Pro.

When you're done editing, you can save your project in SpectraLayers Pro, render the mixed output, or use **Process** > **Send to Sound Forge Pro** to save your changes.

For more information, see Editing with SpectraLayers Pro on page 305.

Levels toolbar

The Levels toolbar displays the audio levels in the left and right channels in the user-specified format. You can right-click to choose the format from a shortcut menu.

Scripting toolbar

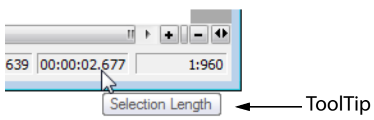
The Scripting toolbar allows you to show, hide, or activate the Script Editor and display the Batch Converter window. You can also add buttons for scripts to the toolbar. *For more information, see Using the Scripting toolbar on page 253.*

Script Editor Allows you to create, edit, or run scripts. *For more information, see Using the Script Editor window on page 249.*	**Batch Converter** Allows you to modify and manipulate multiple audio files without having to process each file individually. *For more information, see Using the Batch Converter on page 255.*

ToolTips

Using ToolTips

Hovering the mouse pointer over a button or status bar box for longer than one second displays a small text box adjacent to the pointer. This text, called a ToolTip, is a brief description of the item's function. Using ToolTips is an effective way to quickly familiarize yourself with features.



Turning off ToolTips

1. From the **View** menu, choose **Toolbars**. The Preferences dialog is displayed.
2. Clear the **Show ToolTips** check box and click **OK**.

Command descriptions

When you click and hold a menu item or a button in a toolbar, a brief description of the command appears in the lower-left corner of the status bar. If you release the mouse button outside of the menu item or toolbar, the command is not executed.

Keyboard shortcuts

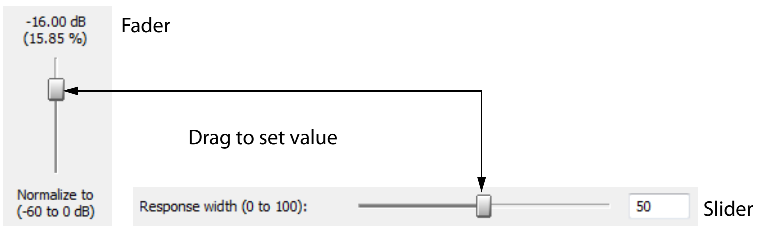
The Keyboard map allows you to customize the keyboard shortcuts available in the Sound Forge interface. You can access the Keyboard map by choosing **Customize Keyboard** from the **Options** menu. *For more information, see Customizing keyboard shortcuts on page 346.*

Controls

A major step in mastering Sound Forge software is becoming familiar with the controls used to set and adjust feature parameters, including faders, sliders, and envelope graphs.

Faders and sliders

Faders and sliders are frequently used to edit effect and process parameters. To use either control, drag the control to the desired position and release.



Resetting fader and slider values

Double-click to return the control to its default value.

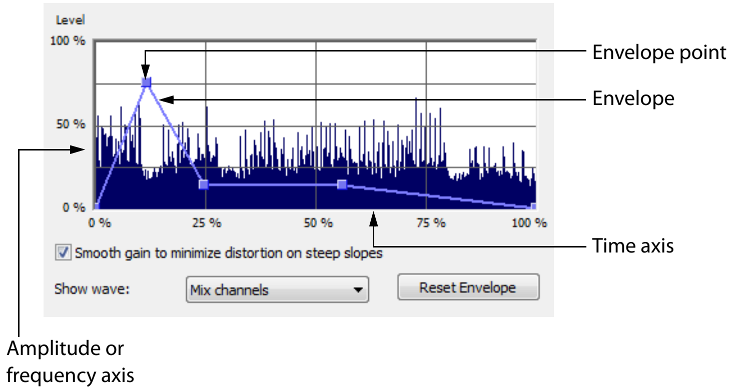
Fader and slider shortcuts

There are numerous keyboard shortcuts available when using faders and sliders.

If you want to	Then use the following shortcuts
Change the value in small increments	Up Arrow, Down Arrow, Left Arrow, and Right Arrow —or— Hover the mouse over the fader or slider control and press Ctrl while moving the mouse wheel.
Change the value in larger increments	Page Up and Page Down —or— Hover the mouse over the fader or slider control and move the mouse wheel.
Set the control to its maximum and minimum values respectively	Home and End

Envelope graphs

Envelope graphs are used to configure the shape of frequency or amplitude envelopes applied to audio waveforms.



Understanding the envelope graph

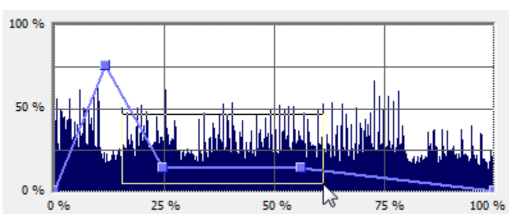
To use the envelope graph, you must first understand what it represents. In the previous example, the horizontal axis represents time, with the leftmost point representing the start of the selection and the rightmost point representing the end of the selection. The vertical axis represents either amplitude or frequency, depending upon the operation.

Moving an envelope point

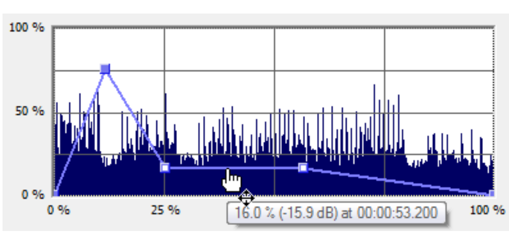
1. Drag an envelope point to a new position.
2. Release the mouse button. The point is repositioned and the envelope adjusts.

Moving multiple envelope points

1. Starting in an unused area of the envelope graph, drag the mouse to create a selection box containing all points to be moved.



2. Release the mouse button. The selected envelope points are displayed with a white square center.
3. Drag any of the selected envelope points to the desired position. The pointer displays as a multi-directional arrow and the selected points move together.
4. Release the mouse button. The entire envelope graph adjusts.



Reposition multiple envelope points

Changing the fade curve between two points

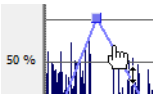
To change the type of fade between two envelope points, right-click an envelope segment and choose a fade type (Linear Fade, Fast Fade, Slow Fade, Smooth Fade, Sharp Fade, and Hold) from the shortcut menu.

Selecting or clearing all envelope points

Press Ctrl+A to select or clear all envelope points.

Adding an envelope point

1. Hover over the envelope.



Place the pointer on the envelope and double-click to add a point.

2. Double-click the mouse. A point is added to the envelope graph and can be positioned as needed. *For more information, see* *Moving an envelope point* *on page* *49*.

Deleting an envelope point

Right-click the point to be deleted and choose **Delete** from the shortcut menu. The point is deleted and the envelope adjusts.

Delete all points

Delete all envelope points by clicking the **Reset Envelope** button.

Displaying the waveform on an envelope graph

Certain envelope graphs (such as in the Graphic Fade dialog) allow you to view the audio waveform on the graph. If the selection is small, the waveform is automatically displayed. Otherwise, selecting an option from the **Show wave** drop-down list displays the waveform.

Displaying multichannel waveforms

The **Show Wave** drop-down list allows you to specify how multichannel files appear in the envelope graph.

Multichannel files

When a data window displays a multichannel file, all channels are shown at the same time.

Working with multichannel files

When playing, editing, or processing multichannel files, you can select a single channel or all channels. However, certain processing tasks cannot be performed on an individual channel of a multichannel file. *For more information, see* *Single-channel editing* *on page* *52*, or *Editing Multichannel Audio* *on page* *115*.

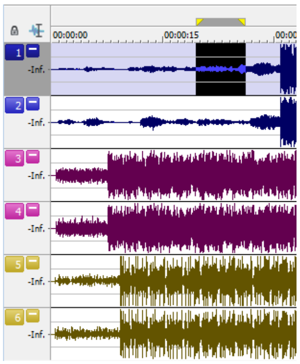
Selecting data in multichannel files

When editing a multichannel file, you can use the mouse to select data by clicking and dragging in a data window. There are several options for data selection in multichannel files.

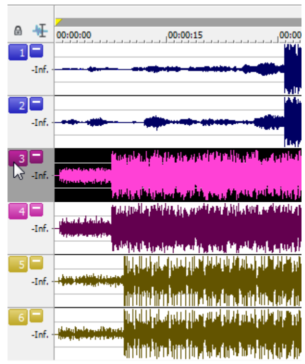
1. Open a multichannel file.
2. From the **Edit** menu, choose **Tool**, and then choose **Edit** from the submenu to select the Edit tool.

> **Tip:** *Press Ctrl+D or click the* **Edit Tool** *button ( ) on the Standard toolbar.*

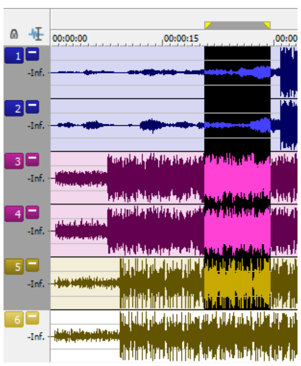
3. Select the data:



Drag within a channel to select that channel only.



Double-click a channel number to select the entire channel.



Drag across channels to select multiple channels.

Drag along the divider between channels (or the loop bar above the ruler) to select all channels.

Hold Ctrl and click a channel to add or remove it from the current selection.

Toggling channel selections

After you place the cursor or create a selection in a multichannel file, you can cycle through channel options by pressing Tab.

Previewing channels

The single channel selection option allows you to preview channels in a multichannel file individually.

1. Open a multichannel file and select all data.
2. Click the **Play Normal** button (). All channels play. Click the **Stop** button ().
3. Press Tab. The first channel is selected.
4. Click the **Play Normal** button (). Only the first channel plays. Click the **Stop** button ().
5. Press Tab. The second channel is selected.
6. Click the **Play Normal** button (). Only the second channel plays. Click the **Stop** button ().

Single-channel editing

You have the ability to cut, copy, and paste data in single channels of a multichannel file. However, channel lengths must always remain equal in multichannel files. *For more information on cutting, copying, and pasting data, see Editing audio on page 79.*

Chapter 3

Metadata windows

From the View menu, choose **Metadata**, and then choose a command from the submenu to display metadata windows, where you can view and edit information about the current data window.

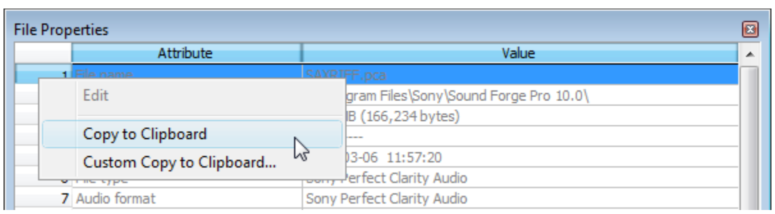
Tips:

- *If you want to sort the contents of a metadata window, you can click a column heading to sort in ascending or descending order.*
- *If you want to display all metadata windows docked together, choose* **View** > **Metadata** > **Show All***, and then choose a command from the submenu to indicate where you'd like to display the docked window.*

Copying metadata to the clipboard

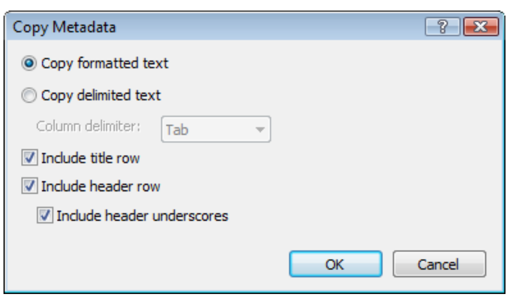
Note: *These procedures do not apply to the Regions List, Playlist, and Track List.*

If you want to copy metadata to the clipboard, right click the window and choose **Copy to Clipboard** from the shortcut menu.



If you want to customize the format for copying metadata to the clipboard, right-click the window and choose **Custom Copy to Clipboard** from the shortcut menu.

The Copy Metadata dialog is displayed to let you choose whether you want to copy the data as formatted text or delimited text, set a delimiter, and choose whether you want to include a header row.



Click **OK** to copy the metadata to the clipboard, and you can then paste the information wherever you need it.

Regions List window (Ctrl+Alt+M, 0)

The Regions List window contains all regions and markers that exist in the active data window. From the **View** menu, choose **Metadata**, and then choose **Regions List** from the submenu to show or hide the Regions List window. *For more information, see Using the Regions List on page 129.*

Playlist window (Ctrl+Alt+M, 1)

The Playlist window is used to arrange regions for playback. From the **View** menu, choose **Metadata**, and then choose **Playlist** from the submenu to show or hide the Playlist window. *For more information, see Using the Playlist on page 130.*

Track List window (Ctrl+Alt+M, 2)

The Track List window is used to arrange tracks for a disc-at-once CD. From the **View** menu, choose **Metadata**, and then choose **Track List** from the submenu to show or hide the Track List window. *For more information, see Using the Track List window on page 318*.

ACID Properties window (Ctrl+Alt+M, 3)

From the View menu, choose **Metadata**, and then choose **ACID Properties** from the submenu to display the ACID Properties window, where you can view and edit ACID-specific information in a sound file. *For more information about creating ACID loops, see Creating loops for ACID software on page 283*

<table>
<tr><th>Item</th><th colspan="2">Description</th></tr>
<tr><td>Time signature</td><td colspan="2">Displays the number of beats in your clip and the note that receives one beat.
You can double-click the value to edit it.</td></tr>
<tr><td rowspan="4">ACID type</td><td colspan="2">Displays the clip’s ACID type.
Click the down arrow (▼) and choose a setting from the drop-down list to change the clip type:</td></tr>
<tr><td>One-Shot</td><td>Choose One-Shot if you want ACID to treat your file as a one-shot.
One-shots are RAM-based audio clips that do not change tempo or pitch with an ACID project and are not designed to loop. Sounds such as cymbal crashes and sound bites could be considered one-shots. Longer files can be treated as one-shots if your computer has sufficient memory.</td></tr>
<tr><td>Loop</td><td>Choose Loop and specify a Number of beats and Root note for transposing if you want ACID to treat your file as a loop.
Loops are small audio clips that are designed to create a repeating beat or pattern. Loops are usually one to four measures long and are stored completely in RAM for playback. Loop files change tempo and can pitch shift with an ACID project.
• Root note for transposing Click the down arrow (▼) and choose a note from the drop-down list to set the base note for tracks that you want to conform to the project key.
If you do not want a track transposed to the project key (a track that contains a drum sample, for example) choose Don’t transpose.
• Number of beats Double-click to edit the length of the file. Selecting a value that does not match the actual file will cause ACID to play the loop at a different speed than normal. For example, specifying a length of 8 beats for a 4-beat loop will cause the loop to play at half speed at any given tempo.</td></tr>
<tr><td>ACID Beatmapped</td><td>Choose ACID Beatmapped if you want to add key and tempo information to a long audio file. By default, ACID will start the Beatmapper Wizard for files longer than 30 seconds.
• Root note for transposing Click the down arrow (▼) and choose a note from the drop-down list to set the base note for tracks that you want to conform to the project key.
If you do not want a track transposed to the project key (a track that contains a drum sample, for example) choose Don’t transpose.
• Tempo Double-click to edit the original tempo of the clip.
• Downbeat offset (samples) Double-click to edit the location (in samples) of the track’s first downbeat.</td></tr>
</table>

Broadcast Wave window (Ctrl+Alt+M, 4)

From the View menu, choose **Metadata**, and then choose **Broadcast Wave** from the submenu to display the Broadcast Wave Information window, where you can view and edit information about a Broadcast Wave Format (BWF) file.

Tips:

- *If you want to sort the contents of a metadata window, you can click a column heading to sort in ascending or descending order.*
- *If you want to sort iXML and BEXT BWF metadata objects, click the* **Type** *column to sort by metadata type. If you want to delete the iXML chunk select all iXML objects, right-click a selected row, and choose* **Delete**.
- *For iXML and BEXT metadata objects that support multiline data, press Ctrl+Enter to add a line break.*
- *If you copy data from a BWF file to a file that does not contain BWF metadata, the source metadata is updated as necessary and added to the destination file.*
- *If you copy data from a BWF file to a file that contains BWF metadata, the metadata in the destination file is not overwritten. The time stamp and date will be updated if sound data is inserted or removed before other data in the file. You can use the* **Autoupdate BWF Origination Time Reference** *setting on the General Preferences tab to choose whether the* **OriginationTimeRef** *metadata is updated when adding or deleting sound data at the beginning of a Broadcast Wave Format file.* *For more information, see General tab on page 331.*
- *If you want to copy metadata to the clipboard, right-click the window and choose* **Copy to Clipboard** *from the shortcut menu.*
- *If you want to customize the format for copying metadata to the clipboard, right-click the window and choose* **Custom Copy to Clipboard** *from the shortcut menu.*

  The Copy Metadata dialog is displayed to let you choose whether you want to copy the data as formatted text or delimited text, set a delimiter, and choose whether you want to include a header row.

  Click **OK** *to copy the metadata to the clipboard, and you can then paste the information wherever you need it.*

Editing metadata

You can double-click values in the Broadcast Wave Information window to edit them.

Autopopulating metadata values

If you want to populate metadata automatically, right-click the Broadcast Wave Information window, choose **Autopopulate** from the shortcut menu, and then choose **BEXT** or **iXML** from the submenu. Sound Forge Pro will create metadata values based on file properties where possible and will create blank metadata entries for the **Description**, **Originator**, **OriginatorRef**, and **CodingHistory** items.

Verifying metadata

If an object's metadata is not valid, the **Value** column is displayed in blue in the Broadcast Wave Information window, and a warning is displayed in the **Description** column:

50	Track Count	-2	iXML	-2 should match the number of audio channels in this file: 1

If you right-click the row in the Broadcast Wave Information window, you can choose a command from the shortcut menu:

- **Set to**: If Sound Forge can suggest a compliant metadata value, you can choose **Set to <suggested value>** to correct the value.
- **Hide Warnings**: Hides warnings for all objects. You can right-click again and choose **Show Warnings** to restore warning messages.
- **Hide This Object's Warnings**: Hides warnings for the current object. You can right-click again and choose **Show This Object's Warnings** to restore warning messages for the current object, or choose **Stop Hiding of All Object-Specific Warnings** to restore warning messages.

Inserting metadata objects

If the data you want to edit is not displayed in the window, you can right-click the window, choose **Insert** from the submenu, and then choose a metadata object from the submenu.

You can choose to add all BWF/BEXT/iXML objects at once, or you can choose commands from the submenus to insert individual objects.

Deleting metadata objects

Right-click the window, choose **Delete** from the submenu, and then choose a metadata object from the submenu.

You can choose to delete selected objects, all BWF/BEXT/iXML objects, or you can choose commands from the submenus to delete individual objects.

Changing the BWF version for saved metadata

If you want to change the version of BWF metadata that is saved in your file, click the **Value** column in the **BWF Version** row and choose **Version 0**, **Version 1**, or **Version 2** from the menu.

- Version 1 expands on the Version 0 metadata set by adding SMPTE UMID.
- Version 2 expands on the Version 1 metadata set by adding loudness metadata.

CD Information window (Ctrl+Alt+M, 5)

From the View menu, choose **Metadata**, and then choose **CD Information** from the submenu to display the CD Information window, where you can view and edit information about a disc-at-once audio CD.

Item	Description
Universal Product Code/ Media Catalog Number	Universal product codes (UPC) or media catalog numbers (MCN) can be written to a CD as a means of identification. However, not all CD-R drives support this feature. Check your CD-R drive documentation to determine if your drive will write these codes. Type the code in this box, and the codes will be written to the CD with the rest of the project. Universal product codes are administered by the Uniform Code Council. *For more information, see http://www.uc-council.org/*.
First track number on disc	Type a number in the box to specify the track number of the first track. **Note:** *Specifying a value other than 1 will produce a valid Red Book CD, but some audio CD players may be unable to play the disc.*
Name/Title (CD Text)	Type a title for the project. If you select the **Write CD Text** check box on the Burn Disc-at-Once CD dialog, this data will be written to your disc. In order to display CD Text, your CD player must support CD Text. **Notes:** • *In order to burn valid CD Text, you must specify a title for the disc and for each track on the disc (artist information is optional). If the* **Name/Title** *box in the CD Information or Track List window is left blank, a warning will be displayed before burning so you can choose to write the disc without CD Text or cancel burning and add title information as needed.* • *You can write a maximum of 5000 characters as CD Text.*
Artist (CD Text)	Type the name of the artist. If you select the **Write CD Text** check box on the Burn Disc-at-Once CD dialog, this data will be written to your disc. In order to display CD Text, your CD player must support CD Text.
Engineer	Type the name of the person who mixed or edited the project.
Copyright	Type copyright information for the project.

Item	Description
Comments	Type any comments you want to associate with the project.

Sampler Loops window (Ctrl+Alt+M, 6)

From the View menu, choose **Metadata**, and then choose **Sampler Loops** from the submenu to view or edit loops and sampler information saved in the active file.

Item	Description
Sample type	Displays the type of sample loop you're creating. Click the down arrow (▼) to choose a new setting. • **None** – Removes the loop from the file. • **One shot** – Causes the sound file to play normally with no loops. • **Sustaining** – Causes the file to repeat the sustaining loop region the specified number of times. • **Sustaining with release** – Causes the sound file to play the sustaining loop region the number of times you specify, play the region between the sustaining and release loops, and then play the release loop region the number of times you specify.
Sustain start	Displays the beginning of the sustaining loop. You can double-click the value to edit it.
Sustain end	Displays the end of the sustaining loop. You can double-click the value to edit it. Editing the end will automatically update the **Sustain length** value.
Sustain length	Displays the length of the sustaining loop. You can double-click the value to edit it. Editing the length will automatically update the **Sustain end** value.
Sustain count	Indicates how many times the sustaining loop should be played. Click the down arrow (▼) to choose a new setting. If you choose **Custom**, you can type a new value.
Release start	Displays the start of the release loop. You can double-click the value to edit it.
Release end	Displays the end of the release loop. You can double-click the value to edit it. Editing the end will automatically update the **Release length** value.
Release length	Displays the length of the release loop. You can double-click the value to edit it. Editing the length will automatically update the **Release end** value.
Release count	Indicates how many times the release loop should be played. Click the down arrow (▼) to choose a new setting. If you choose **Custom**, you can type a new value. For example, if you set the **Sustain count** to 3 and the **Release count** to 2, the sustain loop would be played 3 times, and then the release loop would be played twice.
Manufacturer	The MMA manufacturer code for the target device. If the sample is not intended for a specific manufacturer, set the value to 0. *For more information, see http://www.midi.org/techspecs/manid.php*.
Product	The sampler that created the sample.
Sample period	Click the down arrow (▼) and choose a sample period from the drop-down list, or choose **Custom** to type a value in the edit box to set the duration of each sample in nanoseconds.
Unity note	Indicates the MIDI note that will cause a sampler to play the sound file at the pitch (sample rate) it was originally recorded. Click the down arrow (▼) to choose a new unity note and octave.

Item	Description
Fine tune	Allows you to pitch shift the unity note up from 0 to 99.999 cents. Sound Forge software does not fine-tune the sound file when fine tuning is used, and not all samplers support the setting. This option is an informational setting that will be transmitted to a sampler via a sample-transfer procedure. A sampler such as the K2000 can use this information to play back the sample. The K2000 should accurately display this information on the Master/Sample/Misc. page as Pitch Adjust.
SMPTE format	Indicates the type of SMPTE offset that has been set for the file.
SMPTE offset	Indicates whether a SMPTE time offset has been set for the file. Sound Forge software ignores this offset value, and not all samplers can store a SMPTE offset value in the sample.

Summary Information window (Ctrl+Alt+M, 7)

From the View menu, choose **Metadata**, and then choose **Summary Information** from the submenu to display the Summary Information window, where you can view and edit information saved in the active file.

If the data you want to edit is not displayed in the window, you can right-click the window, choose **Insert** from the shortcut menu, and then choose a metadata field from the submenu.

FourCC Code	Name	Description
IARL	**Archival Location**	Indicates where the subject of the file is archived.
IART	**Artist (CD Text)**	The artist of the original subject of the file.
ICMS	**Commissioned**	The name of the person or organization that commissioned the subject of the file.
ICMT	**Comments**	General comments about the file or the subject of the file. If the comment is several sentences long, end each sentence with a period. Do not include new-line characters.
ICOP	**Copyright**	Copyright information for the file. For example, © **Copyright 2009 Sony Creative Software Inc**. If there are multiple copyrights, separate them with a semicolon followed by a space.
ICRD	**Creation Date**	The date the subject of the file was created. List dates in year-month-day format, padding one-digit months and days with a zero on the left. For example, **1964-03-02** for March 2, 1964.
ICRP	**Cropped**	Describes whether an image or sound has been cropped and, if so, how it was cropped. For example, **Third movement, first through fourth bars**.
IDIM	**Dimensions**	The size of the original subject of the file. For example, **8.5 in h, 11 in w**.
IDPI	**Dots Per Inch**	The dots-per-inch setting of the digitizer used to produce the file.
IENG	**Engineer**	The name of the engineer who worked on the file. If there are multiple engineers, separate the names by a semicolon and a blank: **Engineer, Joe; Mixer, Matt**.
IGNR	**Genre**	Describes the classification of the original work.
IKEY	**Keywords**	Separate multiple keywords with a semicolon and a blank: **Madison; aerial view; scenery**.
ILGT	**Lightness**	Describes the changes in lightness settings on the digitizer required to produce the file. The format of this information depends on hardware used.
IMED	**Medium**	Describes the format of the original subject of the file.
INAM	**Name/Title (CD Text)**	The title of the subject of the file, such as **Madison From Above**.
IPLT	**Palette Setting**	The number of colors requested when digitizing an image.
IPRD	**Product**	The name of the title the file was originally intended for, such as **Encyclopedia of Midwest Geography**.
ISBJ	**Subject**	Describes the contents of the file, such as **Aerial view of Madison**.

FourCC Code	Name	Description
ISFT	**Software**	The name of the software package used to create the file.
ISHP	**Sharpness**	Identifies the changes in sharpness for the digitizer required to produce the file. The format of this information depends on the hardware used.
ISRC	**Source/Album**	The name of the person or organization who supplied the original subject of the file.
ISRF	**Source Form**	The original form of the material that was digitized, such as slide, paper, map, and so forth. This is not necessarily the same as IMED.
ITCH	**Technician**	The technician who digitized the file.
DISP	**Sound Scheme Title**	Sets the title that is displayed for Microsoft Sound Systems.
TLEN	**Text Length (ms)**	The length of the file in milliseconds.
TRCK	**Track Number**	The track number of the media from the original source media.
TURL	**URL**	The Web address associated with the file.
TVER	**Version**	Sets the version of the file. You can use versioning information to keep track of multiple mixes.
LOCA	**Location**	Identifies the location where the file was recorded.
TORG	**Organization**	Identifies the organization that produced the track.

Chapter 4

Getting Started

The Sound Forge® Pro digital audio editing tool is for users from all musical backgrounds. It is an extremely deep program, containing features that may only be required by the most advanced or specialized users. Nonetheless, a firm grasp of Sound Forge basics is essential. This chapter is designed to provide you with information on Sound Forge fundamentals.

Creating a project

You can use Sound Forge project files to organize and work with your media files nondestructively. When you save a project file, two things are created: a .frg file and a subfolder that contains your media file and all of the temporary files created while working on your project. The .frg file is not a multimedia file, but is used to render the final file after editing is finished. When you copy, cut, paste, and otherwise edit your project, the process is nondestructive—meaning you can edit without worrying about corrupting your source files. Within the project file, you can also undo any past operations, including those occurring before your last save. When you are finished working with a project file, you can save your work to a media file using the **Render As** option on the **File** menu.

> **Note:** *To use the advanced undo/redo capabilities mentioned above, you must have the* **Allow Undo past Save** *check box selected on the* **General** *tab of the Preferences dialog. To access the Preferences dialog, choose* **Preferences** *from the* **Options** *menu.*

1. From the **File** menu, choose **Save As** to save the current data window to a project file. The Save As dialog appears.
2. Using the **Save in** drop-down list, locate the folder where you want to save the project.
3. From the **Save as type** drop-down list, choose **Sound Forge Pro Project File (*.frg)**.
4. In the **File name** box, type a name for the file.
5. Click the **Save** button. A .frg file is created with the name you specified, and a folder with a similar name (**projectname_frg**, for example) is created in the same location for the temporary files.

> **Important:** *The associated project folder created by this process should not be deleted, as this will cause your project file to be unusable.*

Getting media files

The software can open a variety of audio and video files. There are two main methods for locating, previewing, and opening media files:

- From the **File** menu, choose **Open** to display the Open dialog.
- From the **View** menu, choose **Explorer** to display the Explorer window.

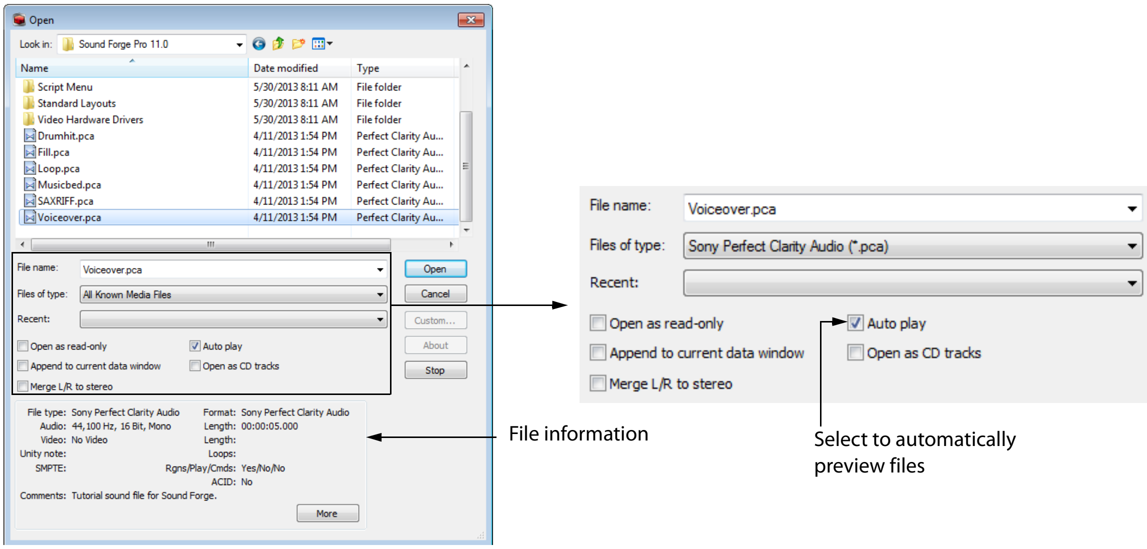
These methods are explained in greater detail in the following sections.

> **Note:** *To have pulldown fields automatically removed when opening 24 fps progressive-scan DV video files, select the* **Allow pulldown removal when opening 24p DV** *check box on the* **Video** *tab of the Preferences dialog. To open your 24p DV video files as 29.97 fps interlaced video (60i), clear this check box.*

Using the Open dialog

1. From the **File** menu, choose **Open**. The Open dialog appears.

> **Tip:** *You can also click the* **Open** *button ( ) on the Standard toolbar or press Ctrl+O.*



The Open dialog contains several features that allow you to locate and open audio files. These features are detailed below.

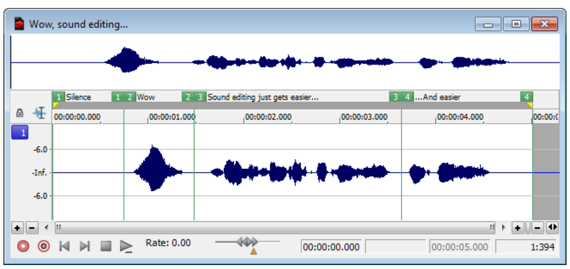
Feature	Description
Files of type	Use this drop-down list to specify the file format displayed in the system. A variety of file formats are supported. **Tip:** *Choose the* **CD Audio (*.cda)** *option from this list to extract audio tracks from a CD.*
Recent	Use this drop-down list to locate recently accessed folders.
Open as read-only	Select this check box if you want to open sound files but you do not want to alter the data in the files. This feature is useful if you only need to play the file or copy sections from the file. You can still change the Regions List, Playlist, and summary information for the file, but these changes must be saved to a new file.
Auto play	Select this check box to automatically preview files as you select them in the Open dialog.
Append to current data window	If you have multiple files selected, you can select the **Append to current data window** check box to add the selected files to the end of the current data window. **Note:** *When appending files to a data window, be sure to use files with matching formats, or perform any required sample rate or bit-depth conversion before appending the files. If the formats of the selected files do not match, they will play back at the target format without any conversion. If the sample rates do not match, for example, pitch will not be preserved.* When the check box is not selected, multiple files will be opened to separate data windows.

Feature	Description
Open as CD tracks	Select this check box if you want to create a disc-at-once CD track when opening a file. If you have multiple files selected, you can select the **Append to current data window** check box to add the selected files to the end of the current data window and create a disc-at-once CD track for each file. When the check box is not selected, multiple files will be opened to separate data windows. **Note:** *Disc-at-once CD tracks must be at least four seconds long. If you select a file that is less than four seconds long, silence will be added as needed.*
Merge L/R to stereo	Select this check box and hold the Ctrl button while selecting two mono files in the browse window. The two mono files will be merged to the left and right channels of a new stereo file. The first file you select will be placed in the left channel, and the second file will be placed in the right channel. Compressed files are not supported for merging.

2. Locate and select a media file using the **Look in** drop-down list at the top of the dialog.
3. To preview the file before adding it to your project, click the **Play** button.

Note: *If you have the* **Auto play** *check box selected, your file will automatically begin previewing when you select it.*

4. Click **Open**. The file is opened and a data window containing the waveform appears.



Using the Explorer window

In addition to using the Explorer window for locating, previewing, and opening media, you can drag files or regions from the Explorer window to an open data window to paste or mix the data. Click the right mouse button while dragging to toggle mix, paste, and CD track drag-and-drop modes. You can also extract audio from a CD.

Previewing media

The Explorer window allows you to easily preview files before you open them. The Explorer window has a mini-transport bar with **Start Preview**, **Stop Preview**, and **Auto Preview** buttons ( ). When you preview a file, its stream is sent to the channel meters on the main workspace (for audio files) or to the Video Preview window (for video files).

Note: *To preview video files, you must have the Video Preview window open. To display the Video Preview window, choose* **Video Preview** *from the* **View** *menu.*

1. Select a file in the Explorer window.
2. Click the **Start Preview** button ( ) to listen to the file.
3. Click the **Stop Preview** button ( ) or select a different file to stop previewing the file.

Tip: *To automatically preview selected files, click the* **Auto Preview** *button ( ) on the Explorer window's transport bar.*

Opening media

To open a media file into a new data window from the Explorer window, double-click the file. To open a media file in a specific data window, drag the media file from the Explorer window to the data window.

Using the Favorites folder

Select the Favorites folder (icon) or choose **Favorites** from the Address Bar to view the contents of the Favorites folder. This folder contains shortcuts to folders that you use often.

> **Tip:** *Favorites are saved in* **C:\Users\<user name>\AppData\Roaming\Sony\Sound Forge Pro\11.0\ExplorerFavorites.txt**.
>
> *The file is saved whenever you close the Explorer window or exit the application. You can copy the file to different computers or user accounts to migrate Favorites settings.*
>
> *To see this file, you must have the* **Show hidden files and folders** *radio button selected on the* **View** *tab of the Folder Options Control Panel in Windows.*

Adding a folder to the Favorites folder

1. Browse to the folder you want to add.
2. Right-click the folder and choose **Add Folder to My Favorites** from the shortcut menu to create a shortcut to the folder.

Removing a folder from the Favorites folder

1. Select the Favorites folder.
2. Right-click the folder you want to delete and choose **Delete** from the shortcut menu.

> **Note:** *Deleting a folder from Favorites deletes only the shortcut to the folder; the target folder is unaffected.*

Obtaining or editing CD information

If Sound Forge can access information about a track or CD (either from the file or CD itself or from a local cache), it automatically reads and displays this information when you insert a CD or browse your computer. However, if this information is not available, the software can retrieve information over the Internet from Gracenote® MusicID™.

Once Sound Forge obtains information from Gracenote MusicID, it is saved to a local cache so the information appears more quickly the next time the tracks are displayed.

If the software cannot connect to the Gracenote Media Database and the appropriate CD information is not available on your computer, the tracks are simply listed numerically. In this case, you can edit CD information and submit it to the Gracenote Media Database.

> **Notes:**
>
> - *Using Gracenote MusicID requires an active Internet connection.*
> - *For more information on using Gracenote MusicID, refer to the Gracenote Web site at http://www.gracenote.com/company_info/FAQ/FAQs/.*

Locating matching CD information using Gracenote

1. Insert a CD in your drive.
2. Browse to the CD and click the **MusicID** button ( ) in the Explorer window.

   Gracenote MusicID attempts to obtain matching CD information and displays artist, album, and track data:

   - If the service locates an exact match, this information automatically appears. No additional action is necessary.
   - If the service locates multiple possible matches, the Match dialog appears. Proceed to step 3.
3. Choose a method for completing the CD information:
   - If none of the possible matches is appropriate, click the **Submit New** button. The Gracenote MusicID Disc Information dialog appears, allowing you to complete information for the CD and submit it for inclusion in the Gracenote Media Database. For help on submitting CD information, click the **Help/Guidelines** button in this dialog.

     When you are finished typing information, click the **OK** button to submit your data.
   - Select the appropriate match from the list and click the **Accept Match** button. The artist, album, and track information is displayed based on your selection in the right side of the PC pane.

Editing and submitting CD information to Gracenote

If a CD is not currently part of the Gracenote Media Database, you can submit it for inclusion.

1. Insert a CD in your drive.
2. Browse to the CD and click the **MusicID** button ( ) in the Explorer window. The Gracenote MusicID Disc Information dialog appears.
3. Use the Gracenote MusicID Disc Information dialog to edit information about the CD. For help on submitting CD information, click the **Help/Guidelines** button in this dialog.
4. When you are finished entering the information, click the **OK** button to submit it for inclusion in the Gracenote database.

Extracting audio from CDs

The Explorer window allows you to easily extract audio from a CD into a data window. Each audio track on the CD is extracted into a separate data window.

1. Use the Explorer window to browse to and select your CD drive. The CD's audio tracks appear in the right pane of the Explorer window.
2. Select the tracks you want to extract.
3. Drag the tracks to the main Sound Forge workspace. The software begins extracting the selected tracks into individual data windows.
4. To stop the extraction process, you can click the **Cancel** button on the status bar to stop the whole process or on the individual data windows to stop extracting a specific track.

Tip: *To extract a single audio track into a new data window, double-click the track in the right pane of the Explorer window.*

Using Explorer views

You can control the information that appears in the Explorer window by clicking the **Views** button ( ) and selecting a view. These options are explained below:

Item	Description
Tree View	Displays all of the available drives and folders that you can choose from to find files.
Region View	Displays any regions that have been defined in the selected media file.
Summary View	Displays a short description of the selected media file at the bottom of the Explorer window.
Details	Displays the file size, date, and when the file was last created or last modified.
All Files	Displays all file types in the active folder.

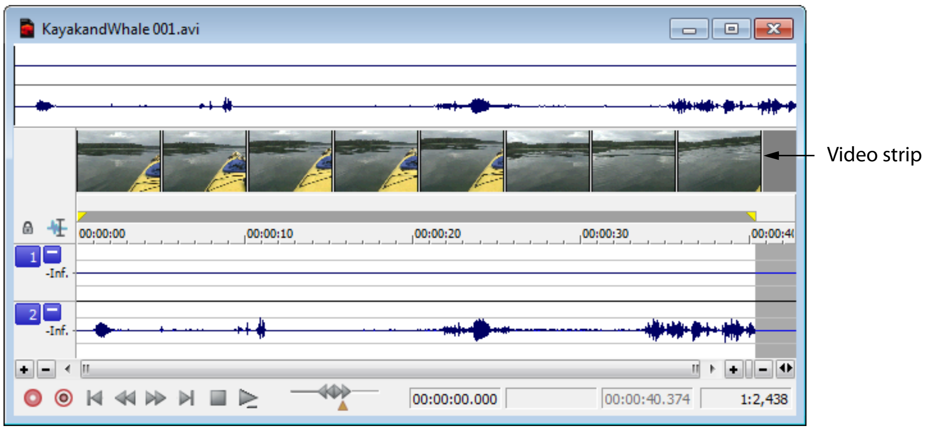
Peak files

When you first open a file, the entire file is scanned and a peak file is created. The peak file is stored with the same name and in the same location as the audio file, but it is given an .sfk extension. This peak file is automatically updated whenever the original file is edited.

Working with video files

The Sound Forge application has the ability to open and save many video file formats. The video files cannot be edited within the software, but this functionality allows you to attach, detach, and edit audio for the video. Once you've edited the audio, you can preview the audio and video together.

When you open a media file containing video, the data window displays the video portion in a video strip above the audio.



For more information, see [*Working with Video*](#) *on page* [*289*](#).

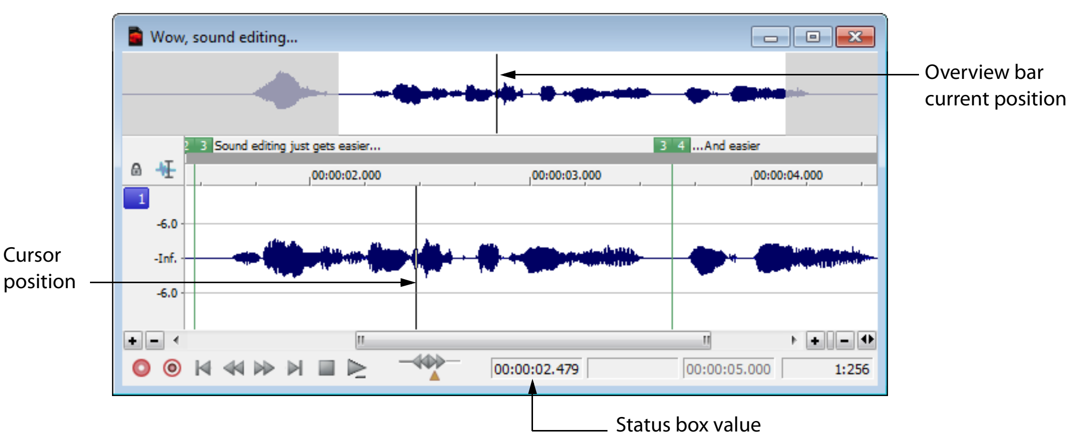
Playing a file

After you open a file, you can play it by clicking the **Play All** button ( ) on the transport bar. *For more information, see Transport toolbar on page 41.*

Viewing the current position

As a file plays, the current playback position is indicated in the data window in three ways:

- A cursor travels across the visible portion of the data window.
- The current playback position in relation to the entire file appears in the overview bar.
- The first selection status box in the playbar displays the current position in the user-specified format. *For more information, see Selecting status formats on page 87.*



Data window scrolling during playback

From the **Options** menu, choose **Scroll Playback** (or press F6) to enable automatic data window scrolling during playback. When the cursor moves off of the current window, it will quickly scroll to show another full window of data.

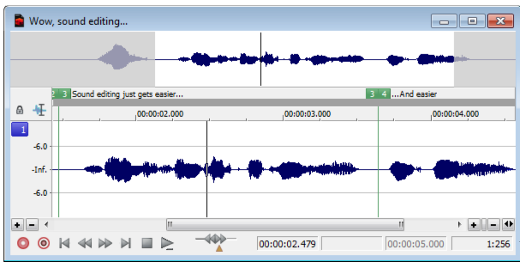
To enable smooth scrolling, select the **Scroll Smoothly** option from the **Options** menu (or press Shift+F6). When this option is selected, the cursor will slowly move back to the center of the display, and the wave data will scroll past it. This allows you to view upcoming data while the file is being played.

Playing a file from a specified point

You can begin playback from any point in a file.

1. Click to position the cursor in the data window. A flashing cursor (spanning the height of the waveform display) is displayed.
2. Click the **Play** button ( ) on the transport bar. The file plays from the cursor position.

   If you do not hear playback, you may have inadvertently created a small selection. To determine if you created a selection, examine the status boxes in the bottom-right corner of the data window.

   - If only the first box contains a value, there is no selection.
   - If all three boxes contain values, a selection has been created. Clear the selection by clicking anywhere in the data window.

For more information, see Viewing selection status on page 68.

> **Tip:** *When* **Options** > **Seek Cursor on Playback** *is selected, playback will restart when you position the cursor. If you do not want to interrupt playback when positioning the cursor, clear this command.*

Playing in Loop Playback mode

You can play an entire file or a selection in Loop Playback mode. In Loop Playback mode, the audio is played in a continuous loop.

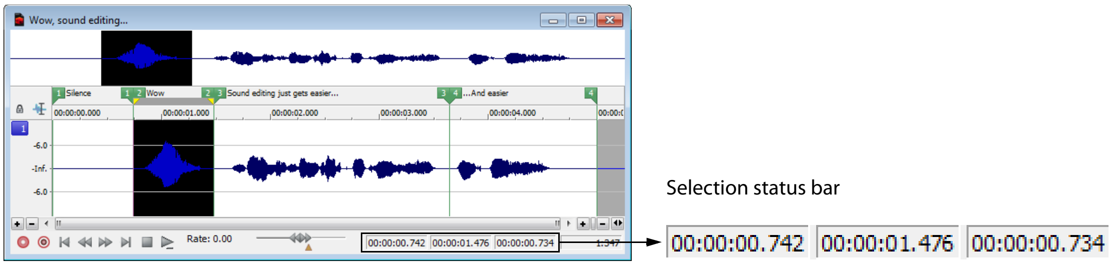
Click the **Loop Playback** button ( ) on the transport bar (or press Q) to turn Loop Playback mode on and off.

> **Tip:** *When* **Options** > **Seek Cursor on Playback** *is selected, playback will restart when you position the cursor. If you do not want to interrupt playback when positioning the cursor, clear this command.*

Playing a selection

You can play specific portions of audio data by creating selections in the waveform display.

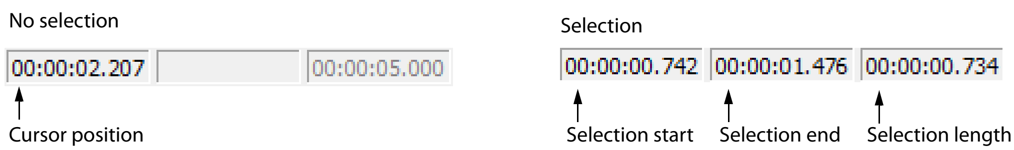
1. Drag the mouse within the data window. Notice that the waveform is selected as the mouse is dragged.
2. Click the **Play** button (▶). Only the selection plays.



Tip: *When* **Options** > **Seek Cursor on Playback** *is selected, playback will restart when you position the cursor. If you do not want to interrupt playback when positioning the cursor, clear this command.*

Viewing selection status

When a selection exists, the boxes in the selection status bar in the bottom-right corner of the data window contain values. These values indicate the start, end, and length of the selection. Double-click a box to edit the value.

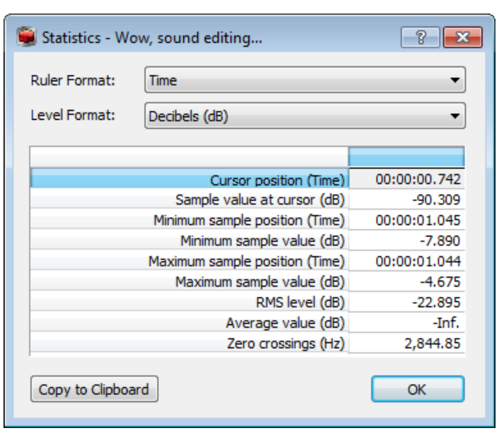


Selecting the status format

You can display status values in any supported format. You can change the format by right-clicking and choosing a new format from the shortcut menu. *For more information, see Selecting status formats on page 87*.

Viewing selection statistics

Choosing **Statistics** from the **Tools** menu displays a Statistics window showing information about the current selection or, if there is no selection, on the entire file.



The following table describes all statistical categories displayed in the Statistics window.

Statistical Category	Description
Ruler Format	Choose a setting from the drop-down list to determine the format you would like to use for the **Cursor position**, **Minimum sample position**, and **Maximum sample position** categories. *For more information, see Selecting status formats on page 87*.
Level Format	Choose a setting from the drop-down list to specify how the left- and right-channel levels at the cursor position will appear. • **Values** – Appears as an integer. The range is from -8388608 to 8388607 in 24-bit audio, -32768 to 32767 in 16-bit audio and -128 and 127 in 8-bit audio. • **Decibels** – Appears as decibels. A value of 0 dB corresponds to maximum absolute amplitude and negative infinity (-Inf.) corresponds to complete silence. In 16-bit audio, -90.3 dB is the lowest possible dB value (sample value of 1). • **Percentages** – Appears as a percentage ranging from -100 to 100 percent.
Cursor position	The cursor position (in samples) from the start of the audio file.
Sample value at cursor	The actual number stored by a single sample. The maximum allowed sample value is often referred to as 100% or 0 dB.
Maximum/minimum sample position and sample value	The maximum and minimum sample values and the locations (in samples) where they occur. These values may help determine if clipping will occur in the audio file. These values can also be used to determine the noise level of a signal for use with the Noise Gate effect (a built-in XFX plug-in installed with Sound Forge). For example, to determine the noise amplitude of a file, run Statistics on a region of noisy silence.
RMS level	The Root Mean Square of the sample values relative to the RMS value of a maximum-amplitude square wave (the loudest possible recording). On short intervals, this value relates to the volume level of the audio file. If used on a large selection with large volume variation, this value becomes less meaningful.
Average value	The sum of all sample values in the selected region divided by the number of samples. If this value is not zero, it usually indicates a DC offset in the recording process.
Zero crossings	The number of times per second that the waveform fluctuates from a negative to a positive value. This value can be used as a rough estimate of the frequency of the audio data for very simple waveforms.
Maximum true peak sample value and position	The maximum true peak sample value and the location where it occurs. This value displays the peak level in dB FS. Please note that true peaks are calculated using a higher sample rate than the **Maximum sample position** and **Maximum sample value** items for increased accuracy.
Maximum filtered true peak sample value and position	The maximum filtered true peak sample value and the location where it occurs. This value displays the peak level in dB FS. Peak levels may be miscalculated if audio signals are asymmetrical or if a DC offset is present. Filtered true peaks are calculated as the maximum of the filtered and unfiltered signals.

<table>
<tr><th>Statistical Category</th><th>Description</th></tr>
<tr><td>Loudness</td><td>Displays loudness statistics about an audio file's momentary loudness, short-term loudness, integrated (overall) loudness, and loudness range. You can use these values when mastering for broadcast to ensure compliance with loudness standards (such as the CALM Act).

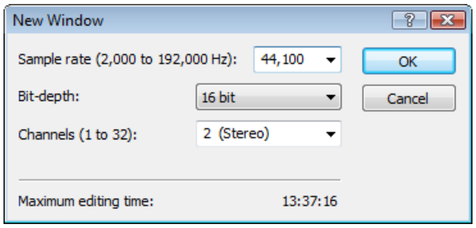
The following values are calculated:
• The Integrated value represents the integrated loudness — in loudness units full scale (LUFS) — across all audio channels over the duration of the program.
• The Loudness Range value represents the loudness range — in loudness units (LU) — of the momentary and short-term levels. The Loudness Range measurement provides a standardized method of determining the dynamic range of the signal.
• The Maximum Short-Term value represents the maximum short-term loudness — in loudness units full scale — across all audio channels based on 3-second integration windows.
• The Maximum Momentary value represents the maximum momentary loudness — in loudness units full scale — across all audio channels based on 400-millisecond integration windows.

Tip: Select the Enable surround processing for files with 6 channels check box on the Status tab of the Preferences dialog if you want to treat audio with six or more channels as surround audio when measuring loudness (a gain of ~1.5 dB is applied to the left and right surround channels). When the check box is cleared, all channels contribute equally to the loudness measurement.</td></tr>
<tr><td>Copy to Clipboard</td><td>Copies all contents of the Statistics window to the clipboard. This can be useful if you want to compare statistics of multiple files in a spreadsheet.

Tip: To copy specific data or cells, select the cells that you want to copy and press Ctrl+C.</td></tr>
</table>

Creating data windows

1. From the **File** menu, choose **New**. The New Window dialog appears.



2. Complete the New Window dialog:
   a. From the **Sample rate** drop-down list, choose a sample rate.
   b. From the **Bit-depth** drop-down list, choose a bit depth.
   c. Choose a setting from the **Channels** drop-down list to select the number of channels stored in the file.

   For more information, see Editing file properties on page 105.
3. Click **OK**. A new data window with the specified attributes appears.

Tip: *New windows are automatically named for you. You can customize this automatic naming feature to suit your needs. For more information, see Editing default data window names on page 336.*

Active data windows vs. inactive data windows

When multiple data windows are displayed on the workspace, only the window currently being edited is active, and all operations affect this window exclusively.

Activating a window

To activate a data window, click anywhere within it. The title bar changes to the color defined as the active window color and the previously active window is deactivated.

Note: *Choosing* **Focus to Data Window** *from the* **View** *menu also results in the focus being returned to the current data window.*

Copying data to a new file

You can create new audio files by copying data to a new data window.

1. Open an audio file and create a selection.
2. From the **Edit** menu, choose **Copy**, or click the **Copy** button ( ). The selection is copied to the clipboard.
3. Create a new data window. *For more information, see Creating data windows on page 71.*
4. From the **Edit** menu, choose **Paste,** or click the **Paste** button ( ). The selected data is pasted in the new data window.

Working with files

You can save a file in a variety of formats, including popular audio formats such as WAV and AIFF, and streaming media formats such as Windows Media®. You can save a file using a standard template, or you can customize the settings to suit your needs. Once you create custom settings, you can save those settings as a template.

You have the option to save all open files at once or to save all open files as a workspace file.

Saving a file

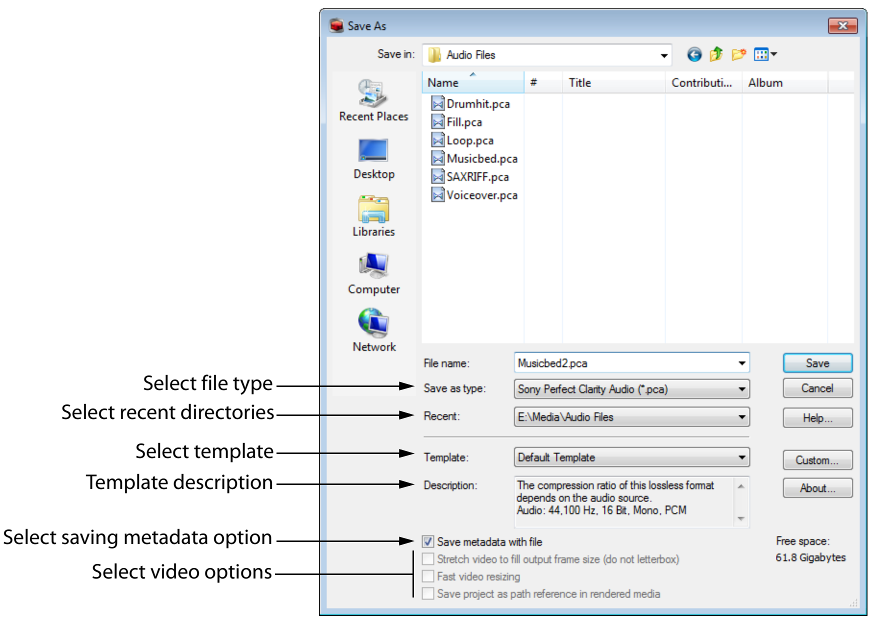
1. Click anywhere in the data window to select it.
2. From the **File** menu, choose **Save**.

> **Note:** *When saving a new file, the Save As dialog appears. If the file was previously saved, choosing* **Save** *automatically saves the file without your input.*

Using the Save As/Render As dialog

The Save As dialog allows you to save an audio file with a new name, in an alternate format, or with new attributes.

The Render As dialog allows you to render a file using a standard template, or you can customize the settings to suit your needs. Once you create custom settings, you can save those settings as a template.



1. Click anywhere in the data window to select it.
2. From the **File** menu, choose **Save As** to display the Save As dialog.

   If you're working with a Sound Forge project file, you can use the Save As dialog to save your project to a different name or location. Choose **Render As** to save your project as a media file.
3. Select the folder where you want to save the file:
   - From the **Save in** drop-down list, choose a drive and folder.
   - From the **Recent** drop-down list, choose a folder where you have previously saved files.
4. Type a name in the **File name** box, or select a file in the browse window to replace the existing file.
5. Choose a file type from the **Save as type** drop-down list.

   If you know that the file format is unsupported, choose **Raw Audio** and click the **Custom** button to display the Custom Template dialog, where you can specify format parameters. *For more information, see Creating custom templates on page 74.*
6. Choose a setting from the **Template** drop-down list to choose the attributes that will be used to save your file, or click the **Custom** button to create a new template. *For more information, see Creating custom templates on page 74.*

Notes:

- *When you convert from mono to stereo, the data will be stored in both channels. When converting from stereo to mono, the data from both channels will be mixed to a single channel.*
- *When determining bit rates, 1K=1024.*

7. Select the **Save metadata with file** check box if you want to preserve metadata (such as embedded data from other applications, regions, markers, disc-at-once CD tracks, commands, playlist, and sampler information) in your file. If the check box is cleared, the data will be ignored.

Note: *If the selected file type doesn't support all the metadata in your file, you will be prompted to save the metadata to an external file with an .sfl extension (using the same base name as your media file). Metadata can be saved internally for the following file formats:*

- *MP3*
- *PCA*
- *SFA*
- *WAV*
- *WAV64*
- *Windows Media Format (WMA and WMV)*

8. Select the **Stretch video to fill output frame (do not letterbox)** check box if you are saving to a format with a different aspect ratio than your source media settings. When this check box is cleared, black bars may appear at the top and bottom (letterboxing) or sides (pillarboxing) of the frame to preserve the aspect ratio.
9. Clear the **Fast video resizing** check box if you see unacceptable video artifacts in the rendered video (these artifacts are most obvious with MPEG and streaming formats). Turning this option off can correct the artifacts, but your rendering times will increase significantly.
10. Select the **Save project as path reference in rendered media** check box if you want to save the path to your Sound Forge project in the rendered file. Saving the project path allows you to easily return to the source project if you use your rendered file in another project.

Note: *The check box will be unavailable if you haven't saved your project or if you're rendering using a third-party file-format plug-in.*

11. Select the **Generate Loudness Log** check box if you want Sound Forge to analyze the loudness of your file and create a log file that summarizes its loudness values.

    The loudness log is created using the same folder and base name as your sound file with **_loud.txt** appended to the name.

    The log will record the file name, format, loudness metering mode, and loudness values throughout the file.

> **Important:** *Loudness logging is performed after the plug-in chain, but before any codec is applied to your rendered file. Because audio compression may affect audio levels, choose* **Tools** > **Generate Loudness Log** *to create a log after saving to a compressed format.*
>
> *Select the* **Enable surround processing for files with 6 channels** *check box on the Status tab of the Preferences dialog if you want to treat audio with six or more channels as surround audio when measuring loudness (a gain of ~1.5 dB is applied to the left and right surround channels). When the check box is cleared, all channels contribute equally to the loudness measurement.*

12. Click the **Save** button.

Creating custom templates

If the file type you select supports it, you can create custom settings for saving files by clicking the **Custom** button.

> **Note:** *If a file type supports custom templates, a* **Custom** *button appears next to the* **Template** *drop-down list after you choose the file type.*

When you click the **Custom** button, a Custom Settings dialog appears. Adjust the settings for the different template properties as needed. For help on the different settings, click the **Help** button ( ) or press Shift+F1.

When you are finished editing the template properties, click the **OK** button.

Saving custom templates

You can save a custom template to use again by typing a template name in the **Template** box and clicking the **Save Template** button ( ).

Deleting custom templates

You can delete a custom template by selecting the template from the **Template** drop-down list and clicking the **Delete Template** button ( ).

Creating custom rendering settings

The Custom Settings dialog appears when you click **Custom** in the Render As dialog. You can use the Custom Settings dialog to create custom encoding templates for many of the file formats available in the software.

1. From the **File** menu, choose **Render As**. The Render As dialog appears.
2. Choose your preferred file format from the **Save as type** drop-down list. If the format allows you to create custom settings, the **Custom** button becomes active.
3. Click **Custom**. The Custom Settings dialog appears.
4. Make the appropriate setting changes for the chosen file format. For help on individual settings, click the **Help** button ( ).

> **Tip:** *To save the custom settings for future use, type a name for the template in the* **Template** *box and click the* **Save Template** *button ( ).*

5. Click **OK**. The Custom Settings dialog closes.

Copy rendering templates between computers or user accounts

You can make your customized rendering templates available on another computer or user account by copying .sft files to the appropriate location in the new account or computer.

Rendering templates are stored in **C:\Users\<user name>\AppData\Roaming\Sony\Render Templates\<plug-in name>**.

Notes:

- *The AppData folder is not visible unless the* **Show hidden files and folders** *radio button is selected on the View tab of the Windows Folder Options control panel.*
- *You can find a plug-in's name by clicking the* **About** *button in the Save As/Render As dialog.*

To make a template available on another computer or user account, copy the .sft file to the same location in another account.

For example, to make JSmith's custom wave template available for the AJones user account in Windows XP, copy the appropriate .sft2 file from this folder:

C:\Documents and Settings\JSmith\Application Data\Sony\Render Templates\wave\

to this folder:

C:\Documents and Settings\AJones\Application Data\Sony\Render Templates\wave\

Tip: *If you're copying templates from an older Sony Creative Software application, templates are saved as .sft files in the following folder: C:\Documents and Settings\<user name>\Application Data\Sony\File Templates\<plug-in name>\<plug-in GUID>.*

Saving all open audio files

Choosing **Save All** from the **File** menu automatically prompts you to save all open audio files on the current workspace.

Note: *Pressing Shift while choosing the* **Save All** *command automatically saves all open files without prompting you to approve each save.*

Saving files as a workspace

To accommodate complex editing scenarios, you can save the entire workspace as an alternative to saving individual files. Workspaces are saved as Sound Forge Workspace (.sfw) files. When you open a workspace file, all files are restored to their previous sizes, positions, and magnifications. In addition, each file's current cursor position, custom views, and plug-ins in the Plug-In Chain are restored. *For more information, see Creating and using views on page 104 and Using the Plug-In Chain on page 209*.

Saving the current workspace

1. From the **File** menu, choose **Workspace**, and choose **Save As** from the submenu. The Save Workspace dialog appears.
2. Browse to the folder where the file will be saved.
3. Type a name for the file in the **File name** box and click **Save**.

Opening a workspace

1. From the **File** menu, choose **Workspace**, and choose **Open** from the submenu. The Open Workspace dialog appears.
2. Browse to the folder containing the desired .sfw file.
3. Select the desired file and click **Open**.

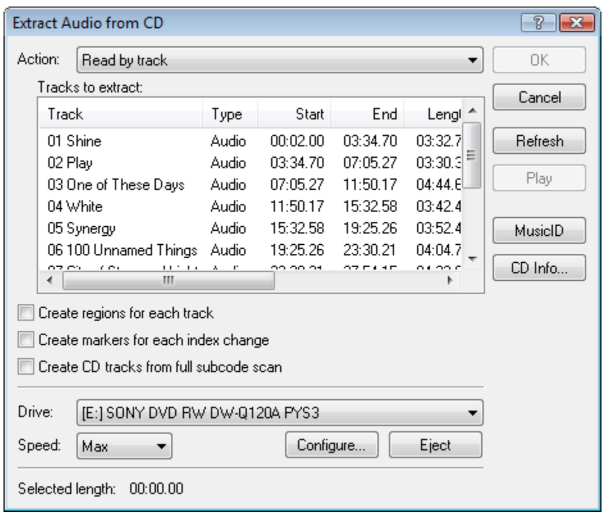
Extracting audio from CDs

You can extract data from CDs and open tracks in the Sound Forge workspace.

Tip: *Double-click a .cda file in the Explorer window (or drag it to the workspace) to extract a CD track without opening the Extract Audio from CD dialog. You can also extract audio from the Open dialog by choosing CD Audio (*.cda) from the* **Files of type** *drop-down list in the Open dialog.*

Important: *Sound Forge software is not intended for, and should not be used for, illegal or infringing purposes, such as the illegal copying or sharing of copyrighted materials. Using Sound Forge software for such purposes is, among other things, against United States and international copyright laws and contrary to the terms and conditions of the End User License Agreement. Such activity may be punishable by law and may also subject you to the breach remedies set forth in the End User License Agreement.*

1. Insert a CD in the drive.
2. From the **File** menu, choose **Extract Audio from CD**. The system's drives are identified. The Extract Audio from CD dialog is displayed. If the system is equipped with multiple CD-ROM or DVD-ROM drives, you must select the desired drive from the **Drive** drop-down list near the bottom of the dialog.



3. From the **Action** drop-down list, choose the method you want to use for extracting the CD audio:

Method	Description
Read by track	Use this option to select the tracks you want to extract from the CD. Each track is extracted into a unique data window.
Read entire disc	Use this option to automatically extract all tracks on the disc. The entire CD is extracted into a single data window.
Read by range	Use this option to extract audio from a specified range of time. Type appropriate values in the **Start** and **End** (or **Length**) boxes. The range of audio is extracted into a single data window.

4. If you choose **Read by track** or **Read by range** from the **Action** drop-down list, select the tracks or time range you want to extract.

Note: *Click* **Play** *to preview your selection. During playback, the button changes to a* **Stop** *button.*

5. Select extraction options as needed:
   - Select the **Create regions for each track** check box to add each extracted track to the file's Regions List.
   - Select the **Create markers for each index change** check box to place markers in the extracted file at all points where indices occur in the original track.

6. Select the **Create CD tracks from full subcode scan** check box if you want to create a disc-at-once CD track for each extracted track.

> **Note:** *When you select the* **Create CD tracks from full subcode scan** *check box, the software will create a disc-at-once track list based on the PQ data on the disc.*
>
> *ISRC data is added to each track if the data exists on the disc. Universal product code/media catalog number information is updated in the CD Information window.*

7. From the **Drive** drop-down list, choose the CD drive that contains the CD from which you want to extract audio.
8. Click the **MusicID** button if you want to obtain CD information using Gracenote MusicID.

   If CD information is not available, you can click the **CD Info** button to display a dialog box where you can edit the CD information and submit it for inclusion in the Gracenote Media Database. *For more information, see Obtaining or editing CD information on page 64*.
9. From the **Speed** drop-down list, choose the rate at which you want to extract audio. If you experience gapping or glitching, decrease the speed or click **Configure** and adjust the **Audio extract optimization** setting.

> **Note:** *To eject the CD at any time prior to beginning the extraction process, click the* **Eject** *button.*

10. Click **OK**. The data extraction from the CD begins, and a progress meter is displayed.

Previewing CD tracks

In the Extract Audio from CD dialog, select a track and click the **Play** button to preview a track prior to extracting it from the CD. To end the preview, click **Stop**.

Refreshing the Extract Audio from CD dialog

Click the **Refresh** button after you insert a new CD in the system's CD or DVD drive. This allows you to view the contents of the new CD without closing and reopening the Extract Audio from CD dialog.

Working with projects

Projects are new to Sound Forge software; however, if you've used ACID or Vegas software, then you'll be quite familiar with how to use Sound Forge projects. You should note that Sound Forge projects do function slightly different than ACID and Vegas projects.

A project file is not a multimedia file. It contains pointers to the original source files, so you can edit your project nondestructively without changing your source files. When you edit a Sound Forge project, you can undo edit operations even past your last save. *For more information, see Using Undo and Redo on page 85*.

Saving projects

1. Click anywhere in the data window to activate it.
2. From the **File** menu, choose **Save As** to display the Save As dialog.
3. Select the folder where you want to save the file from one of the following locations:
   - From the **Save in** drop-down list, choose a drive and folder.
   - From the **Recent** drop-down list, choose a folder where you have previously saved files.
4. In the **File name** box, type a name for the file or select a file in the browse window to replace the existing file.
5. From the **Save as type** drop-down list, choose **Sound Forge Pro Project File (*.frg)**. Sound Forge software creates a .frg file in the folder you specified and creates a subfolder to store your sound and temporary files.

> **Note:** *Because a Sound Forge project contains all your original sound data plus all PCM temporary files, they can take some time to create.*

Warning: *Deleting a project's [filename]_frg folder will break the project.*

Saving the project path in the rendered file

1. Save your Sound Forge project. The project must be saved before you can embed the project reference in the rendered file.
2. Follow the steps in *Using the Save As/Render As dialog on page 72* to choose the file type and location for rendering your files and then select the **Save project as path reference in rendered media** check box.

Note: *This check box will be unavailable if you did not save your project or if you are rendering using a third-party file format plug-in.*

3. Click **Yes** if you want to open the file in a new window or click **No** if you want to close the dialog and return to the Sound Forge window.

Note: *If you modify the project file after rendering, the project data will no longer match the rendered file. To edit a project using a path reference, the project file and all media must be available on your computer.*

Editing a media file's source project

When your Sound Forge project uses source media files that are rendered with an embedded project path reference, you can easily open the source project in the associated application if you need to edit the media. By saving your project path reference when you render files in ACID, Sound Forge, or Vegas, you can quickly access the media from Sound Forge via the **Edit Source Project** shortcut menu.

Note*: The project information in the rendered file is only a reference to a project file. If you modify the source project file after rendering, the project data will no longer match the rendered file. To edit a project using a path reference, the project file and all media must be available on your computer.*

1. Right-click one of the following items:
   - The waveform in a data window
   - A media file in the Explorer window
2. From the shortcut menu, choose **Edit Source Project**. An ACID, Vegas, or Sound Forge window will open with the source project.

   If you are editing a source project using a computer other than the computer where the project was created, then the editing computer must meet the following requirements:

   - The software that was used to create the project must be installed and the project file extension (.acd, .acd-zip, .veg, or .frg) must be registered on the editing computer.
   - The editing computer must have the same version (or later) of the software as the computer where the project was created.
   - The project file must exist on the editing computer using the same file path as on the computer where the project was created.
   - The project's source media must exist on the editing computer. If the media files do not use the same file path as on the computer where the project was created, you will be prompted to choose a new folder or replacement files.
3. Edit the project as necessary.
4. Render the edited project using the same name as the original media file and close the editing application.

Note*: If you are editing an existing track, your project will automatically be updated with the latest rendered media file.*

Editing audio

New Sound Forge users should remember that even the most complex editing is derived from a few simple operations: copy, paste, cut, delete (clear), trim/crop, and mix. The following table provides a brief description of the basic editing operations.

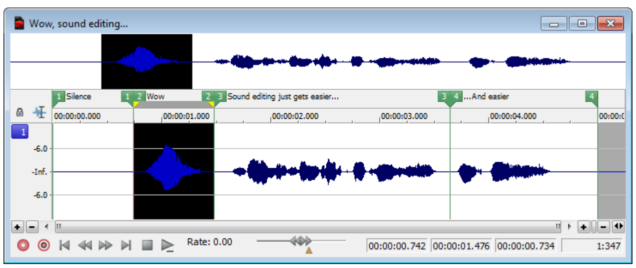
Editing Operation	Description
Copy	Copies data from the window to the clipboard.
Paste	Inserts the contents of the clipboard into the window at the current cursor position. If a selection exists in the data window, the pasted data replaces the current selection.
Cut	Deletes data from the window and copies it to the clipboard.
Delete (Clear)	Deletes data from the window, but does not copy it to the clipboard.
Trim/Crop	Deletes all data in the window with the exception of the selection.
Mix	Mixes data from the clipboard with the data in the current window, starting at the current cursor position or the start of the current selection.

Copying

You can copy audio data from a data window to the clipboard without changing the original file. Once audio data is on the clipboard, you can paste it into existing files or use it to create new files.

Copying data to the clipboard

1. Open the Voiceover.pca file. This file is located in the same folder as the application.
2. Create a selection containing "Wow."

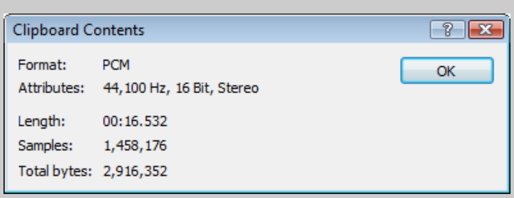


3. From the **Edit** menu, choose **Copy**, or click the **Copy** button ( ). The selected data is copied to the clipboard, but the waveform is unchanged.

Previewing clipboard contents

To preview the contents of the clipboard, choose **Clipboard** from the **View** menu, and choose **Play** from the submenu.

Tip: *You can view detailed information on the size and attributes of the clipboard contents in the Clipboard Contents window. From the* **View** *menu, choose* **Clipboard***, and then choose* **Contents** *from the submenu to display the Clipboard Contents window.*



Recycling clipboard contents

Once audio data is on the clipboard, you can paste or mix it into an infinite number of windows. Data remains on the clipboard until you replace it with new data.

Pasting

Once audio data is on the clipboard, you can paste or mix it into an existing data window or use it to create a new data window.

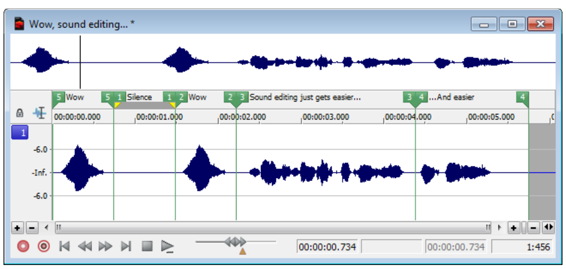
Pasting data in an existing data window

Notes:

- *Pasting into a multichannel file will insert data to all channels—the channels in a multichannel file must always be equal in length. Silence is pasted to the unselected channel. If multiple channels are selected, the same data is pasted to all selected channels, and silence is pasted to the unselected channel. If no channels are selected, the same data is pasted to all channels.*
- *Pasting data of different sample rates will cause the data in the clipboard to play at the same rate as the rate of the window in which the data is pasted.*
- *If any regions, markers, or loops are present in the original sound data, they will also be pasted into the destination sound file. To turn this feature off, turn off the* **Lock to Selection** > **Markers/Regions** *command on the Options menu.*

1. After you have cut or copied your data, move the cursor to the beginning of the Voiceover.pca file by clicking the **Go to Start** button ( ) in the playbar.
   - *For more information on cutting or copying data, see Copying on page 79* or *Cutting on page 81*.
   - *For more information on the playbar, see Playbar on page 35*.
2. From the **Edit** menu, choose **Paste**, or click the **Paste** button ( ). The clipboard data is inserted into the file and the data for “Wow” appears on the left side of the waveform.

Note: *If there is a selection, the* **Paste** *command deletes the selected data before inserting.*



3. To confirm that the data has been pasted into the file, click the **Play All** button ( ). “Wow. Wow. Sound editing just gets easier and easier” plays back.

Pasting by dragging and dropping a selection

1. Choose the Edit tool ( ).
2. Drag the mouse in the data window to create a selection anywhere in Voiceover.pca.

Tip: *If the* **Always open dropped files in new window** *check box on the* **General** *tab of the Preferences dialog is cleared, you can also hold Ctrl while dragging a file (or region) from the Explorer window to a data window to paste sound data. When the check box is selected, dropping a file on the Sound Forge workspace always creates a new data window.*

3. Hold Ctrl and drag the selection to the location where you want to paste the data. The cursor appears as a mouse pointer with the letter P ( ), and a vertical line appears to show you where the paste will occur.

   You can click the right mouse button while dragging to toggle mix, paste, and CD track drag-and-drop modes.
4. When you release the mouse button to drop the selection, the selection is pasted.

Pasting in a new data window

To use data from the clipboard to create a new data window, go to the **Edit** menu, choose **Paste Special,** and choose **Paste to New** from the submenu**.** A new window containing the clipboard data is created.

Cutting

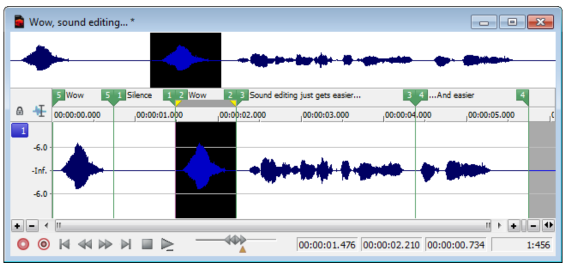
Cutting allows you to remove a section of sound data from a data window and store it on the clipboard until you paste or mix it into another file. Cutting sound data replaces the previous contents of the clipboard. When deciding between cut and copy, consider the following information:

- Copying data has no effect on the original file.
- Cutting data modifies the original file.

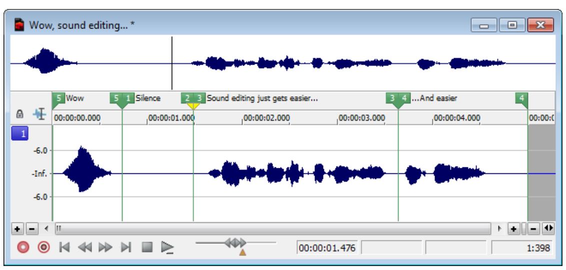
Cutting data from a window

> **Note:** *If you cut data from individual channels of multichannel files, the waveform will contain silence at the end of the cut channel. The channels in a multichannel file must always be equal in length.*

1. Create a selection containing the second "Wow" (there should be two if you are following the examples) in Voiceover.pca.



2. From the **Edit** menu, choose **Cut**, or click the **Cut** button ( ). The selected data is removed from the file and placed on the clipboard.



3. Click the **Play All** button ( ). "Wow. Sound editing just gets easier and easier" plays back.

Previewing a cut

You can preview cuts prior to performing the edit. This option allows you to determine if you made the selection accurately and if the results are desirable by playing the data before and after the current selection.

1. Create a selection anywhere in Voiceover.pca.
2. From the **Transport** menu, choose **Preview Cut/Cursor** (or press Ctrl+K). The selection is ignored and the audio before and after the selection is played to allow you to preview the cut.

Notes:

- *To set the amount of pre- and post-roll that will be played when you preview a cut, choose* **Preferences** *from the* **Options** *menu and choose the* **Previews** *tab. Type values in the* **Pre-roll** *and* **Post-roll** *boxes in the Cut preview configuration section of the dialog.*
- *If there is no selection, the playback will pre- and post-roll around the cursor position.*

Configuring cut pre-roll and post-roll lengths

Frequently, the default pre-roll and post-roll lengths are insufficient to evaluate the accuracy of an edit. For this reason, you can configure pre-roll and post-roll lengths.

1. From the **Options** menu, choose **Preferences**. The Preferences dialog appears.
2. Click the **Previews** tab.
3. Configure the **Pre-roll** and **Post-roll** values in the **Cut preview configuration** area of the dialog and click **OK**.

Deleting

Deleting a selection permanently removes it without placing it on the clipboard. To delete data, choose **Delete (Clear)** from the **Edit** menu (or press the Delete key).

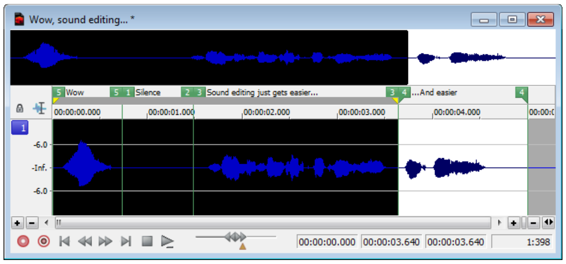
Notes:

- *If you delete data from individual channels of multichannel files, the waveform will contain silence at the end of the deleted channel. The channels in a multichannel file must always be equal in length.*
- *If the* **Treat as Cutlist** *command (available in the* **Edit** *menu,* **Playlist/Cutlist** *submenu) is selected, deleting a selection creates a region in the Cutlist window, but does not remove the selection. For more information, see* *Configuring the Playlist as a Cutlist on page 133.*

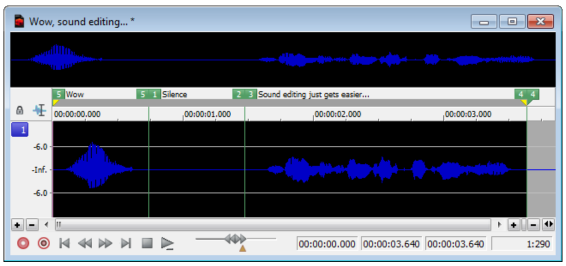
Trimming/Cropping

Trimming allows you to retain a selection while deleting all surrounding data.

1. Create a selection containing “Wow, sound editing just gets easier” in Voiceover.pca, but do not select the second “and easier.”



2. From the **Edit** menu, choose **Trim/Crop** (or press Ctrl+T). Only “Wow, sound editing just gets easier” remains in the data window.



Mixing

Mixing is a powerful editing function that allows you to mix a copy of the clipboard contents at the current cursor position.

Mixing by dragging and dropping a selection

1. Open and play the Drumhit.pca file. The file contains a snare drum and crash cymbal sound.
2. Choose the Edit tool ( ).
3. Drag the mouse over the data window to select the entire waveform.

> **Tip:** *If the* **Always open dropped files in new window** *check box on the* **General** *page of the Preferences dialog is cleared, you can also drag a file (or region) from the Explorer window to a data window to paste sound data. When the check box is selected, dropping a file on the Sound Forge workspace always creates a new data window. For more information, see General tab on page 331*.

4. Drag the selection to the beginning of the Voiceover.pca file. The cursor appears as a mouse pointer with the letter M ( ), and a shaded selection box appears to show you where the mix will occur. An envelope is drawn to show you the mix and fade levels. (The last-used settings from the Mix/Replace dialog are used by default.)

   You can click the right mouse button while dragging to toggle mix, paste, and CD track drag-and-drop modes.

> **Tip:** *If you want the Fade In and Fade Out curves to pay attention to the destination selection and file length when mixing between files, select the* **Auto-crossfade Mix with selection** *check box on the* **Editing** *tab of the Preferences dialog.*

5. When you release the mouse button to drop the selection, the Mix/Replace dialog appears. If you want to bypass the Mix/Replace dialog, hold Shift when you release the mouse button.

   To customize your mix settings, choose a setting from the **Preset** drop-down list in the Mix/Replace dialog, or adjust the controls as needed:

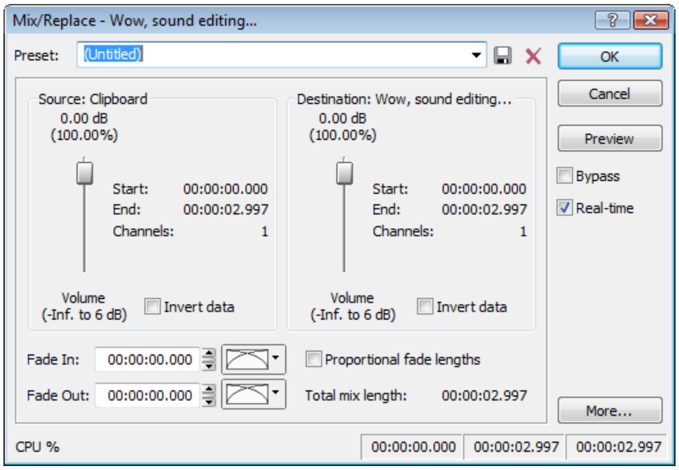
Item	Description
Source	Drag the **Source** fader to adjust the volume of the selection you want to mix. Changing this setting has the same effect as dragging the sustain portion of the wet gain envelope in the data window. Select the **Invert Data** check box to invert the source audio at the baseline (reverse the phase). Inverting data can help match transitions and compare the phase relationship of the two sound files.
Destination	Drag the **Destination** fader to adjust the volume of the selection you want to mix over. Changing this setting has the same effect as dragging the sustain portion of the dry gain envelope in the data window. Select the **Invert Data** check box to invert the destination audio at the baseline (reverse the phase). Inverting data can help match transitions and compare the phase relationship of the two sound files.
Fade In	Type a value in the **Fade In** box (or use the spinner) to set the length of the fade in between the source and destination audio. Changing this setting has the same effect as dragging the attack portion of the envelope in the data window. Click the **Fade Curves** button ( ) and choose a curve type from the menu to set the speed of the fade in.
Proportional Fade Lengths	Select the **Proportional fade lengths** check box if you want to specify fade lengths as a percentage of the selection.
Fade Out	Type a value in the **Fade Out** box (or use the spinner) to set the length of the fade out between the source and destination audio. Changing this setting has the same effect as dragging the attack portion of the envelope in the data window. Click the **Fade Curves** button ( ) and choose a curve type from the menu to set the speed of the fade out.
More	Click to display additional controls at the bottom of the dialog that you can use to change the selection you want to process.

6. Click the **OK** button to apply the mix.

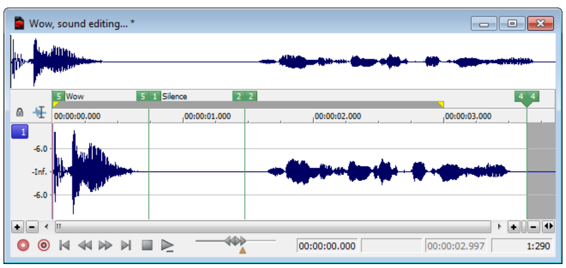
Mixing audio from the clipboard

1. Open and play the Drumhit.pca file. The file contains a snare drum and crash cymbal sound.
2. Verify that the Drumhit.pca window is active and choose **Select All** from the **Edit** menu, or press Ctrl+A. The entire waveform is selected.
3. From the **Edit** menu, choose **Copy**, or click the **Copy** button ( ).
4. Activate the Voiceover.pca data window and click the **Go To Start** button ( ) on the playbar. The cursor moves to the start of the file.

5. From the **Edit** menu, choose **Paste Special**, and choose **Mix** from the submenu, or click the **Mix** button ( ). The Mix/Replace dialog appears.



6. Verify that the **Source** and **Destination** volume faders are set to 0 dB and click **OK**. The drum hit is mixed equally with the spoken passage.



Preview the file and notice that mixing does not change the length of the file.

Using Undo and Redo

You can easily undo and redo edit operations, even prior to your last save operation.

- You can undo any edit operation by choosing **Undo** from the **Edit** menu, or by clicking the **Undo** button ( ) on the Standard toolbar.
- You can redo any undone edit operation by choosing **Redo** from the **Edit** menu, or by clicking the **Redo** button ( ) on the Standard toolbar.

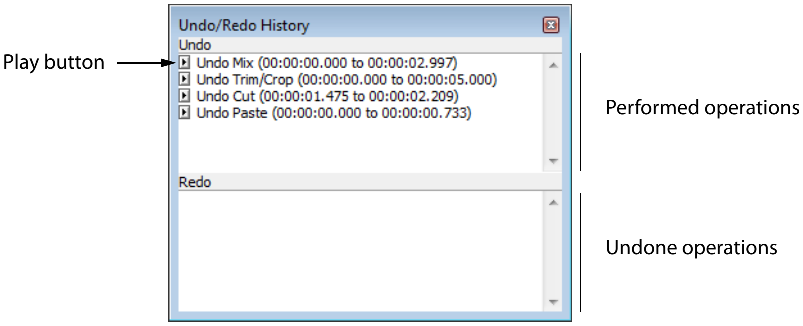
Important: *The ability to undo past save is disabled by default. To enable this functionality, choose* **Preferences** *from the* **Options** *menu, click the* **General** *tab, and select the* **Allow Undo past Save** *check box. When this option is enabled, your undo/redo history is retained until you close the file or exit the software.*

Using the Undo/Redo History window

The Undo/Redo History window may seem confusing at first, but you will find it invaluable once you have mastered it. This window allows the audio file to be auditioned in various versions by undoing and redoing multiple operations.

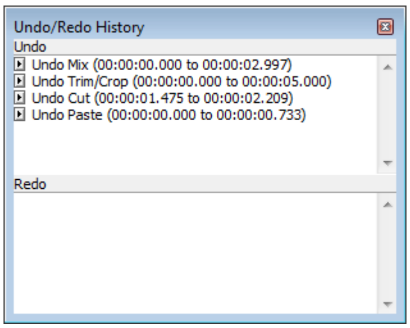
To display the Undo/Redo History window, choose **Undo/Redo History** from the **View** menu (or press Alt+7).

Note: *The undo/redo history for an audio file is retained until you close the file or exit the software. If you want to retain undo/redo history indefinitely, you should work with a Sound Forge project (.frg) file.*



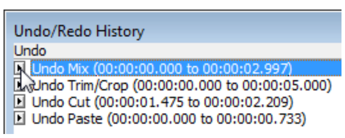
Undoing and redoing

1. Verify that the Voiceover.pca data window is active and choose **Undo/Redo History** from the **View** menu. The Undo/Redo History window appears. If you have performed the previous procedures, the window should look like the figure below:

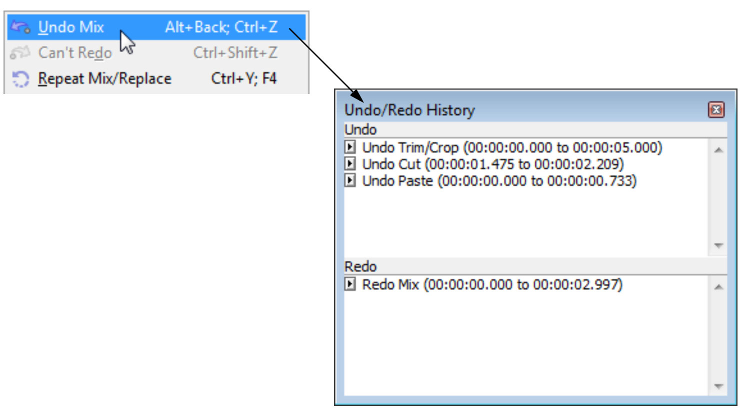


   Notice that the **Mix** operation appears at the top of the **Undo** pane. The most recent operations always appear at the top of the appropriate list.

   In the **Undo** pane, click the **Play** button ( ) corresponding to the **Mix** operation. The audio file plays without the drum track.

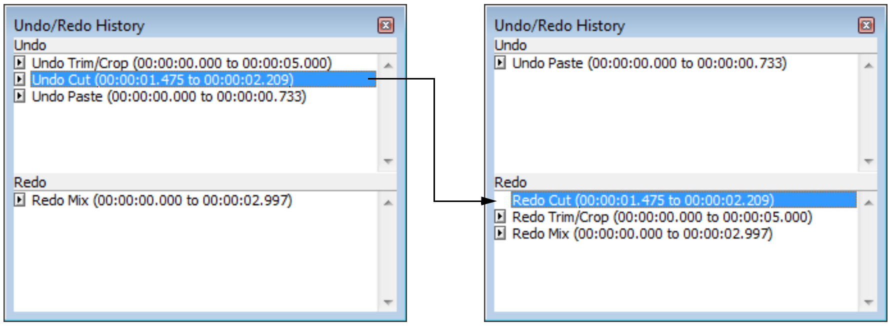


2. Select the **Mix** operation and choose **Undo** from the **Edit** menu. The drum track is extracted from the Voiceover.pca data window and the **Mix** operation moves to the **Redo** pane.



3. In the **Redo** pane, click the **Play** button ( ) corresponding to the **Mix** operation. The audio file plays with the mixed drum track.
4. Select the **Mix** operation again and choose **Redo** from the **Edit** menu. The drum track is remixed into the Voiceover.pca waveform and the **Mix** operation is returned to the **Undo** pane.
5. Select the **Trim/Crop** operation in the **Undo** pane and click the **Undo** button ( ). Only the **Mix** operation is undone and moved to the **Redo** pane. This is due to the fact that operations can only be undone or redone in the order originally performed.

6. Double-click the **Cut** operation in the **Undo** pane. The **Cut** and **Trim/Crop** operations are both undone in the waveform and moved to the **Redo** pane.



Tip: *To quickly undo and redo operations in the Undo/Redo History window, double-click the operation.*

Clearing the Undo/Redo History for the current file

Clearing the current file's Undo/Redo History frees up disk space by deleting the file's temporary undo/redo files. However, deleting these temporary files prevents you from undoing changes made to the file since it was last saved (or beyond, if you have the **Allow Undo past Save** check box enabled on the **General** tab of the Preferences dialog). To clear the undo/redo history for the current file, go to the **Edit** menu and choose **Clear Undo/Redo History**.

Note: *A file's undo/redo history is also automatically cleared when you close the file or exit the software.*

Selecting status formats

The status format determines how a file's position and length information is displayed. The following table briefly describes supported status formats (hh=hours, mm=minutes, ss=seconds, and ff=frames). *For more information, see SMPTE Timecode on page 361*.

Format name	Description	Format
Samples	Number of samples	Numbered (starting with zero)
Time	Hours, minutes, seconds, and milliseconds	hh:mm:ss.sss
Seconds	Seconds and fractions of seconds	sssss.sss (to three decimal places)
Time & Frames	Hours, minutes, seconds, and frames.	hh:mm:ss.ff
Absolute Frames	Frames and fractions of frames	Numbered (starting with zero, to three decimal places)
Measures & Beats	Measures, beats, and quarter beats	measures:beats.quarters
SMPTE Film Sync (24 fps)	SMPTE at 24 frames per second for synchronizing with film	hh:mm:ss:ff
SMPTE EBU (25 fps, Video)	SMPTE at 25 frames per second for European Broadcasting Union	hh:mm:ss:ff
SMPTE Non-Drop (29.97 fps, Video)	SMPTE at 29.97 frames per second	hh:mm:ss:ff
SMPTE Drop (29.97 fps, Video)	SMPTE at 29.97 frames per second using dropped frame numbers	hh:mm:ss:ff
SMPTE 30 (30 fps, Audio)	SMPTE at 30 frames per second	hh:mm:ss:ff
Audio CD Time	Hours, minutes, seconds, and frames with a frame rate of 75 frames per second for creating disc-at-once CDs.	hh:mm:ss:ff

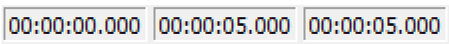
Experimenting with status formats

You can experiment with the Voiceover.pca file to see how status formats affect values in the selection status bar display boxes.

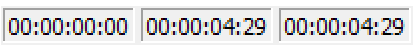
1. Open the Voiceover.pca file.
2. From the **Options** menu, choose **Status Format**, and choose **Samples** from the submenu.
3. Select all data in the Voiceover.pca window by choosing **Select All** from the **Edit** menu. Notice the selection status boxes.

   - The first selected sample is sample 0.
   - The last selected sample is 220,507.
   - The total number of samples in the selection is 220,507.

4. From the **Options** menu, choose **Status Format**, and choose **Time** from the submenu. Notice that status values change from samples to hours, minutes, and seconds.



5. From the **Options** menu, choose **Status Format**, and choose **SMPTE Non-Drop (29.97 fps, Video)** from the submenu. Notice that status values change to hours, minutes, seconds, and frames.



6. Experiment with each status format and make note of how each format appears.

Notes:

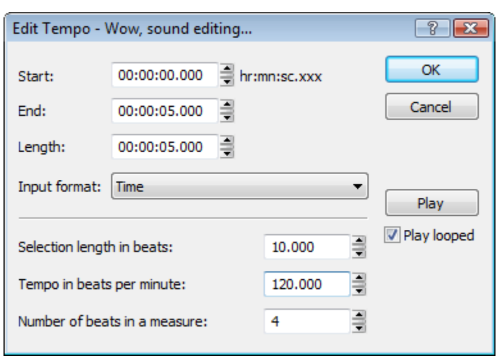
- *Selecting a new format changes the status format for the current data window only.*
- *To quickly change a file's status format, right-click any of the data window's status display boxes and choose a new format from the shortcut menu.*

Configuring the Measures & Beats format

Choosing the **Measures & Beats** format allows you to specify the beats per minute and beats per measure values used to calculate measures and beats.

Changing a file's beat values

1. From the **Options** menu, choose **Status Format**, and then choose **Edit Tempo** from the submenu. The Edit Tempo dialog appears.



2. Type an appropriate value in the **Tempo in beats per minute** box.
3. Type an appropriate value in the **Number of beats in a measure** box and click **OK**.

Alternately, you can make a selection in the file equal to one measure, and then type the number of beats in the sample measure in the **Selection length in beats** box. The **Tempo** value is automatically calculated based on the selection length and number of beats.

Changing the default beat values

The previous procedure changes the beat values for the current audio file only. Use the following steps to change the Sound Forge default beat values.

1. From the **Options** menu, choose **Preferences**. The Preferences dialog appears.
2. Click the **Status** tab.
3. Type an appropriate value in the **Default beats per measure** box.
4. Type an appropriate value in the **Default beats per minute** box and click **OK**.

Publishing files to the Web

You can share your media file with others by publishing it to the Web from within the software. You can upload your file to ACIDplanet.com or another publishing provider. From the **File** menu, choose **Publish** and follow the instructions to set up your publishing providers and upload your content.

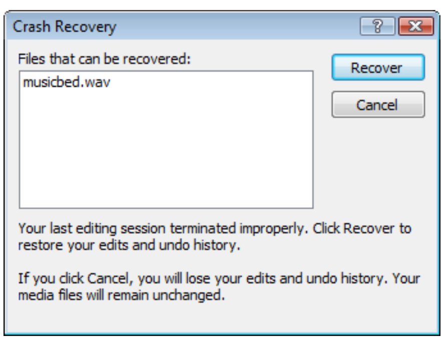
Recovering files after a crash

If Sound Forge software terminates improperly, you can recover all open and unsaved audio files not opened in read-only mode. When a file is opened, it automatically creates temporary files that it uses to save any changes made to the file. The original file remains unchanged until it is saved. If the software terminates improperly, the temporary files remain on your hard drive and can be used to recover any unsaved changes made prior to the crash.

Tip: *You can specify the folder used to store temporary files by choosing* **Preferences** *from the* **Options** *menu and designating a* **Temporary files and record folder** *location on the* **General** *tab.*

Recovering files

Click the **Recover** button to restore the changes and undo history for the files listed in the **Files that can be recovered** list.



Deleting recovered files

Click the **Cancel** button to delete the temporary files. The original media files remain unchanged.

Chapter 5

Navigating, Zooming, and Selecting

This chapter introduces some of the Sound Forge® Pro navigation and selection features.

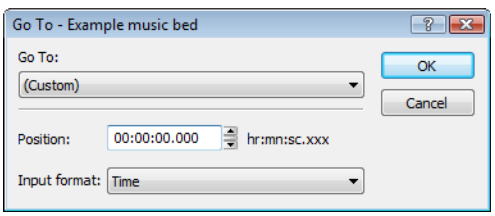
Setting the cursor position

While you can click anywhere in thìe waveform to position the cursor, there are times when you may need to position the cursor more precisely. You can use the Go To dialog to move the cursor to a specific point in an audio file and center it in the data window.

> **Tip:** *You can also use a variety of keyboard shortcuts to position the cursor. For more information, see* *Cursor movement shortcuts on page 352*.

1. Choose **Go To** from the **Edit** menu. The Go To dialog appears.

> **Tip:** *You can also use the following methods:*
>
> - *Right-click the waveform, choose* **Cursor***, and choose* **Go To** *from the submenu.*
> - *Press Ctrl+G.*



2. Set the cursor position using one of the following methods:
   - From the **Go To** drop-down list, choose a preset.
   - From the **Input format** drop-down list, choose a format and type an appropriate value in the **Position** box.
3. Click **OK**. The cursor is placed at the specified position in the data window.

Previewing audio with pre-roll

Many audio editing operations depend upon accurate placement of the cursor in the data window. The **Pre-roll to Cursor** command allows you to preview audio data leading up to the current cursor position. This command is extremely useful when recording punch-ins. *For more information, see* *Generating MTC/SMPTE synchronization during recording on page 156*.

A 1.5 second pre-roll is automatically designated. However, you can change this value if necessary. *For more information, see* *Configuring cut pre-roll and post-roll lengths on page 82*.

1. Place the cursor anywhere in the data window.
2. From the **Transport** menu, choose **Pre-roll to Cursor** (or press Ctrl+Shift+K). Sound Forge software plays the audio leading up to the cursor and stops at the cursor.

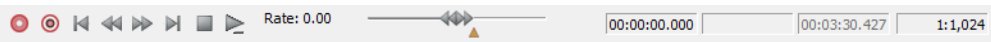
Scrubbing

Scrubbing is a type of timeline playback that gives you precise control over the speed and direction of playback. Both linear and logarithmic scale scrubbing are allowed.

Tip: *Choose a setting from the* **JKL/shuttle speed** *drop-down list on the* **Editing** *tab of the Preferences dialog to control the scrub speed and range when using the keyboard or multimedia controllers.*

Scrubbing with the scrub control slider

The scrub control slider ( ), located at the bottom of the data window, can be dragged back and forth. The farther from the center that the slider is dragged, the faster the playback, both forward and in reverse.

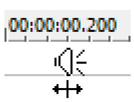


Note: *You can also drag the* **Normal Rate** *indicator ( ), which is located below the scrub control, to adjust playback speed or double-click* **Rate** *and type a playback rate.*

Scrubbing on the timeline

You can scrub the project by using the timeline.

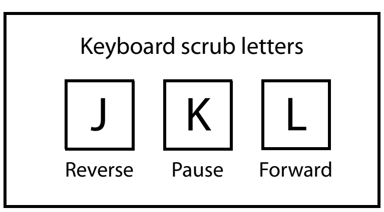
1. Position the cursor on the timeline, hover the mouse pointer over the cursor and press Ctrl. The mouse pointer changes to a speaker icon.
2. Left click and drag the mouse left or right to scrub the timeline. The cursor changes again to a pan/scrub icon.



Press Ctrl over the timeline cursor

Left-click and drag to scrub

Scrubbing with the keyboard



Three letters (JKL) are used as a keyboard scrub control.

- Press J for reverse playback. Press again to accelerate the playback rate.
- Press K to pause playback.
- Press L for forward playback. Press again to accelerate the playback rate.

There are several ways to adjust the playback speed:

- Hold K while pressing J or L to emulate a shuttle knob mode.
- Press K+J to turn the knob to the left or K+L to turn the knob to the right.
- Press K again or Spacebar to return to normal mode.

Scrubbing with the audio event locator

Holding Ctrl and dragging the mouse within the overview bar initiates playback of small audio loops adjacent to the cursor position. This is not technically a scrub function, but it serves a similar purpose. It allows you to audition brief audio segments and quickly locate specific events within a file. Playback stops when the mouse button is released.

Configuring the audio event locator

You can set the amount of pre-roll and loop duration for the audio event locator.

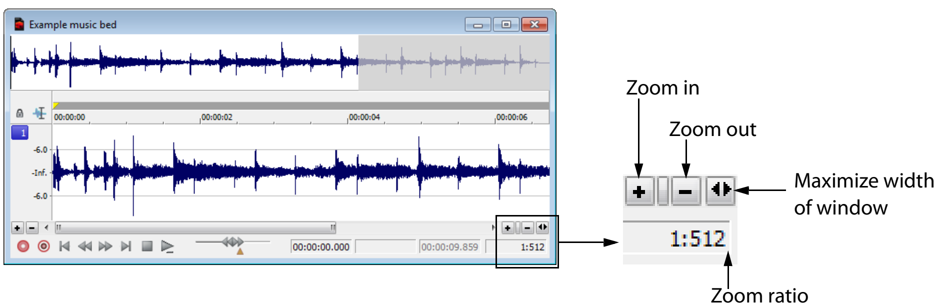
1. From the **Options** menu, choose **Preferences**. The Preferences dialog appears.
2. Click the **Previews** tab.
3. In the **Audio event locator** section, edit the **Pre-roll** and **Loop time** values as desired and click **OK**.

Zooming and magnifying

Because there are considerably more samples in a sound file than horizontal points (pixels) on the screen, many data samples must be represented by each horizontal point when audio data displays in the data window. Depending upon the editing operation, you may want to view the entire file at once or a small portion of data in greater detail. For this reason, you can utilize two varieties of zooming: time ruler zooming and level ruler zooming. You can also zoom to events when using the Event tool ( ).

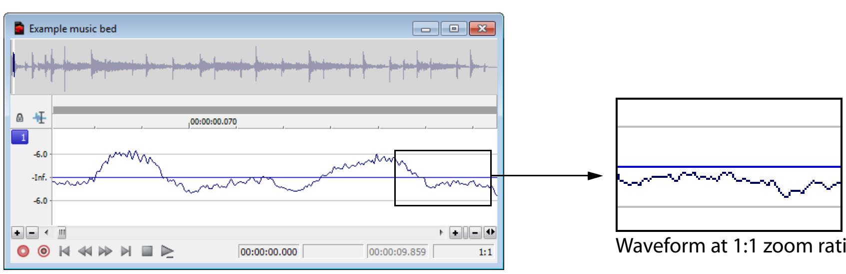
Zooming the time ruler (horizontal)

The current time ruler magnification ratio appears in the lower-right corner of the data window above the status boxes.



Understanding the zoom ratio

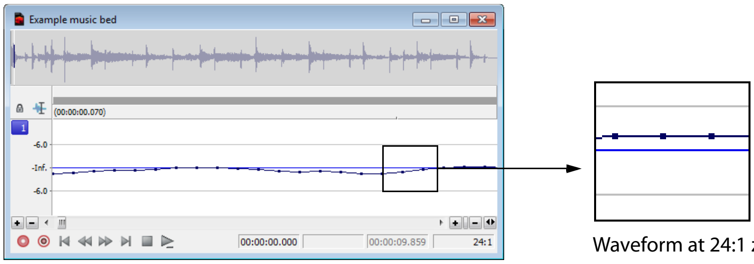
The zoom ratio determines the number of samples represented by each horizontal point on the screen. The zoom ratio is a value of X:Y, where X is the number of horizontal points and Y is the number of samples. If the ratio is 1:1, each point on the screen represents one sample. At this zoom ratio, a brief but detailed selection of time is displayed.



Waveform at 1:1 zoom ratio

Conversely, if the zoom ratio is 1:1024, 1,024 samples are represented by each point on the screen and a greater length of time is displayed.

For very precise editing, you may want to zoom in more tightly than a 1:1 ratio. Sound Forge allows up to a 24:1 ratio, where 24 points on the screen represent one sample. This high level of zoom can be useful when editing with the Pencil tool ( ). *For more information, see* *Repairing audio glitches manually with the Pencil tool* *on page* *165*.



Waveform at 24:1 zoom ratio

Changing the zoom ratio

To edit the zoom ratio, use the **Zoom In/Out** spin control located adjacent to the zoom ratio display.

- Clicking the plus/minus buttons increases/decreases the zoom ratio by single-step increments.
- Dragging the spin control increments the zoom ratio quickly in the corresponding direction.

Notes:

- *When a file is opened, the horizontal magnification is set to the value specified by the* **Normal zoom ratio** *setting in the* **Display** *tab in the Preferences dialog. To access the Preferences dialog, choose* **Preferences** *from the* **Options** *menu.*
- *Right-clicking the waveform display allows you to quickly access time ruler zoom commands from the shortcut menu.*

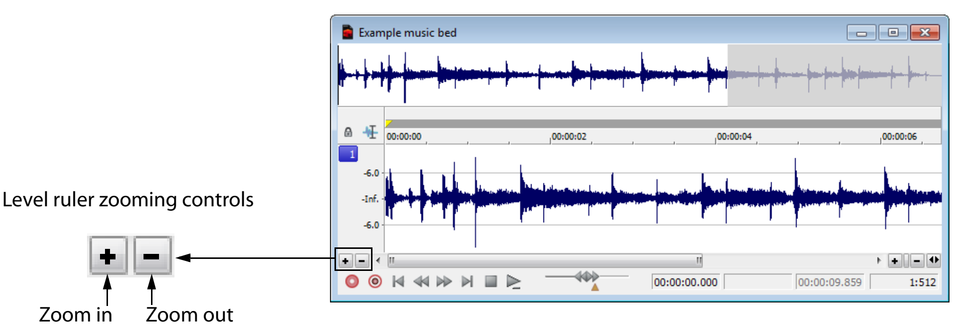
Using zoom time commands

If you prefer using commands, you can control the time magnification from the **View** menu. The following table briefly describes the available time zoom commands. You can access these commands from the **View** menu by choosing **Zoom Time** and choosing the desired command from the submenu.

Command	Description
In Full	Increases the zoom ratio to represent each audio sample with 24 screen pixels (24:1 zoom ratio).
Normal	Returns the file to its default zoom ratio.
Out Full	Changes the zoom ratio to display the entire file within the data window.
Selection	Changes the zoom ratio to maximize the display and center the selection within the data window.
Custom Zoom X:Y	Sets the zoom ratio to a custom setting. *For more information, see Using custom zoom settings on page 95.*

Zooming the level ruler (vertical)

Zooming along the level ruler displays a larger vertical waveform and allows for more precise editing at low audio amplitudes.

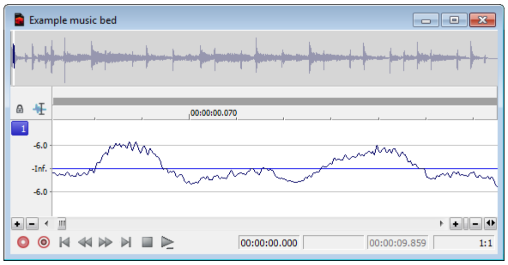


Changing the level zoom

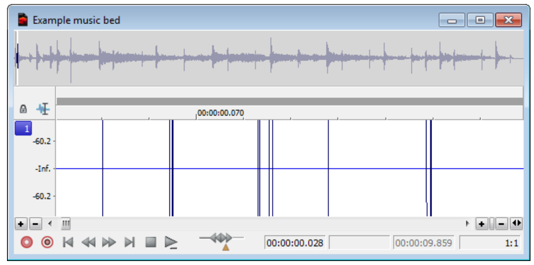
To edit the level ruler zoom, use the **Zoom In/Out** spinner control located above the playbar.

- Clicking the plus/minus buttons increases/decreases the level ruler zoom by single-step increments.
- Dragging the spin control increments the level ruler zoom quickly in the corresponding direction.

At high zoom levels, only low-level samples are visible because the peaks of the waveform move beyond the vertical scope of the data window. Consider the following data windows.



Level zoom out maximum



Level zoom in maximum

Both data windows display the same audio file at a 1:1 zoom ratio. The window on the left shows the level ruler zoomed to its maximum out position. The window on the right shows the level ruler zoomed to its maximum in position. Notice that wave peaks clearly visible in the left window are out of display range in the right window.

Using zoom level commands

If you prefer using commands, you can control the level magnification from the **View** menu. The following table briefly describes the three available zoom level commands. You can access these commands from the **View** menu by choosing **Zoom Level** and choosing the desired command from the submenu.

Command	Description
Out Full	Decreases the zoom level to minimize the display of the file's amplitude.
Window	Changes the level zoom to display the entire waveform amplitude in the data window.
Selection	Maximizes the display of the selection (vertically and horizontally) in the data window.

Formatting the level ruler

You can configure the level ruler to appear in decibels or percent by right-clicking the ruler and choosing **Label in Percent** or **Label in dB** from the shortcut menu.

Using custom zoom settings

You can create two custom time zoom settings for quick access to time magnification levels that you use frequently.

Creating custom zoom settings

1. From the **Options** menu, choose **Preferences**. The Preferences dialog appears.
2. Click the **Display** tab.
3. Select time magnification settings from the **Custom zoom ratio 1** and **Custom zoom ratio 2** drop-down lists.
4. Click **OK**.

Zooming to custom settings

From the **View** menu, choose **Zoom Time**, and choose a custom zoom setting from the submenu.

> **Tip:** *You can also click a* **Custom Zoom** *button (or) on the Navigation toolbar or press 1 or 2 on the numeric keypad.*

Using zooming shortcuts

Zooming to a selection

1. Create a selection. If no selection is created, the **Zoom Selection** function is not available.
2. Right-click the waveform and choose **Zoom Selection** from the shortcut menu. The minimum zoom ratio that allows the full selection to appear in the window is calculated, and the selection is then zoomed and centered in the data window.

> **Note:** *To reverse this function, right-click the waveform, choose* **Zoom***, and choose* **Out Full** *from the submenu.*

Zooming the window

Right-click the level ruler and choose **Zoom Window** from the shortcut menu. The maximum zoom level that allows the loudest portion of the selection to appear in the window is calculated and the entire sound file is adjusted.

> **Note:** *To reverse this function, right-click the level ruler and choose* **Zoom Out Full** *from the shortcut menu.*

Zooming out full

To quickly display all data in a data window, right-click the waveform, choose **Zoom**, and choose **Out Full** from the submenu. This command sets the zoom ratio and zoom level to the lowest values required to display all data in the window.

> **Note:** *To reverse this function, go to the* **View** *menu, choose* **Zoom Time***, and choose* **Normal** *from the submenu.*

Zooming in full

To quickly set the zoom factor to its maximum magnification, right-click the waveform and choose **Zoom In Full** from the shortcut menu. The maximum magnification available is 24:1.

> **Note:** *To reverse this function, right-click the waveform and choose* **Zoom Normal** *from the shortcut menu.*

Optimizing time and level ruler scaling

To optimize both the time ruler and level ruler display of a selection, double-click the level ruler. Double-clicking the level ruler a second time restores both displays to their default levels.

Using the Magnify tool

The Magnify tool provides an additional way to magnify a section of an audio file. You can access the Magnify tool in three ways:

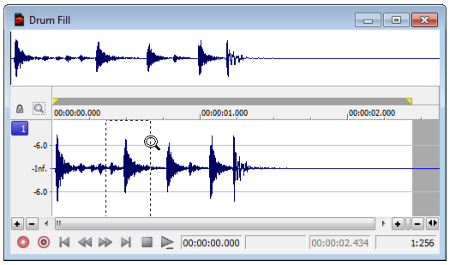
- From the **Edit** menu, choose **Tool**, and choose **Magnify** from the submenu.
- Click the **Magnify Tool** button ( ) on the Standard toolbar.
- Click the Edit Tool Selector in the upper-left corner of the data window until the Magnify tool is selected.

> **Tip:** *When the* **Allow Ctrl+drag style zoom in data windows** *check box is selected on the General tab of the Preferences dialog, you can hold Ctrl while creating a selection to temporarily use the Magnify tool.*

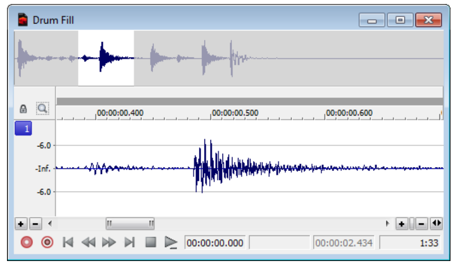
When you select the Magnify tool, the cursor appears as a magnifying glass ( ). You can use this tool to create a selection box indicating how audio data is magnified. By using the Magnify tool and toggle-clicking the mouse, you can toggle between time zoom, level zoom, and simultaneous time/level zoom. *For more information, see Using the mouse on page 27*.

Zooming the time ruler with the Magnify tool

1. Drag the Magnify tool on the waveform to make a small selection box.
2. Toggle-click the mouse until the selection box is the same height as the data window.
3. Drag the Magnify tool to create a time zoom selection and release the mouse button. The zoom ratio of the selection increases.



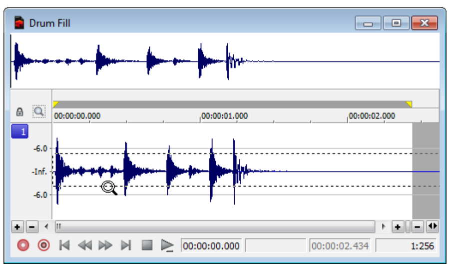
Selection box spans the height of the window



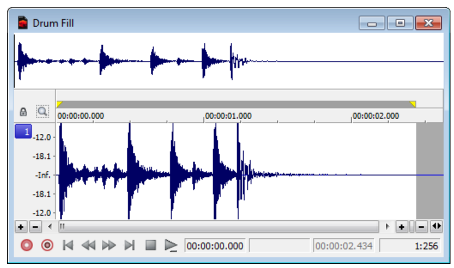
The selection is time zoomed

Zooming the level ruler with the Magnify tool

1. Drag the Magnify tool on the waveform to make a small selection box.
2. Toggle-click the mouse until the selection box is the full width of the data window.
3. Drag the Magnify tool to create a level zoom selection and release the mouse button. The zoom ratio of the selection increases.



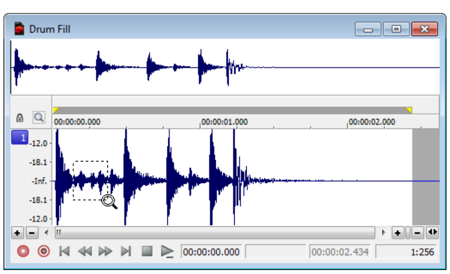
Selection box spans the width of the window



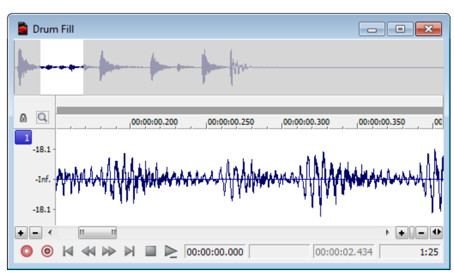
The selection is level zoomed

Zooming both time and level with the Magnify tool

1. Drag the Magnify tool on the waveform to make a small selection box.
2. Toggle-click the mouse until the selection appears as a box.
3. Drag the Magnify tool to create a time/level zoom selection and release the mouse button. The level zoom and time zoom of the selection increase.



Selection displays as a box



The selection time and level are zoomed

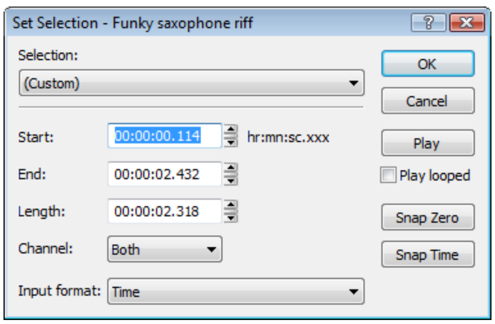
Selecting audio using start and end values

You can select audio by dragging the mouse or by using keyboard shortcuts. *(For more information, see Data selection shortcuts on page 353.)* For the sake of accuracy, however, it is often useful to create selections by entering specific start and end point values. The Set Selection dialog allows you to create selections in this way or by choosing a preset selection from the **Selection** drop-down list.

1. From the **Edit** menu, choose **Selection**, and then choose **Set** from the submenu (or press Ctrl+Shift+D). The Set Selection dialog appears.
2. From the **Input format** drop-down list, choose the format to be used for creating the selection. The values in the **Start**, **End**, and **Length** boxes change to reflect the specified format.
3. Configure the selection by typing appropriate values in the **Start** and **End** or the **Start** and **Length** boxes.
4. If you are working with a stereo file, choose **Left**, **Right**, or **Both** from the **Channel** drop-down list, or, for a multichannel file, type the appropriate channel numbers in the **Channel** box.
5. Click **OK**.

Using the Set Selection dialog

The following sections briefly describe additional controls located in the Set Selection dialog.



Control	Description
Play	Clicking **Play** plays the current selection.
Play looped	Selecting the **Play looped** check box allows you to play the selection in Looped Playback mode.
Snap Zero	Clicking **Snap Zero** forces the **Start** and **End** values of the selected area to the next zero-crossing.
Snap Time	Clicking **Snap Time** forces the **Start** and **End** values of the selected area to a whole time division as designated by the markings on the data window's time ruler.

Zero-crossing preference

When using a **Snap Zero** command, you can configure the application to snap to positive slope, negative slope, or either slope zero-crossings.

1. From the **Options** menu, choose **Preferences,** and click the **Editing** tab.
2. From the **Snap to zero-crossing slope** drop-down list, choose the desired slope and click **OK**.

Selecting audio during playback

You can create selections during playback using the **Mark In** and **Mark Out** commands. These commands place temporary markers in the data window, which are then used to create a loop region. While you can place these markers by choosing **Mark In** and **Mark Out** from the **Selection** submenu under the **Edit** menu, the keyboard equivalents are more useful.

1. Play the audio file in the current data window.
2. During playback, press I where the selection will begin.
3. Press O where the selection will end. A loop region is created using the in and out points you identified.

You can then right-click the bar at the top of the loop region to insert a region, toggle looped playback, or select the loop region.

Tip: *Select the* **Update loop bar on Mark In/Out** *check box on the* **Editing** *tab of the Preferences dialog if you want the loop bar in a data window to be updated when you mark the beginning or end of a selection. When the check box is cleared, the loop bar isn't updated until you've marked both ends of the selection.*

Fine-tuning a selection

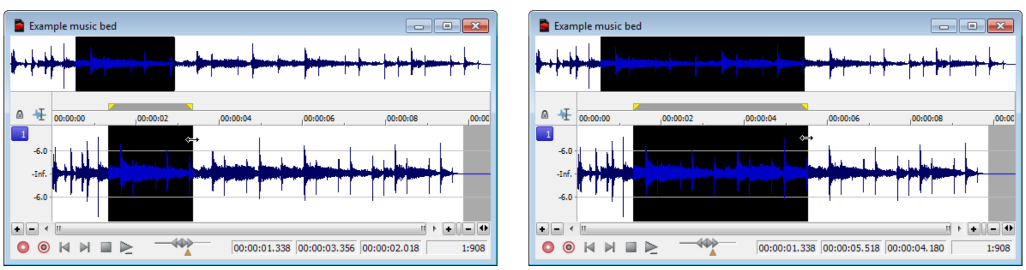
After creating a selection, you may discover that the start or end point has not been positioned properly. In cases like this, you can try to reselect the data, but it can be difficult to accurately create selection points. For this reason, you have a number of tools designed to help you fine-tune selections.

If you find that the selection jumps unexpectedly as you fine-tune it, snapping may be turned on. *For more information, see Enable Snapping on page 100*.

Adjusting a selection with the mouse

You can fine-tune selection start and end points by dragging the edge of the selection to a new location.

1. Open a file and create a selection in the waveform.
2. Position the mouse pointer over one of the selection edges. The pointer is displayed as a bi-directional arrow (↔).
3. Drag the selection edge to a new position.



Drag the edge of the selection to a new position.

4. Release the mouse button. The selection is updated.

Adjusting a selection with the keyboard

Using the keyboard, you can quickly and accurately select data or update a selection. *For more information, see Data selection shortcuts on page 353*.

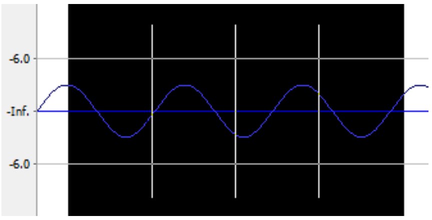
Restoring a selection

If you lose a selection while editing, you can step backward through the previous five time selections. From the **Edit** menu, choose **Selection**, and then choose **Cycle** from the submenu, or press Backspace.

From the **Edit** menu, choose **Selection**, and then choose **Toggle** from the submenu (or press S or /) to switch between the last selection and the last cursor position.

Using selection grid lines

From the **Options** menu, choose **Selection Grid Lines** to display grid lines that divide the selection into four equal parts. These lines make creating loops from existing material easier.



To change the number of divisions that will be used, choose **Options** > **Set Grid Divisions**, and then choose a setting from the submenu.

For example, if you're trying to create a loop in 3/4 time, changing the grid divisions to 3 allows you to divide a selection into three beats.

Enable Snapping

From the **Options** menu, choose **Snapping**, and then choose **Enable** from the submenu to turn automatic snapping on or off in data windows.

Snapping helps you position the cursor, make selections and align items along the grid when you paste, mix, trim, or work with markers and regions.

As you drag items in a data window, snap points are highlighted.

If you want to modify an existing selection, you can use the commands on the **Edit** > **Selection** submenu to snap the selection to the grid and to zero crossings.

Tip: *Hold the Shift key to temporarily override snapping.*

Important: *In previous versions of Sound Forge, both ends of a selection would snap to the grid when you released the mouse. Sound Forge now uses a softer snapping mechanism that gives you more control: if you click near a grid line with snapping enabled, the cursor will snap to that grid line. As you drag a selection along the timeline, the cursor will stick to grid lines to allow you to snap to them. If you want to snap both edges of a selection to the grid as in previous versions of Sound Forge, you can press T.*

Turn snapping on or off

From the **Options** menu, choose **Snapping**, and then choose **Enable** from the submenu (or press F8) to turn snapping on or off. When snapping is enabled, objects will snap to the following points:

- The cursor
- Time selection edges

You can also choose to snap events to grid divisions, markers, or zero crossings.

Automatically snap to the grid

When snapping is enabled, you can also choose to have objects snap to whole time divisions as designated by the marks on the time ruler above the data window.

From the **Options** menu, choose **Snapping**, and then choose **Grid** from the submenu (or press Ctrl+F8) to toggle snapping to grid lines.

Tip: *To change the resolution of the grid, choose* **Status Format** *from the* **Options** *menu and then choose a setting from the submenu (or right-click the time ruler and choose a format from the shortcut menu).*

Automatically snap to markers

When snapping is enabled, you can also choose to have elements in the data windows snap to markers.

From the **Options** menu, choose **Snapping**, and then choose **Markers** from the submenu (or press Shift+F8) to toggle snapping for the following marker types:

- Markers
- Regions
- Command markers
- Disc-at-once CD tracks and indexes
- Sample markers

Automatically snap to events

When snapping is enabled, you can also choose to have elements in the data windows snap to event boundaries.

From the **Options** menu, choose **Snapping**, and then choose **Events** from the submenu (or press Ctrl+Shift+F8) to toggle snapping to event edges.

Automatically snap to zero crossings

When snapping is enabled, you can also choose to have elements in the timeline snap to zero crossings.

From the **Options** menu, choose **Snapping**, and then choose **Zero Crossings** from the submenu (or press Ctrl+B) to toggle snapping to zero crossings.

Turn off automatic snapping at high zoom levels

When you're zoomed in, you may wish to turn off snapping so you can position a selection's start and end points exactly where you choose.

1. From the **Options** menu, choose **Preferences**, and then click the **Editing** tab.
2. Select the **Disable auto-snapping below 1:4 zoom ratios** check box.
3. Click **OK**.

Snap an existing selection to the grid or zero crossings

If you've created a selection without automatic snapping enabled or have modified a selection so its edges no longer align with the grid/zero crossings, you can use the commands on the **Edit** > **Selection** submenu to snap the selection to the grid and to zero crossings. *For more information, see Selection Snapping on page 101*.

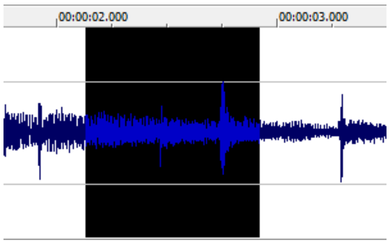
Selection Snapping

From the **Edit** menu, choose **Selection**, and then choose a command from the submenu to force the edges of the current selection to the points you choose. Snapping helps you align your selection with items in the data window.

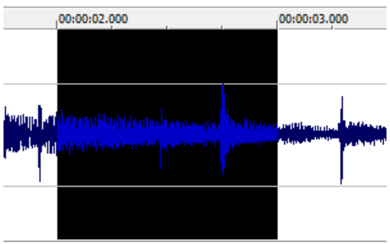
If you want to use snapping when positioning the cursor and making selections, you can use the commands on the **Options** > **Snapping** submenu to enable snapping and set snapping options. *For more information, see Enable Snapping on page 100*.

Snap to Grid

From the **Edit** menu, choose **Selection**, and then choose **Snap to Grid** from the submenu (or press T) to force both edges of a selection to a whole time division as designated by the marks on the time ruler above the data window.



Before snapping.



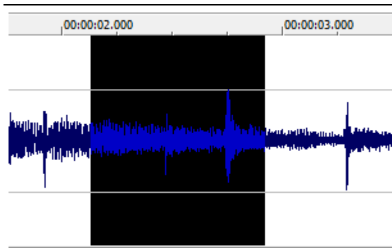
After snapping, the edges of the selection snap to the 2.000 and 3.000 divisions on the ruler.

Tip: *To change the resolution of the grid, choose* **Status Format** *from the* **Options** *menu and then choose a setting from the submenu (or right-click the time ruler and choose a format from the shortcut menu).*

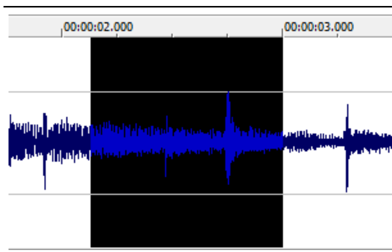
Snap Edge to Grid

From the **Edit** menu, choose **Selection**, and then choose **Snap Edge to Grid** from the submenu (or press Shift+T) to force the active edge of a selection to a whole time division as designated by the marks on the time ruler above the data window.

The active edge of a selection is defined by the blinking cursor. Press Home or End to change the active edge.



Before snapping.



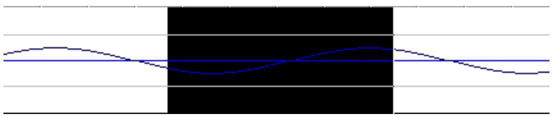
After snapping, the active (right) edge of the selection snaps to the 3.000 division on the ruler.

Tip: *To change the resolution of the grid, choose* **Status Format** *from the* **Options** *menu and then choose a setting from the submenu (or right-click the time ruler and choose a format from the shortcut menu).*

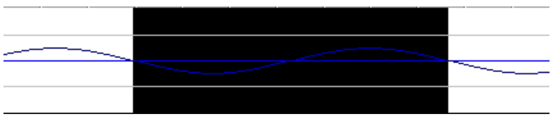
Snap to Zero

Performing edits at zero-crossings reduces the possibility of introducing glitches in your sound file.

From the **Edit** menu, choose **Selection**, and then choose **Snap to Zero** from the submenu (or press Z) to force both edges of a selection to the next zero-crossing of the waveform.



Before snapping.



After snapping, the edges of the selection snap to the nearest locations where the waveform crosses the baseline.

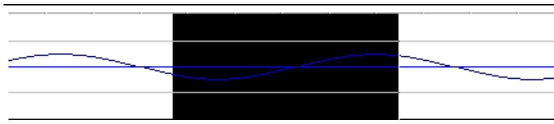
> **Note:** *The* **Editing** *tab in the Preferences dialog allows you to choose whether this is a negative, positive or any zero-crossing. For more information, see Editing tab on page 335.*

Snap Edge to Zero

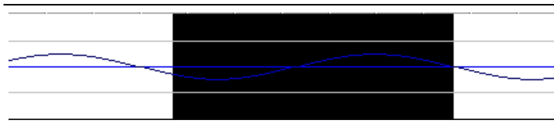
Performing edits at zero-crossings reduces the possibility of introducing glitches in your sound file.

From the **Edit** menu, choose **Selection**, and then choose **Snap Edge to Zero** from the submenu (or press Shift+Z) to force the active edge of a selection to the next zero-crossing of the waveform.

The active edge of a selection is defined by the blinking cursor. Press Home or End to change the active edge.



Before snapping.



After snapping, the active (right) edge of the selection snaps to the nearest location where the waveform crosses the baseline.

> **Note:** *The* **Editing** *tab in the Preferences dialog allows you to choose whether this is a negative, positive or any zero-crossing. For more information, see Editing tab on page 335.*

Extend to Next Zero

Performing edits at zero-crossings reduces the possibility of introducing glitches in your sound file.

From the **Edit** menu, choose **Selection**, and then choose **Extend to Next Zero** from the submenu (or press Z) to force both edges of a selection to the next zero-crossing of the waveform.

> **Note:** *The* **Editing** *tab in the Preferences dialog allows you to choose whether this is a negative, positive or any zero-crossing. For more information, see Editing tab on page 335.*

Extend Edge to Next Zero

Performing edits at zero-crossings reduces the possibility of introducing glitches in your sound file.

From the **Edit** menu, choose **Selection**, and then choose **Extend Edge to Next Zero** from the submenu (or press Shift+Z) to force the active edge of a selection to the next zero-crossing of the waveform.

The active edge of a selection is defined by the blinking cursor. Press Home or End to change the active edge.

> **Note:** *The* **Editing** *tab in the Preferences dialog allows you to choose whether this is a negative, positive or any zero-crossing. For more information, see Editing tab on page 335.*

Quantize to Frames

From the **Options** menu, choose **Quantize to Frames** to force edits to occur on project frame boundaries. This setting is independent of grid and marker snapping and is useful when editing audio for video and creating disc-at-once CD projects.

When **Quantize to Frames** is turned on, the following actions will always occur on frame boundaries:

- Positioning the cursor
- Making selections
- Placing markers and regions

> **Note:** *If you drag to a snap point that does not occur on a frame boundary when* **Quantize to Frames** *is enabled, the snap point will be quantized to the nearest frame boundary.*

Creating and using views

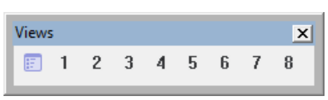
Views are used to save and recall selections, zoom ratios, and waveform display positions. Sound Forge software can retain eight different views for any audio file, each containing any or all of the following elements:

- Selection
- Cursor position
- Magnification
- Position scroll bar placement

> **Tip:** *Views are discarded when you close the file. To save views with a file, save the file as part of a workspace. For more information, see Saving files as a workspace on page 75.*

Displaying the Views toolbar

1. From the **View** menu, choose **Toolbars**.
2. Select the **Views** check box and click **OK**. The Views toolbar appears.



Creating views

1. Open the Voiceover.pca file and create a selection containing “Wow.”
2. Click the **Set** button ( ) on the Views toolbar. A view can now be created.
3. Click the **1** button (1). The selection is saved as view 1 and the **1** button (1) is underscored to indicate that a view was created.
4. Create a new selection anywhere in the audio file, preferably at an increased magnification.
5. Click the **Set** button ( ) followed by the **2** button (2). The selection is saved as view 2.
6. Click the **1** button (1). The view 1 selection is used.
7. Click the **2** button (2). The view 2 selection is used.

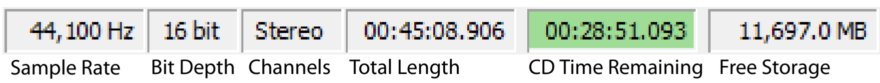
Chapter 6

Changing File Properties and Formats

This chapter deals with the supported file properties and formats in Sound Forge® Pro software and discusses file summary information.

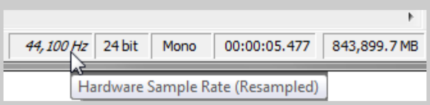
Editing file properties

When you open or create a file, its properties are displayed in the first four boxes of the status bar at the bottom of the Sound Forge workspace. The file properties are sample rate, bit depth, channels, and length.



Notes:

- *The* **CD Time Remaining** *box is displayed only when CD tracks exist in the active data window. You can use the CD Settings tab in the Preferences dialog to specify whether the software should automatically detect CD lengths or to set a default CD length. For more information, see* *CD Settings tab on page 341*.
- *If the active data window's sample rate is not supported by your audio hardware, the output will be resampled to a supported rate for playback when you're using an ASIO audio device. During playback, the* **Sample Rate** *box in the Status Bar is displayed in italics to indicate that the output has been resampled.*



Double-click the **Sample Rate**, **Bit Depth**, **Channels**, or **Total Length** box in the status bar to edit properties quickly.

Note: *You cannot modify file properties for musical instrument files.*

Editing file properties in the File Properties window

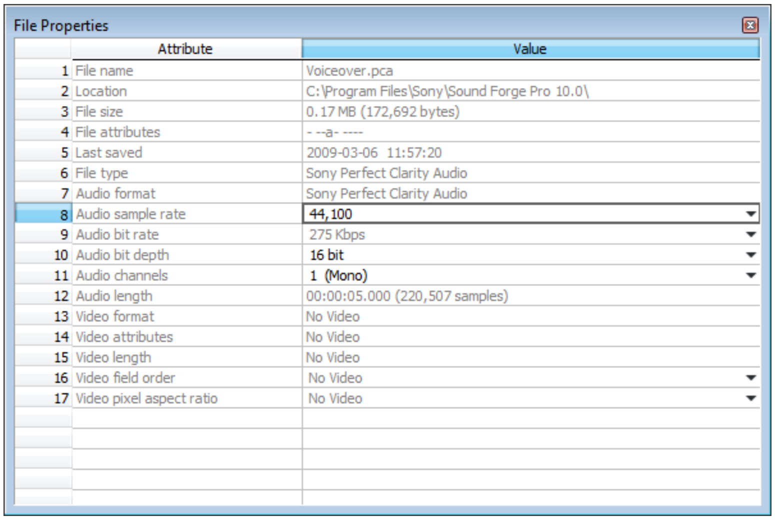
You can edit file properties in the File Properties window.

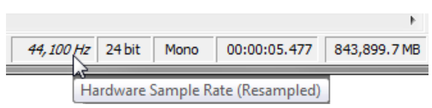
1. From the **View** menu, choose **File Properties**. The File Properties window appears.

Tip: *You can also access the File Properties window by performing either of the following actions:*

- *Right-click the waveform display and choose* **Properties**.
- *Press Alt+Enter.*

2. Edit the file properties as needed.



Item	Description
File name	The name of the file saved on disk.
Location	The folder where the file is saved.
File size	The size of the file on disk.
File attributes	Indicates whether file attributes (read-only, hidden, etc.) have been set.
Last saved	The date and time the file was saved.
File type	The file type for the file.
Audio format	The format used to save the audio stream.
Audio sample rate	Click the down arrow (▼) and choose a sample rate from the drop-down list, or choose **Custom** to type a value in the edit box to set the number of samples per second used to represent the audio. **Notes:** • *This setting will not resample the sound file. If the playback rate is different from the originally recorded rate, the pitch will vary unless the file is resampled.* • *If the active data window's sample rate is not supported by your audio hardware, the output will be resampled to a supported rate for playback. During playback, the* **Sample Rate** *box in the Status Bar is displayed in italics to indicate that the output has been resampled.* 
Audio bit rate	Displays the bit rate of the audio file.
Audio bit depth	Click the down arrow (▼) and choose a bit depth from the drop-down list to set the number of bits used to represent each sample. **Note:** *Use the Bit-Depth Converter to perform advanced channel mixing.*
Audio channels	Click the down arrow (▼) and choose a setting from the drop-down list to set the number of channels stored in the file. **Note:** *Use the Channel Converter to perform advanced channel mixing.*
Audio length	The duration (in time and samples) of the audio file.
Video format	Displays the format used to save the video stream.
Video attributes	Displays the frame size, color depth and frame rate of the video stream.
Video length	Displays the length (in time and frames) of the video stream.

Item	Description
Video field order	Displays the field order of the video stream. Click the down arrow (▾) and choose a setting from the drop-down list to change the field order.
Video pixel aspect ratio	Displays the pixel aspect ratio of the video stream. Click the down arrow (▾) and choose a setting from the drop-down list to change the pixel aspect ratio. Computers display pixels as squares, or a ratio of 1.0. Televisions display pixels as rectangles (ratios other than 1.0). Using the incorrect setting can result in distortion or stretching. Consult your capture/video output card's manual for the proper settings.

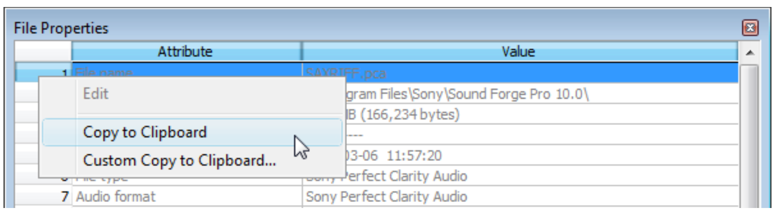
Editing file properties in the status bar

You can quickly edit individual file properties from the status bar using either of the following methods:

- Right-click the status value to be changed and choose a new value from the shortcut menu.
- Double-click the status value to be changed and type a new value in the edit box.

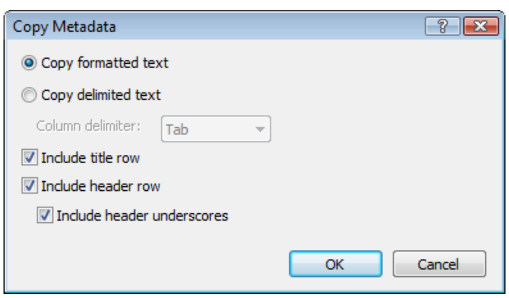
Copying file properties to the clipboard

If you want to copy file properties to the clipboard, right click the File Properties window and choose **Copy to Clipboard** from the shortcut menu.



If you want to customize the format for copying file properties to the clipboard, right-click the File Properties window and choose **Custom Copy to Clipboard** from the shortcut menu.

The Copy Metadata dialog is displayed to let you choose whether you want to copy the data as formatted text or delimited text, set a delimiter, and choose whether you want to include a header row.



Click **OK** to copy the file's properties to the clipboard, and you can then paste the information wherever you need it.

Changing the sample rate

The sample rate is the number of samples per second, measured in hertz (Hz), used to record audio. You can specify sample rates from 2,000 Hz to 192,000 Hz. Typical sample rates are stored as presets in the **Sample rate** drop-down list. In addition, you can increase or decrease the sample rate of an existing audio file.

1. Open and play the Voiceover.pca file. This file is located in the same folder as the application.
2. Right-click the **Sample Rate** status box and choose **48,000** from the shortcut menu.
3. Play the file. Notice that the pitch is higher and the duration is slightly shorter.
4. Right-click the **Sample Rate** status box and choose **8,000** from the shortcut menu.
5. Play the file. Notice that the pitch is lower and the duration is longer.

Changing the sample rate of a file also changes the pitch and duration. To change the sample rate of a file while preserving its duration and pitch, use the **Resample** command. *For more information, see Resample on page 200.*

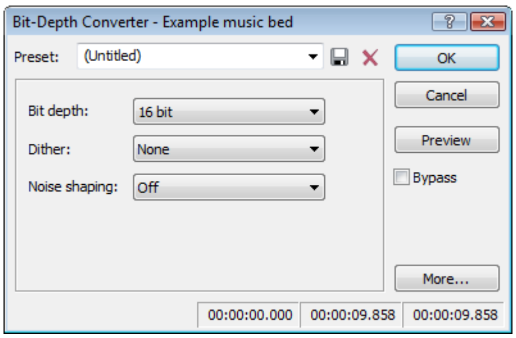
Changing the bit depth

Bit depth refers to the number of bits used to represent a sound. You can increase or decrease a file's bit depth.

Increasing bit depth

Increasing the bit depth does not improve the quality of a file, but it allows subsequent processing to be performed with increased precision.

1. Open a file with a small bit depth.
2. From the **Process** menu, choose **Bit Depth**, and then choose **Bit-Depth Converter** from the submenu. The Bit-Depth Converter dialog appears.



3. From the **Bit depth** drop-down list, choose a larger value and click **OK**.

> **Note:** *When increasing a file's bit depth, the* **Dither** *and* **Noise shaping** *controls should be set to* **None** *and* **Off***, respectively.*

Decreasing bit depth

To maximize storage space, larger sound files (24- and 16-bit) are frequently converted to smaller (16- and 8-bit) files. However, representing a sound file at a decreased bit depth results in audible distortion referred to as quantization error.

1. Open a 16-bit file.
2. From the **Process** menu, choose **Bit Depth**, and then choose **Bit-Depth Converter** from the submenu. The Bit-Depth Converter dialog appears.
3. From the **Bit depth** drop-down list, choose **8 bit**.
4. If desired, choose an option from the **Dither** drop-down list. *For more information, see Dither on page 109.*
5. If desired, choose a **Noise shaping** type. *For more information, see Noise shaping on page 109.*
6. Click **OK**.

> **Note:** *There are no rules regarding maintaining audio quality when decreasing bit depth. Experiment with the* **Dither** *and* **Noise shaping** *controls to determine the optimum settings for each audio file.*

Understanding dither and noise shaping

You can adjust **Dither** and **Noise shaping** settings when decreasing a file's bit depth.

Dither

The **Dither** value determines the randomness of the dither (generated noise) used to mask quantization distortion resulting from conversion to a lower bit depth. This drop-down list requires you to select from several shapes, each of which roughly describes the pattern that would be produced if you plotted a graph with the dither amplitude on the X-axis and the probability of the dither values on the Y-axis.

As is frequently the case when working with audio, you should experiment with dither values to yield the best results. However, keep the following information in mind:

Setting	Description
Half Rectangular	Eliminates distortion resulting from conversion to a lower bit depth, but the noise level is more likely to be dependent on the signal. This setting uses a maximum dither noise amplitude of 0.5 LSB (least significant bit).
Rectangular	Identical to **Half Rectangular**, but with a maximum dither noise amplitude of 1 LSB (least significant bit).
Triangular	Eliminates distortion products as well as any noise floor modulation, but results in a slightly higher noise level. The option typically works well in conjunction with noise shaping. *For more information, see Noise shaping on page 109*.
Highpass Triangular	Behaves like triangular dither, but shifts its noise into higher frequencies. This is typically the best option when used in conjunction with noise shaping. *For more information, see Noise shaping on page 109*.
Gaussian	Does not perform as well as Rectangular and Triangular dither, but may be suitable for certain audio.

Noise shaping

The **Noise shaping** value determines the aural positioning of quantization noise. Using this control, you can shift the noise into audio registers that are less perceptible to human hearing. This lowers the perceived noise floor and creates the illusion of cleaner audio.

- **High-pass contour** noise shaping attempts to push all quantization noise and error into high frequencies.
- **Equal-loudness contour** noise shaping attempts to push the noise under an equal-loudness type of curve.

Noise shaping dangers

Noise shaping places quantization noise near the audio's Nyquist frequency, a value equal to one-half of the file's sample rate. Consider the following information:

- A file with a sample rate of 44.1 kHz has a Nyquist frequency of 22.05 kHz (at the high end of human hearing). Applying noise shaping to this file results in audio perceived to be cleaner than it actually is.
- A file with a sample rate of 22 kHz has a Nyquist frequency of 11 kHz (well within the sensitive range of human hearing). Applying noise shaping to this file results in audio that is perceived to be noisier than it actually is. Ironically, this defeats the entire purpose of noise shaping.

For this reason, we do not recommend using noise shaping on files with sample rates less than 44.1 kHz.

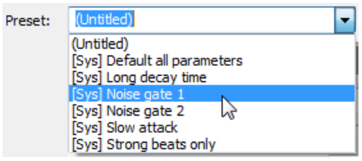
Minimizing quantization error

There are at least three methods of minimizing quantization error when decreasing a file's bit depth: noise gating, compression, and normalization.

Noise gating

Frequently, low-level signals become noise when a file's bit depth is decreased. For this reason, it is preferable to have complete silence between sounds in an audio file.

1. From the **Effects** menu, choose **Noise Gate**. The Noise Gate dialog appears
2. Choose a noise gate preset from the **Preset** drop-down list and click **OK**. A noise gate is applied to the audio, negating its low-level signals.



Compressing

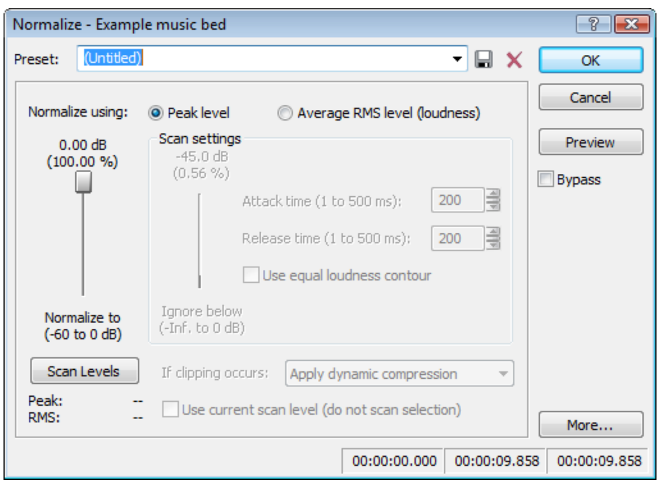
Decreasing the dynamic range of a sound file makes it easier to represent with decreased bit depth.

1. From the **Effects** menu, choose **Dynamics**, and choose **Graphic** from the submenu. The Graphic Dynamics dialog appears.
2. Choose a preset with a small amount of compression (2:1 or less) from the **Preset** drop-down list and click **OK**.

Normalizing

Normalizing a file prior to decreasing its bit depth ensures that the entire dynamic range is used. In addition, normalization lowers the signal-to-noise ratio.

1. From the **Process** menu, choose **Normalize**. The Normalize dialog appears.



2. Select the **Peak level** radio button.
3. Set the **Normalize to** fader to 0 dB (peak) and click **OK**.

Applying compression and normalization simultaneously

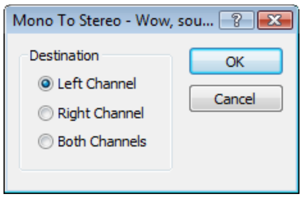
1. From the **Process** menu, choose **Normalize**. The Normalize dialog appears.
2. Select the **Average RMS level** radio button.
3. Choose **Apply dynamic compression** in the **If clipping occurs** drop-down list and click **OK**.

Converting mono/stereo channels

You can convert mono files to stereo or stereo files to mono. To perform quick channel conversion without specifying the mix, use the **Audio channels** box on the File Properties window or right-click the **Channels** box in the status bar and choose **2 (Stereo)** or **1 (Mono)** from the shortcut menu. *For more information, see Editing file properties on page 105.*

Converting from mono to stereo

1. Open the Voiceover.pca file. This file is located in the same folder as the application.
2. Right-click the **Channels** box in the status bar and choose **Stereo** from the shortcut menu. The Mono To Stereo dialog is displayed.



3. Select the **Left Channel** radio button and click **OK**. The mono data is placed in the upper half of the data window (left channel) and silence is placed in the right channel.

   For more information, see Specifying the audio destination on page 111.
4. Play the file. "Wow, sound editing just gets easier and easier" plays in only the left channel.

Tip: *If your sound card supports only mono data, stereo files can be played by specifying the Microsoft Sound Mapper as the playback device. To do this, choose* **Preferences** *from the* **Options** *menu. Click the* **Audio** *tab and choose* **Microsoft Sound Mapper** *from the* **Audio device type** *drop-down list.*

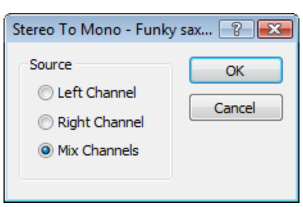
Specifying the audio destination

The **Destination** radio buttons in the Mono To Stereo dialog allow you to specify where the mono audio data is placed in a stereo file. The following table describes the available data destinations.

Destination	Description
Left Channel	The mono data is placed in the left channel. The right channel is set to silence.
Right Channel	The mono data is placed in the right channel. The left channel is set to silence.
Both Channels	The mono data is copied into both channels.

Converting from stereo to mono

1. Open the Saxriff.pca file. This file is located in the same folder as the application.
2. Right-click the **Channels** status box (indicating **Stereo**) and choose **Mono** from the shortcut menu. The Stereo To Mono dialog appears.



3. Select the **Mix Channels** radio button and click **OK**. The left and right channels combine into a mono channel.

   For more information, see Specifying the audio source on page 112.

Specifying the audio source

The **Source** radio buttons in the Stereo To Mono dialog allow you to specify what stereo data is used to create the mono file. The following table describes the available data sources.

Source	Description
Left Channel	Mono data is taken only from the left channel of the stereo file.
Right Channel	Mono data is taken only from the right channel of the stereo file.
Mix Channels	Mono data is created by mixing both channels of the stereo file.

Using the Channel Converter

You can also use the Channel Converter to convert files between mono and multichannel formats. Using the Channel Converter provides the added flexibility of independent level settings for each channel, thereby allowing you to intermix the channels of a multichannel file to create pan effects. To use this tool, choose **Channel Converter** from the **Process** menu. *For more information, see Channel Converter on page 189.*

Converting file formats

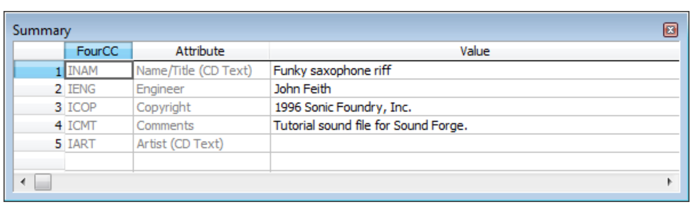
The previous sections have described changing a file's sample rate, bit depth, and channel configuration. You can also convert a file's format and compression settings.

To demonstrate this, open the Voiceover.pca file and choose **Save As** from the **File** menu. Notice the **Save as Type** and **Template** drop-down lists. *For more information, see Using the Save As/Render As dialog on page 72.*

Option	Description
Save as type	In the Save As dialog, the **Save as type** drop-down list defaults to the Sound Forge Pro Project File (.frg) format. However, using the **Save as type** drop-down list, you can specify any supported file type.
Template	The **Template** drop-down list provides standard settings for saving your audio file. If the templates do not match your particular needs, click the **Custom** button to create custom settings.

Adding summary information

Specific audio file types allow you to store text fields of summary information in addition to the audio and video data. File types offering this feature include WAV, AVI, and ASF file formats. You can view and edit these text fields.

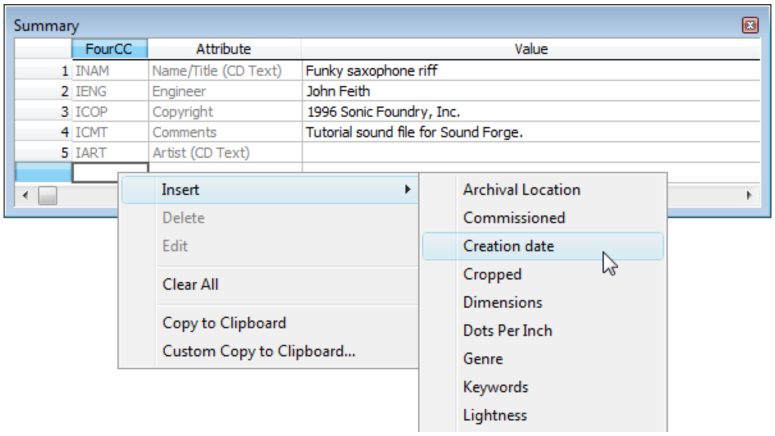


Viewing and editing summary information

The Summary window is used to view and edit the summary information stored in the file.

1. From the **View** menu, choose **Metadata**, and then choose **Summary Information** from the submenu. The Summary window appears.

2. Edit the summary information as needed. You can insert additional summary information by right-clicking in the Summary window and choosing additional summary fields from the **Insert** submenu.



Saving summary information

You can save files containing summary information that have been edited in Sound Forge software with or without summary information.

1. From the **File** menu, choose **Save As**. The Save As dialog appears.
2. Select the **Save metadata with file** check box and click **OK**.

> **Note:** *If you save to a file type that doesn't support metadata, this check box is unavailable.*

Including additional embedded information

Some file formats allow non-text data (such as embedded bitmaps and metafiles) to be embedded in files. If you use the Sound Forge software to edit a file containing data created in another application, Sound Forge software tracks the embedded data and places it back in the file when it is saved in its original format.

Saving additional embedded information

To save additional embedded information, choose **Save As** from the **File** menu and select the **Save metadata with file** check box. If the file type does not support metadata, you are prompted to save the metadata in an external file with an .sfl extension.

Removing additional embedded information

To save a file without additional embedded information, choose **Save As** from the **File** menu and clear the **Save metadata with file** check box.

Editing Multichannel Audio

With Sound Forge® Pro software, you can edit multichannel audio files in the same way you work with mono or stereo files.

Sound Forge software supports multichannel files in the following formats:

- Dolby AC-3 (.ac3) (AC-3 is available as a render format only)
- Raw Audio (.raw)
- Sony AVC (.mp4, .m2ts, .avc)
- Sony Wave64 (.w64)
- Wave (.wav)
- Video For Windows (.avi) – specifically DV, SDI
- Windows Media Audio/Video (.wma or .wmv)
- ATRAC (.oma)
- Material Exchange Format (.mxf)

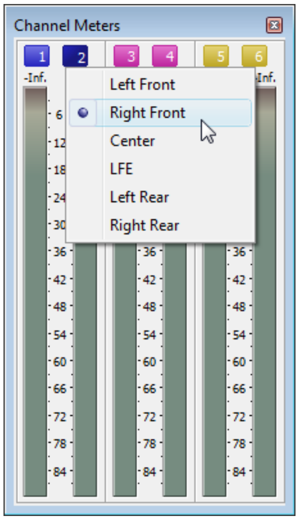
Note: *MXF files require a video stream. If you want to save an audio-only file to MXF format, you must first attach a video stream. For more information, see* *Attaching video to an audio file on page 292*.

When working with MXF files, the number of channels in your source media must match the number of output channels specified by the rendering template. If necessary, use the Channel Converter before rendering.

Render format	Number of channels
DV MXF	Always contains 4 audio channels. You can use the **Channels** drop-down list on the **Audio** tab of the Custom Template dialog to choose how many channels will be filled with audio. For example, if you choose **2** from the **Channels** drop-down list, the rendered file will contain 4 audio channels: two channels will contain audio, and two channels will contain silence.
IMX MXF	Always contains 8 audio channels. You can use the **Channels** drop-down list on the **Audio** tab of the Custom Template dialog to choose how many channels will be filled with audio. For example, if you choose **2** from the **Channels** drop-down list, the rendered file will contain 8 audio channels: two channels will contain audio, and six channels will contain silence.
HD MXF	Can contain 2 or 4 audio channels. You can use the **Channels** drop-down list on the **Audio** tab of the Custom Template dialog to choose how many channels will be rendered. For example, if you choose **2** from the **Channels** drop-down list, the rendered file will contain only 2 audio channels.

Routing channels to hardware outputs

If you're working with multichannel files and have a sound card with multiple outputs, Sound Forge provides you with a great deal of flexibility in routing the channels to the outputs on your sound card: you can route each channel to a separate output, or you can route all the stereo pairs to a single set of outputs to simulate a stereo downmix.



The Hardware Meters window displays a meter and gain fader for each enabled output port. *For more information, see Using the hardware meters on page 117.*

You can change channel assignments from the **Audio** tab in the Preferences dialog or the Channel Meters window. Changing the setting in either location updates your preferences and affects all open data windows. *For information about using the* **Audio** *tab of the Preferences dialog to enable and map channels, see Audio tab on page 342.*

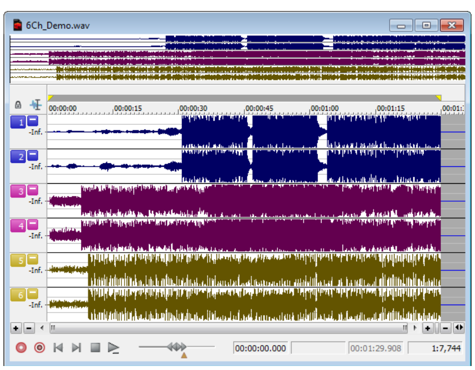
To change a channel's output device using the Channel Meters window, click the channel number and choose a new output port from the menu.

Opening and editing multichannel audio files

If you've used Sound Forge to edit stereo files before, you already know everything you need to know to edit multichannel files.

You can open multichannel audio files just like any other supported media type. *For more information, see Getting media files on page 61.*

When you open the file, you'll notice that the data window displays the channels as stereo pairs:



You can then edit the file just as you would any mono or stereo file.

Click the **Minimize** button ( ) to reduce the height of individual channels, or click the **Restore** button ( ) to restore a channel's height. Hold **Shift** while clicking a **Minimize** button ( ) to minimize all channels except the one you clicked.

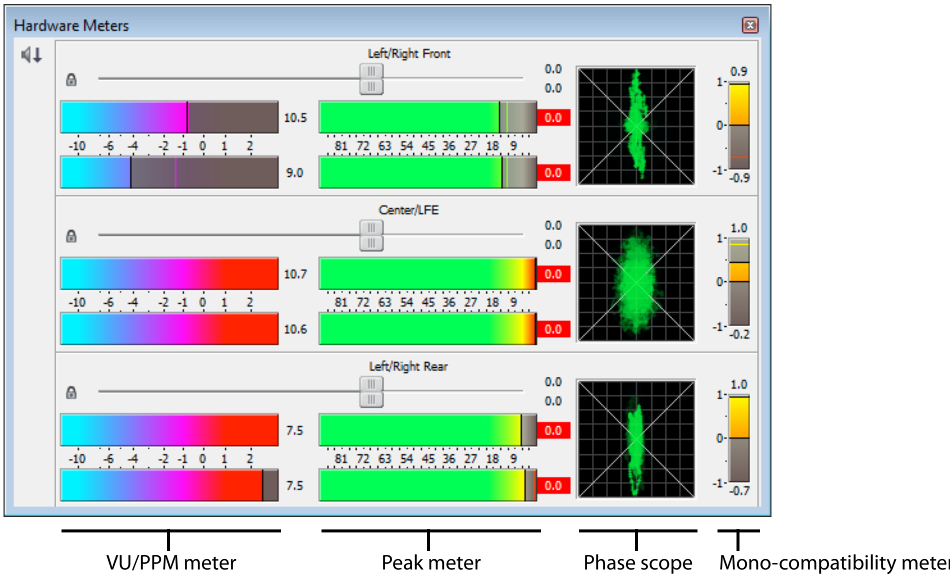
> **Tip:** *You can use the* **Display** *tab in the Preferences dialog to change the colors used to represent each channel. For more information, see Display tab on page 333.*

Recording multichannel audio files

With Sound Forge, you have the ability to record multichannel audio if your hardware supports this feature. *For more information, see Recording multichannel audio on page 154.*

Using the hardware meters

From the **View** menu, choose **Hardware Meters** to toggle the display of the Hardware Meters window. You can use this window to adjust the levels of your audio device's hardware outputs for monitoring and to view a peak meter, VU/PPM meter, phase scope, and mono-compatibility meter.



Adjusting output levels

The Hardware Meters window displays a gain fader for each output that is enabled on the **Audio** tab of the Preferences dialog. *For more information, see Audio tab on page 342.* You can use these faders to adjust preview levels.

Important: *The faders in the Hardware Meters window are used to control preview volume only. If you want to mix channel levels, use the Volume or Envelope plug-in.*

Drag to adjust the volume of the channel. Double-click the center of the thumb to reset the fader to 0.0 dB.

If the right and left channels are set differently, you can double-click either thumb to force the other channel to match it.

Click the **Lock/Unlock Fader Channels** button ( ) to lock (gang) the faders so the left and right channels will always move together. Click again to unlock the faders.

Tip: *Hold Shift while dragging a fader to temporarily override the current state of the Lock/Unlock Fader Channels button: if the button is turned off, you can hold Shift to drag the faders in locked mode; if the button is selected, hold Shift to drag the faders independently.*

Showing or hiding meters

You can display a peak meter, VU/PPM, a phase scope, and mono-compatibility meter for each hardware output. To toggle the display of each meter, right-click the Hardware Meters window and choose a command from the shortcut menu.

A check mark is displayed to indicate which meters are currently visible.

For more information about peak meters, see Channel Meters on page 135.

For more information about VU/PPM meters, see VU and Peak Program Meters on page 138.

For more information about phase scopes, see Phase Scope on page 139.

For more information about mono-compatibility meters, see Mono-Compatibility Meter on page 139.

Chapter 8

Using markers, regions, and commands

Markers and regions serve as reference points along the timeline. You can use markers for annotations, to insert metadata commands, or for MIDI triggers.

Using markers

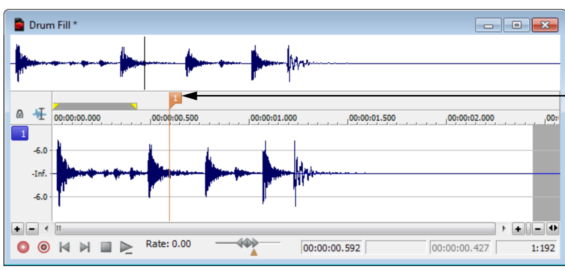
From the Insert menu, choose **Marker** to add a marker at the current cursor position. Markers are reference points you can place throughout a file. You can use markers to identify positions for editing or to seek forward and back within a streaming media file.

Markers can be quickly selected from the list in the Go To dialog. Also, markers are displayed in the Regions List for quick playback.

Inserting a marker

1. Position the cursor where you want to add a marker.
2. From the Insert menu, choose **Marker**. A marker () will be added at the cursor position.
3. If you want to name the marker, right-click the tag and choose **Rename** from the shortcut menu. Type a name for the marker in the edit box and press Enter.

> **Tip:** *You can also insert markers during playback by pressing the M key.*



Naming or renaming a marker

- Right-click the marker tag () and choose **Rename** from the shortcut menu. Type the name of the marker in the edit box and press Enter when you're finished.
- Double-click to the right of the marker and type a name in the edit box.

Moving a marker

Drag the marker tag () to a new location.

Markers will snap to other markers, regions, and command markers. Hold Shift while dragging to override snapping.

Deleting a marker

Right-click the marker tag () and choose **Delete** from the shortcut menu.

Deleting all markers and regions

Right-click above the loop region, choose **Markers/Regions**, and choose **Delete All** from the submenu. All markers () and regions () are removed.

Deleting all markers and regions within the selected area

Right-click above the loop region, choose **Markers/Regions**, and choose **Delete All in Selection** from the submenu. All markers ( ) and regions ( ) in the selected area are removed.

Previewing a marker

Click a marker's **Play** button ( ) in the Regions list.

Triggering a marker using MIDI commands

1. Right-click the marker tag and choose **Edit** from the shortcut menu. The Regions List window is displayed.
2. Click the down arrow in the marker's **Trigger** column to display a drop-down list.

Trigger type	Description
Note On - Play	The marker will be played when the specified note on message is received and will play for the full length of the file.
Note On - Play / Off - Stop	The marker will be played when the specified note on message is received and will stop when the full file is played or the specified Note Off message is received.
Note On - Queue / Off - Play	The marker will be queued for play when the specified note on message is received and will play when the corresponding Note Off message is received. This is used to reduce the time between receiving a trigger and playing a marker.

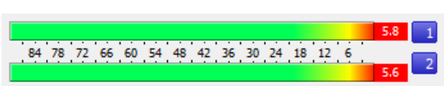
3. In the **Chan** box, specify the MIDI input channel for triggering.
4. In the **Note** box, specify the MIDI note that will trigger region playback. This value can be entered as a MIDI note value such as C4 or as a MIDI note number such as 60.

Notes:

- *If the Trigger from MIDI Timecode command is selected while using this dialog, you can auto-complete the* **Chan** *and* **Note** *values by pressing a key on your MIDI keyboard.*
- *Triggers in the Regions List function differently from triggers specified in the MIDI Triggers dialog and the Playlist. When using triggers in the Playlist, Regions List, or MIDI Triggers dialog, be aware that they can interact to create unexpected results. Sound Forge software first looks at the MIDI Triggers, then the Regions List, and then the Playlist when determining what to do when a MIDI command is detected. If you only want to use the triggers in the Regions List, turn off all the triggers in the MIDI Triggers dialog and the Playlist.*

Detecting and marking clipping

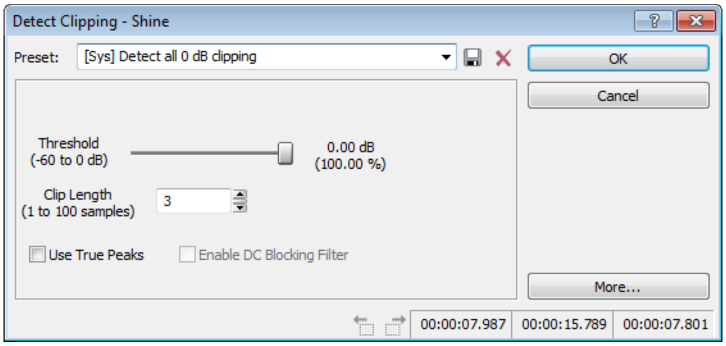
The clip indicators in the channel meters help you determine whether clipping occurs in your file, and you can use the **Find** command to find audio that matches levels you specify. For more control, however, you can use the detect clipping tool.



From the Tools menu, choose **Detect Clipping** to scan a selection of audio for clipping and add markers where clipping occurs.

Markers can be quickly selected from the list in the Go To dialog. Also, markers are displayed in the Regions List for quick playback.

1. Select the audio you want to scan.
2. From the Tools menu, choose **Detect Clipping**. The Detect Clipping dialog is displayed.



3. Choose a setting from the **Preset** drop-down list or adjust the controls as necessary.
   a. Drag the **Threshold** fader to determine the sound level you want to find.
   b. Set a value in the **Clip Length** box to specify how many sequential samples must meet the **Threshold** setting to constitute clipping.
   c. Select the **Use True Peaks** check box if you want to detect clipping of true peaks in dB FS for loudness measurements.

Tip: *Please note that true peaks are calculated using a higher sample rate than peaks in the Channel Meters for increased accuracy.*

   d. When the **Use True Peaks** check box is selected, select the **Enable DC Blocking Filter** check box if you want to filter the true peak sample value to compensate for asymmetrical audio signals or DC offset. Filtered true peaks are calculated as the maximum of the filtered and unfiltered signals.
4. Click the **OK** button.

   The selection is scanned, and markers are added whenever the software finds a number of sequential samples (determined by the **Clip Length** setting) with the same value above the **Threshold** setting.

Tip: *Use the* **Detect all clip-related plateaus** *preset to detect clipped peaks that may exist in your file after decreasing the levels in the file. You can then use the Pencil tool or the Clipped Peak Restoration tool in the Sony Noise Reduction plug-in to restore the clipped peaks.*

Using regions

From the Insert menu, choose **Region** to add region markers at each end of the current selection. Regions can be used to indicate sections of projects such as choruses or verses, or they can be used to make notes in the project.

The Regions List window contains all of the regions and markers that exist in the active data window.

Inserting a region

1. Drag the cursor in the data window or marker bar to make a time selection.
2. From the Insert menu, choose **Region**. Numbered region markers () are placed at the start and end of the selected area.
3. If you want to name the region, right-click the tag and choose **Rename** from the shortcut menu. Type a name for the region in the edit box and press Enter.

Tip: *You can also insert regions by pressing the R key (or Ctrl+Alt+R when the Event tool is selected).*

Naming or renaming a region

- Right-click the starting region marker (2) and choose **Rename** from the shortcut menu. Type the name of the region in the edit box and press Enter when you're finished.
- Double-click to the right of the region marker and type a name in the edit box.

Selecting a region

- Right-click the starting or ending region marker (2) and choose **Select Region** from the shortcut menu. The region is highlighted.
- Double-click the start or end region marker. The region is highlighted.

Moving a region

Drag either region tag (2) to move the tab and change the region's size.

Hold the Alt key while dragging either region tag to move a region and preserve its length.

Regions will snap to other markers, regions, and command markers. Hold Shift while dragging to override snapping.

Deleting a region

Right-click the region marker (2) and choose **Delete** from the shortcut menu.

Deleting all markers and regions

Right-click above the loop region, choose **Markers/Regions**, and choose **Delete All** from the submenu. All markers (1) and regions (2) are removed.

Deleting all markers and regions within the selected area

Right-click above the loop region, choose **Markers/Regions**, and choose **Delete All in Selection** from the submenu. AAll markers (1) and regions (2) in the selected area are removed.

Previewing a region

Click a region's **Play** button () in the Regions list.

Triggering a region using MIDI commands

1. Right-click the marker tag and choose **Edit** from the shortcut menu. The Regions List is displayed.
2. Click the down arrow in the region's **Trigger** column to display a drop-down list.

Trigger type	Description
Note On - Play	The region will be played when the specified note on message is received and will play for the full length of the file.
Note On - Play / Off - Stop	The region will be played when the specified note on message is received and will stop when the full file is played or the specified Note Off message is received.
Note On - Queue / Off - Play	The region will be queued for play when the specified note on message is received and will play when the corresponding Note Off message is received. This is used to reduce the time between receiving a trigger and playing a region.

3. In the **Chan** box, specify the MIDI input channel for triggering.

4. In the **Note** box, specify the MIDI note that will trigger region playback. This value can be entered as a MIDI note value such as C4 or as a MIDI note number such as 60.

Notes:

- *If the Trigger from MIDI Timecode command is selected while using this dialog, you can auto-complete the* **Chan** *and* **Note** *values by pressing a key on your MIDI keyboard.*
- *Triggers in the Regions List function differently from triggers specified in the MIDI Triggers dialog and the Playlist. When using triggers in the Playlist, Regions List, or MIDI Triggers dialog, be aware that they can interact to create unexpected results. Sound Forge software first looks at the MIDI Triggers, then the Regions List, and then the Playlist when determining what to do when a MIDI command is detected. If you only want to use the triggers in the Regions List, turn off all the triggers in the MIDI Triggers dialog and the Playlist.*

Converting markers to regions

From the Edit menu, choose **Regions List**, and then choose **Markers to Regions** from the submenu. All existing markers will be converted to regions using the data between each consecutive marker as the region boundary.

For example, if your file contains three markers, this command will create two regions; the first region will span the area between the first and second markers, and the second region will span the area between the second and third markers.

Tip: *Right-click the Regions List and choose* **Markers to Regions** *from the shortcut menu.*

Creating regions automatically

The Auto Region dialog allows you to automatically create regions in a sound file for the Regions List and Playlist. To display this dialog, choose **Auto Region** from the Tools menu.

Regions can be detected according to fast sound attacks (such as drum beats or words) or according to the **Selection tempo** value specified in the File Properties window.

Creating regions based on fast attacks

1. From the Tools menu, choose **Auto Region**.
2. Drag the **Attack sensitivity** slider to determine how sensitive the attack-detection algorithm is to fast increases in volume.

   With a high setting, regions are created when the sound level increases by very small amounts, and more regions are created. With a low setting, the sound level must increase by a large amount before a new region is created, and fewer regions are created.
3. Drag the **Release sensitivity** slider to determine the minimum decrease in sound level that must occur before a region end is created.

   With a high setting, regions are created when the sound level decreases by very small amounts, and more regions are created. With a low setting, the sound level must decrease by a large amount before a new region is created, and fewer regions are created. A low setting is useful if you want regions to be created after quiet breaks.
4. Drag the **Minimum level** fader to determine the threshold sound level that must be found before a new region is created.

   With a high setting, only high-level sounds will trigger the creation of a new region. This is useful if you want loud instrument attacks in a song (such as the bass drum) to mark the beginning of a region, since they often correspond to the beginning of a measure or beat. Low threshold settings will also allow low-level sound attacks to create new regions.
5. In the **Minimum beat duration** box, specify the minimum length that must elapse before a new region can be created.

   A low setting will allow very short regions to be created if sound attacks occur in fast succession. This is useful for uptempo music. A higher setting will prevent quick sound attacks from being separated into different regions.
6. Select the **Use release point for end of region** check box to end a region when the sound level drops by a factor determined by the **Release sensitivity**. This is useful if you don’t want the silence between sounds or phrases to be included in the regions.

   When this check box is cleared, region ends are only created when attacks are detected.
7. Click the **OK** button.

Creating regions based on a musical time interval

When you select the **Build regions using the current tempo** check box, regions are created according to the file's current tempo.

1. Use the Edit Tempo dialog to edit or calculate the musical tempo of your file.
2. From the Tools menu, choose **Auto Region**.
3. Select the **Build regions using the current tempo** check box.
4. Use the **Measures** and **Beats** boxes to specify the interval between regions.

   For example, if you want a region to be created at every beat, set **Beats** to 1 and **Measures** to 0. To create a region at every measure, set **Measures** to 1 and **Beats** to 0.

For more information about using processing dialog controls, see *Processing Audio on page 181*.

Extracting regions to new files

From the Tools menu, choose **Extract Regions** to create new files from regions in the Regions List.

1. From the Tools menu, choose **Extract Regions**. The Extract Regions dialog is displayed.
2. In the **Regions to extract** box, select the regions you want to extract. You can hold the Ctrl or Shift keys to select multiple regions.
3. In the **Destination folder** box, specify the folder where the extracted regions will be saved, or click the **Browse** button to choose a new folder.
4. Type a name in the **File name prefix** box if you want add a prefix to extracted regions. For example, enter Test to extract the files Test Region 001.wav, Test Region 002.wav, Test Region 003.wav, and so on.

Note: *Select the* **Use long file names for destination file names** *check box to allow file names of up to 128 characters including spaces. The files names will consist of the value in the* **File name prefix** *box and the region name.*

When this check box is cleared, file names will conform to the 8.3 naming convention. These names consist of the first 5 characters from the **File name prefix** *and a unique three-digit number starting with the number specified in the* **Start file counter index** *box. For example, if you have 4 regions selected for extraction, and your prefix is set to PREFIX, the names used will be PREFI000.wav, PREFI001.wav, PREFI002.wav PREFI003.wav.*

5. Click the **Extract** button to extract the selected regions.

Updating marker or region positions

From the Edit menu, choose **Regions List**, and then choose **Update** from the submenu to move a marker or region to match the current cursor position or selection.

Tip: *To update a marker quickly, right-click the marker ( ) or region tag ( ) and choose* **Update** *from the submenu.*

1. If the Regions List isn't already visible, choose **View** > **Metadata** > **Regions List**.
2. In the Regions List window, select the marker or region you want to update.
3. In the data window, indicate the new position of the marker or region:
   - If you want to update a marker's position, place the cursor where you want to move the marker.
   - If you want to update a region's position, select a range of data in the data window.
4. From the Edit menu, choose **Regions List**, and then choose **Update** from the submenu:
   - If you are updating a marker, the marker moves to the cursor position. If you have a range of data selected, the cursor will blink at one end of a selection; press the Home key to move the cursor to the beginning of the selection, or press End to move to the end of the selection.

- If you are updating a region, the beginning and ending points of the region are moved to match the current selection, and the length of the region is modified if the selection is a different length than the original region.

Tip: *Hold Alt while dragging a region tag to move a region while preserving its length.*

Replicating markers or regions

Replicating a marker or region creates an exact copy of an existing marker or region in a file. You can use the Regions List window to modify the properties of the replicated entry.

1. Select an entry in the Regions List.

   If the Regions List is not visible, choose **Regions List** from the View menu.
2. From the Edit menu, choose **Regions List** and choose **Replicate** from the submenu.

Tip: *Right-click an entry in the Regions List and choose* **Replicate** *from the shortcut menu.*

Deleting markers or regions

From the Edit menu, choose **Regions List**, and then choose **Delete** from the submenu to remove the selected marker or region from the Regions List window.

Tips:

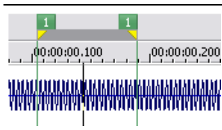
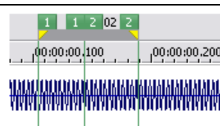
- *Select a region or marker in the Regions List window and press Delete.*
- *Right-click above the loop region, choose* **Markers/Regions***, and choose* **Delete All** *or* **Delete All in Selection** *from the submenu.*

Splitting regions

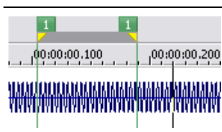
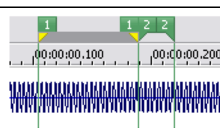
Splitting a region divides an existing region at the current cursor position, producing two separate regions.

1. In the Regions List window, select the region you want to split.
2. In the data window, position the cursor where you want the split to occur.
3. From the Edit menu, choose **Regions List**, and choose **Split** from the submenu (or right-click a region in the Regions List window and choose **Split** from the shortcut menu).

   If the cursor is placed within the region you're splitting, the region will be split at the cursor position.

	
Before split	After split

   If the cursor is placed outside the region you're splitting, a new region will be created from the closest edge of the original region to the cursor position.

	
Before split	After split

Saving a regions/playlist file

You can save a file's regions list and playlist/cutlist to an external file. This command offers the flexibility of using multiple playlists for the same sound file.

1. From the Edit menu, choose **Regions List** or **Playlist/Cutlist**, and then choose **Save As** from the submenu.

> **Tip:** *Right-click the Regions List or Playlist/Cutlist window and choose* **Save As** *from the shortcut menu.*

2. Use the Save Regions/Playlist As dialog to specify a folder and file name.
3. Specify the type of regions you want to save from the **Save as type** drop-down list:
   - Choose **Playlist File (.sfl)** to save a Sound Forge regions/playlist file.
   - Choose **Session 8 File (.prm)** to save a file that supports both Session 8 and Sound Forge regions.
   - Choose **Windows Media Script File (.txt)** to save a file that includes Windows Media script commands.
   - Choose **Text File (Tab delimited) (.txt)** to save markers and regions to a plain text file.
4. Click the **Save** button.

Importing a regions/playlist file

From the Edit menu, choose **Regions List** or **Playlist/Cutlist**, and then choose **Open** from the submenu to import an existing regions/playlist file (.sfl) into the current sound file. This command offers the flexibility of using multiple playlists for the same sound file.

1. From the Edit menu, choose **Regions List** or **Playlist/Cutlist**, and then choose **Open** from the submenu.

> **Tip:** *Right-click the Regions List or Playlist/Cutlist window and choose* **Open** *from the shortcut menu.*

2. Use the Open Regions/Playlist dialog to locate an existing regions/playlist file.
3. Specify the type of regions you want to import from the **Files of type** drop-down list:
   - Choose **Playlist File (.sfl)** to import a Sound Forge regions/playlist file.
   - Choose **Session 8 File (.prm)** to import a file that supports both Session 8 and Sound Forge regions.
   - Choose **Windows Media Script File (.txt)** to import a file that includes Windows Media script commands.
   - Choose **Text File (Tab delimited) (.txt)** to import a plain text file.
   - Choose **Wave File (.wav)** to import markers and regions from another sound file.
4. Click the **Open** button.

> **Note:** *Opening a new regions/playlist file will clear the current regions and playlist entries. Make sure you have saved the current regions/playlist before continuing.*

Copying regions to the clipboard

From the Edit menu, choose **Regions List** and choose **Copy onto Clipboard** to copy the text of the Regions List onto the clipboard for use with a text editor.

Editing a regions list in a text editor allows you to make an annotated list that you can print for reference.

> **Tip:** *Right-click the Regions List and choose* **Copy onto Clipboard** *from the shortcut menu.*

Locking loop and region lengths

From the Options menu, choose **Lock Loop/Region Length** to force the length of a region to remain constant when changing the start or end time of a region or loop.

> **Tip:** *When this option is not selected, you can hold the Alt key while dragging region markers ( ) to lock the length of a region. To move a loop without changing its length, drag the bar between the loop markers.*

Clearing markers and regions

From the Edit menu, choose **Regions List** and choose **Clear All** from the submenu to remove all markers and regions from the current file. This command will also clear the Playlist/Cutlist window.

> **Tip:** *Right-click above the loop region, choose* **Markers/Regions**, *and choose* **Delete All** *or* **Delete All in Selection** *from the submenu.*

Using commands

From the Insert menu, choose **Command** to place a metadata command marker at the current cursor position.

Command markers indicate when an instruction (function) will occur in a streaming media file. You can use command markers to display headlines, captions, link to Web sites, or any other function you define.

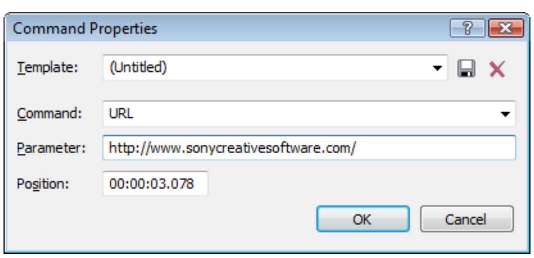
> **Important:** *Windows Media Player 9 and later will ignore metadata commands unless the* **Run script commands when present** *check box is selected on the Security tab of the player's Preferences dialog. Be sure to instruct your audience to select this check box before playing your file.*

Inserting a command marker

1. Place the cursor where you want to insert the command marker.

> **Note:** *Commands snap to the nearest millisecond.*

2. From the Insert menu, choose **Command**. The Command Properties dialog is displayed.

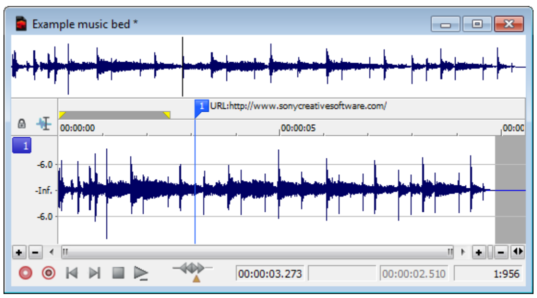


3. From the **Command** drop-down list, choose the type of command you want to insert, or type a custom command in the box.
4. In the **Parameter** box, enter the argument that should be passed to the command. For example, if you're using an URL command, enter the address of the Web page you want to display.

Command	Player type	Description
URL	Windows Media	Indicates when an instruction is sent to the user's Internet browser to change the content being displayed. In the **Parameter** box, enter the URL that will display at a specific time during the rendered project's playback.

Command	Player type	Description
Text	Windows Media	Displays text in the captioning area of the Windows Media Player located below the video display area. In the **Parameter** box, enter the text that will display during playback. **Note:** *To view captions during playback in Windows Media Player 9, choose* **Captions and Subtitles** *from the Windows Media Player Play menu, and then choose* **On if Available** *from the submenu.*
WMClosedCaption	Windows Media	Displays the text from the **Parameter** box in the captioning window that is defined by an HTML layout file.
WMTextBodyText	Windows Media	Displays the text from the **Parameter** box in the text window that is defined by an HTML layout file.
WMTextHeadline	Windows Media	Displays the text from the **Parameter** box in the headline window that is defined by an HTML layout file.

5. In the **Position** box, type the time you want the command to occur in your project. The command is inserted at the cursor position by default.



Deleting a command marker

Right-click the command marker tag () and choose **Delete** from the shortcut menu.

Editing a command marker

- Right-click the command marker tag () and choose **Edit** from the shortcut menu.
- Double-click the command marker tag.

Moving the cursor to a command marker

Click the command marker tag ().

Using command templates

If you frequently insert commands that use similar settings, you can create a template to insert command settings automatically.

Creating a template

1. From the Insert menu, choose **Command** to display the Command Properties dialog.
2. Enter the settings you want to use in the **Command**, **Parameter**, and **Position** boxes.
3. In the **Template** box, enter the name you want to use to store the template.
4. Click the **Save** button ().

Recalling a template

1. From the Insert menu, choose **Command** to display the Command Properties dialog.
2. Choose the template you want to use from the **Template** drop-down list. The **Command**, **Parameter**, and **Position** boxes are automatically filled in using the information stored in the template.
3. Edit the settings in the **Command**, **Parameter**, and **Position** boxes as necessary.
4. Click **OK**.

Editing metadata commands

Your metadata command templates are saved in the following file: **C:\Users\[user name]\AppData\Local\Sony\Sound Forge Pro\11.0\cmdtemp.xml**.

You can edit this file directly to modify your templates.

Using the Regions List

The Regions List contains information pertaining to all regions in the current data window. The Regions List information can be saved as metadata in most file types. You also have the option of saving the Regions List to an external Playlist file.

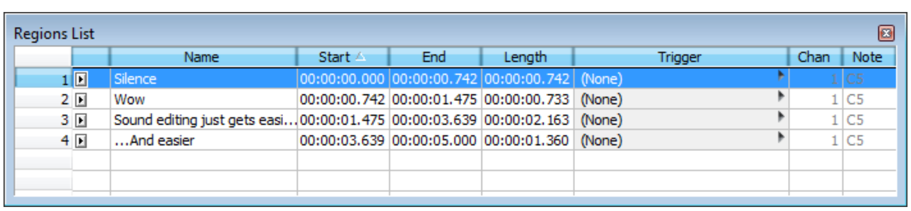
Displaying the Regions List

1. Open the Voiceover.pca file. This file is located in the same folder as the application.
2. From the **View** menu, choose **Metadata**, and then choose **Regions List** from the submenu (or press Ctrl+Alt+M, 0). The Regions List for Voiceover.pca appears.

Working with the Regions List

By default, the Regions List displays the following information for each region in the current data window:

- A small **Play** button ( ) dedicated to the region.
- The region's name.
- The region's start point.
- The region's end point.
- The region's length.
- The region's trigger.
- The region's channel.
- The region's note.


Regions List

	Name	Start	End	Length	Trigger	Chan	Note
1	Silence	00:00:00.000	00:00:00.742	00:00:00.742	(None)	1	C5
2	Wow	00:00:00.742	00:00:01.475	00:00:00.733	(None)	1	C5
3	Sound editing just gets easi…	00:00:01.475	00:00:03.639	00:00:02.163	(None)	1	C5
4	…And easier	00:00:03.639	00:00:05.000	00:00:01.360	(None)	1	C5

Changing region order

By default, the Regions List displays regions in alphabetical order by name, but you can specify an alternate order by clicking the column heading to sort in ascending (Start) or descending (Start) order.

Saving a Regions List file

You can save a file's Regions List to an external file. This offers the flexibility of using multiple Regions Lists for the same audio file.

1. From the **Edit** menu, choose **Regions List**, and choose **Save As** from the submenu (or right-click the Regions List and choose **Save As** from the shortcut menu).
2. Use the Save Regions/Playlist As dialog to specify a folder and file name.
3. Click **Save**.

Opening a Regions List file

Importing a Regions List file offers the flexibility of using multiple Regions List files for the same audio file. Opening a new Regions List file clears the current Regions List. Make sure you have saved the current Regions List before continuing.

1. From the **Edit** menu, choose **Regions List**, and choose **Open** from the submenu (or right-click the Regions List and choose **Open** from the shortcut menu).
2. Use the Open Regions/Playlist dialog to locate an existing file.
3. Specify the type of regions you want to import from the **Files of type** drop-down list:
   - Choose **Playlist File (.sfl)** to import a Sound Forge regions/Playlist file.
   - Choose **Session 8 File (.prm)** to import a file that supports both Session 8 and Sound Forge regions.
   - Choose **Windows Media Script File (.txt)** to import a file that includes Windows Media script commands.
   - Choose **Wave File (.wav)** to import markers and regions from another audio file.
4. Click **Open**.

Copying the Regions List to the clipboard

Editing a Regions List in a text editor allows you to make an annotated list that you can print for reference.

From the **Edit** menu, choose **Regions List**, and then choose **Copy onto Clipboard** (or right-click the Regions List and choose **Copy onto Clipboard** from the shortcut menu). The list is copied to the Windows clipboard.

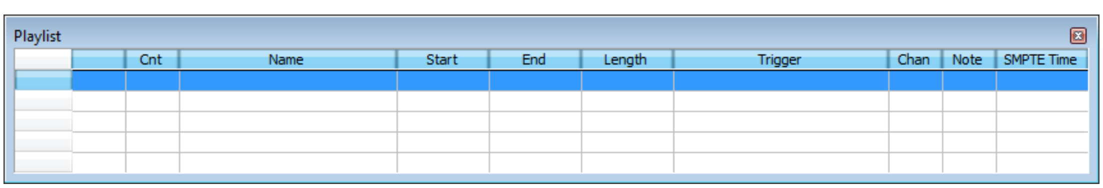
Using the Playlist

After you create regions, you can arrange them in the Playlist. Unlike the Regions List, which displays its contents in alphabetical or chronological order, the Playlist displays and plays its regions in a user-specified arrangement. In addition, you can rearrange and audition regions endlessly in the Playlist without performing a destructive edit when you save the file.

As with the Regions List, you can save the Playlist information as metadata in most file types. You also have the option of saving the Playlist to an external Playlist file.

Displaying the Playlist

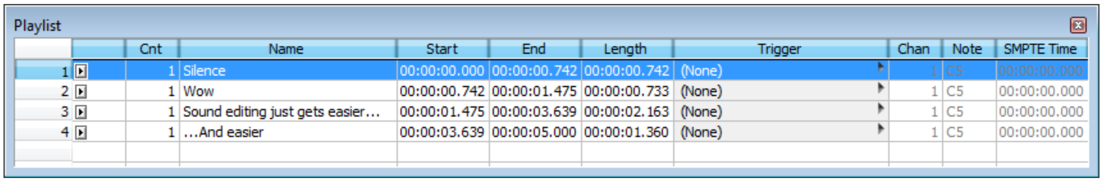
1. Open the Voiceover.pca file. This file is located in the same folder as the application.
2. From the **View** menu, choose **Metadata**, and then choose **Regions List** from the submenu (or press Ctrl+Alt+M, 0). The Regions List window for Voiceover.pca appears.
3. From the **View** menu, choose **Metadata**, and then choose **Playlist** from the submenu (or press Ctrl+Alt+M, 1). The Playlist window for Voiceover.pca appears.



Notice that the file contains regions, but the Playlist is empty. You must add regions to the Playlist before arranging them.

Understanding the Playlist display

When you add a region to the Playlist, its appearance is similar to its appearance in the Regions List, with the exception of the **Count (Cnt)** column. Located to the left of the **Name** column, the **Count (Cnt)** column displays the number of times the corresponding region plays before the Playlist proceeds to the next region.



Adding regions to the Playlist

You can add regions from the Regions List to the Playlist using commands or drag-and-drop. You can also add regions to the Playlist directly from the data window.

Adding regions to the Playlist using commands

1. Select a region in the Regions List.
2. From the **Edit** menu, choose **Playlist/Cutlist**, and choose **Add** from the submenu (or right-click a region in the Regions List window and choose **Add to Playlist** from the shortcut menu). The region is added to the Playlist.

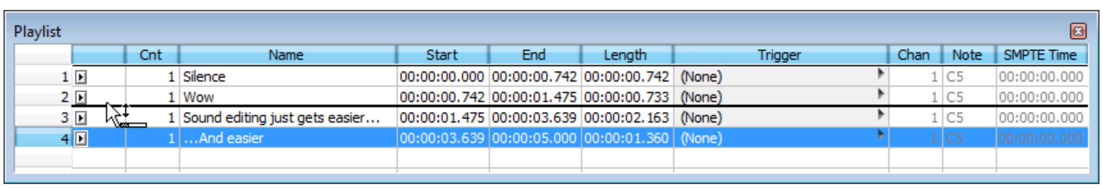
Adding regions to the Playlist using drag-and-drop

1. Select a region in the Regions List.
2. Drag the region into the Playlist.
3. Release the mouse button.

Arranging the Playlist

Moving regions

After you have added regions to the Playlist, you can arrange them using drag-and-drop.



Replicating a region in the Playlist

A major advantage of arranging the Playlist is the ability to repeat a region in multiple places without actually copying the audio data. This feature is called replicating.

1. Right-click the region to be replicated and choose **Replicate** from the shortcut menu. The region is replicated in the Playlist.
2. Drag the replicated region to its new position in the Playlist.

Deleting a region from the Playlist

You can delete regions from the Playlist without affecting the audio file.

1. Select the region that you would like to delete.
2. From the **Edit** menu, choose **Playlist/Cutlist**, and then choose **Delete** from the submenu (or right-click the Playlist window and choose **Delete** from the shortcut menu).

Editing a Playlist/Cutlist region

You can edit a Playlist/Cutlist region by typing new values in the **Cnt**, **Trigger**, **Chan**, **Note**, and **SMPTE Time** boxes.

Repeating a region during Playlist playback

You can specify the number of times a region repeats during Playlist playback.

Type a value in the **Cnt** box in the Playlist window to specify the number of times the Playlist region will repeat before playing the next region.

Using stop points

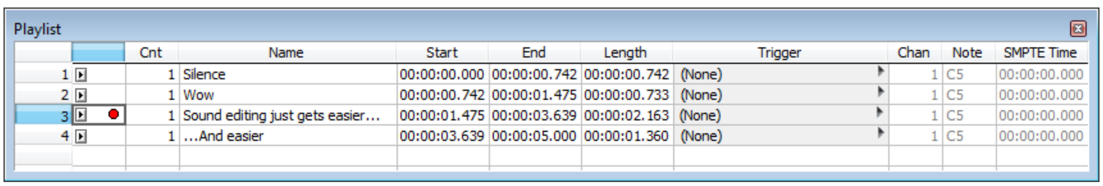
You can attach stop points to regions in the Playlist. When a stop point is encountered during playback, the corresponding region is repeated the number of times specified by the **Count** value and playback is halted.

Creating a stop point

Perform one of the following actions to set the stop point for a Playlist:

- Right-click a region in the Playlist window and choose **Stop Point** from the shortcut menu.
- Select a region in the Playlist window and press Ctrl+E or Ctrl+8 (not on the numeric keypad).

A check mark appears adjacent to the comment in the shortcut menu and a stop point (indicated by a red circle) appears in the Playlist.



> **Note:** *When you play your Playlist, it will continue to play through the regions until it encounters a stop point. This is useful when triggering playback from incoming MIDI or timecode and you only want certain sections of the Playlist to be played at a time.*

Deleting a stop point

Perform one of the following actions to remove the stop point for a Playlist:

- Right-click a region in the Playlist window and choose **Stop Point** from the shortcut menu.
- Select a region in the Playlist window and press Ctrl+E or Ctrl+8 (not on the numeric keypad).

The corresponding check mark is cleared from the shortcut menu and the stop point (indicated by a red circle) is removed from the Playlist.

Playing from the Playlist

The Playlist displays the sequential order in which regions play. To play a region, click the corresponding **Play** button. Playback begins with the selected region and continues through the end of the Playlist, playing a region multiple times when instructed by the **Count** value.

> **Note:** *Playback is interrupted if a stop point is present. For more information, see Using stop points on page 132.*

Creating a new file from the Playlist

After you have auditioned and arranged all regions in the Playlist, you can create a new file based on the Playlist arrangement. To create a new file from the Playlist, right-click the Playlist and choose **Convert to New** from the shortcut menu.

> **Note:** *If the original file has both audio and video components (such as an AVI file), the new file created from the Playlist contains the audio portion only.*

Configuring the Playlist as a Cutlist

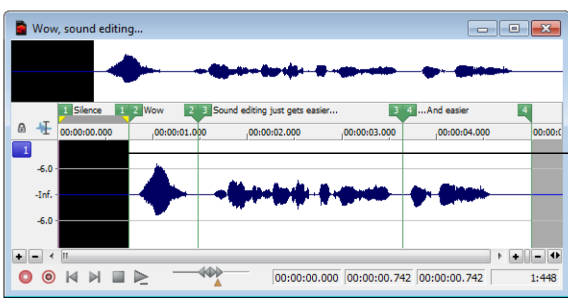
When trimming lengthy recordings, configuring the Playlist as a Cutlist can sometimes decrease editing time. In Play as Cutlist mode, the original file is played, but all regions placed on the Cutlist are ignored. Click the **Play as Cutlist** button ( ) on the playbar to enter Play as Cutlist mode.

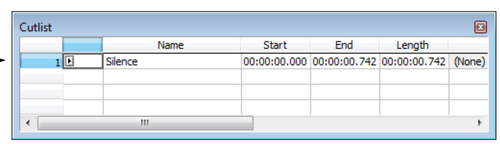
Treating the Playlist as a Cutlist

1. From the **View** menu, choose **Metadata**, and then choose **Playlist** from the submenu. The Playlist window is displayed.
2. Right-click the Playlist and choose **Treat as Cutlist** from the shortcut menu. A check mark appears adjacent to the command in the shortcut menu and the Cutlist appears. The **Play as Cutlist** button ( ) appears in the playbar.

Adding regions to the Cutlist

1. Open the Voiceover.pca file. This file is located in the same folder as the application.
2. Open the Cutlist.
3. Select the “Silence” region and drag it to the Cutlist (or press Delete). The region is added to the Cutlist and the selection area in the waveform display is shaded.





New region is added to the Cutlist

4. Click the **Play Cutlist** button ( ) on the data window’s playbar. The file plays with the Cutlisted region omitted.

Creating a new file from the Cutlist

Once all superfluous regions are placed in the Cutlist, you can create a new audio file and Regions List from the remaining region. From the **Edit** menu, choose **Playlist/Cutlist** and choose **Convert to New** from the submenu (or right-click the Cutlist and choose **Convert to New** from the shortcut menu).

Deleting all Cutlist regions

1. Select a region in the Playlist/Cutlist window. If the window is not visible, press Ctrl+Alt+M, 1.
2. From the **Edit** menu, choose **Playlist/Cutlist** and then choose **Delete** from the submenu (or press Delete).

Reverting to Playlist function

To use the Cutlist as a Playlist again, right-click the Cutlist and choose **Treat as Cutlist** from the shortcut menu. The check mark is cleared from the corresponding command in the shortcut menu and the Playlist function is restored.

Saving a Playlist/Cutlist file

You can save a file's Playlist/Cutlist to an external file. This offers the flexibility of using multiple Playlists for the same file.

1. From the **Edit** menu, choose **Regions List** or **Playlist/Cutlist**, and choose **Save As** from the submenu (or right-click the Playlist/Cutlist and choose **Save As** from the shortcut menu).
2. Use the Save Regions/Playlist dialog to specify a folder and file name.
3. Click **Save**.

Opening a Playlist/Cutlist file

Importing a Playlist file offers the flexibility of using multiple Playlists for a file. Opening a new Playlist file clears the current Playlist. Make sure you have saved the current Playlist before continuing.

1. From the **Edit** menu, choose **Playlist/Cutlist**, and choose **Open** from the submenu (or right-click the Playlist/Cutlist and choose **Open** from the shortcut menu).
2. Use the Open Regions/Playlist window to browse to an existing regions file.
3. Specify the type of file you want to import from the **Files of Type** drop-down list:
   - Choose **Playlist File (.sfl)** to import a Sound Forge Regions List/Playlist file.
   - Choose **Session 8 File (.prm)** to import a file that supports both Session 8 and Sound Forge regions.
   - Choose **Windows Media Script File (.txt)** to import a file that includes Windows Media script commands.
   - Choose **Wave File (.wav)** to import markers and regions from another sound file.
4. Click **Open**.

Copying the Playlist/Cutlist to the clipboard

Editing a Playlist/Cutlist in a text editor allows you to make an annotated list that you can print for reference.

From the **Edit** menu, choose **Playlist/Cutlist**, and then choose **Copy onto Clipboard** (or right-click the Playlist/Cutlist and choose **Copy onto Clipboard** from the shortcut menu). The list is copied to the Windows clipboard for use with a text editor.

Chapter 9

Monitoring levels in digital audio

The Sound Forge channel meters display peak levels during playback. Use the meters to monitor levels and ensure no clipping occurs in your file.

Decibels

The standard method for digital metering is to use the maximum possible sample value as a reference point. This value is referred to as 0 dB. Decibels are used to represent fractions logarithmically. In this case, the fraction is: sample amplitude divided by the maximum possible amplitude. The actual equation used to convert to decibels is: dB = 20 log (amplitude/32,768).

To illustrate this, consider a sine wave with a peak amplitude of 50 percent of full scale. Inserting the values in the appropriate places yields 20 log (0.50) = -6.0 dB. Each time a signal's amplitude is divided by two, its dB value is decreased by 6 dB. Likewise, doubling the amplitude of a signal increases its dB value by 6 dB. Dividing the sine wave until its peak amplitude is equal to 1 produces lowest peak dB possible, -90.3 dB.

Why are dBs used when talking about audio? Decibels are typically used when dealing with sound pressure levels because of the vast range of sound (about 120 dB) that the human ear can perceive. It's also easier to say -90 dB than 0.000030 (1/32,768).

Digital versus analog levels

When recording to an analog medium such as magnetic tape, recording engineers typically try to keep VU (volume unit) meters as close to zero as possible. This ensures a high signal-to-noise ratio while preserving adequate headroom to keep the tape from saturating and distorting. In addition, occasional peaks above 0 do not cause problems because the tape saturation point is not an absolute.

However, this is not true in the digital realm, where amplitudes are stored as discrete numbers instead of continuous variables. The flexible recording ceiling of analog is replaced by the absolute maximum sample values of digital audio. Stored signals must never have a value above these maximums, as the wave peaks are literally clipped. This clipping adds audible distortion and though it can go unnoticed, it can also ruin an entire project. Therefore, sample with the understanding that digital audio has absolutely no headroom.

Setting digital audio levels

Because digital audio has no headroom, setting the sampling level becomes critical. If the loudest section of the audio is identified in advance, the recording level should be set so that the peak is as close to 0 dB as possible to maximize the dynamic range of the digital medium. If the loudest section of audio is unknown, allow 3 to 6 dB of headroom for unexpected peaks.

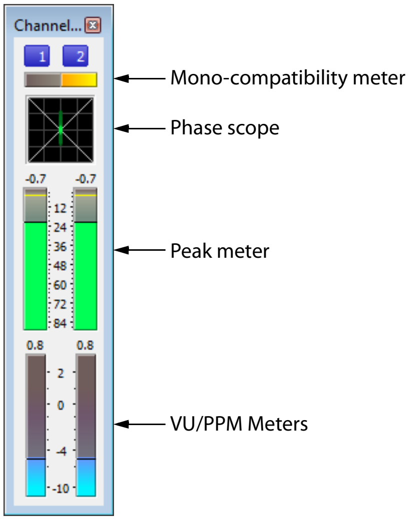
Tip: *From the Tools menu, choose* **Find** *and use the Find dialog to identify the largest peak in your file.*

Channel Meters

From the View menu, choose **Channel Meters** to open or close the channel meters. By default, Sound Forge software provides peak meters that you can use to monitor your audio levels. You can also choose to display VU/PPM (peak program) meters, a phase scope, and a mono-compatibility meter.

The peak meters display instantaneous levels during playback to help you determine the loudest level in your audio signal and whether the signal is clipping.

To prevent clipping, keep an eye on your peak meters. Peak levels should never exceed 0 dB. You can use the **Status** tab in the Preferences dialog to calibrate the VU/PPM meters to their associated levels on the peak meters and adjust the VU meters' sensitivity.



Showing or hiding the Channel Meters window

From the **View** menu, choose **Channel Meters** to open or close the channel meters window. You can dock the Channel Meters window on any edge of the Sound Forge workspace.

Showing or hiding meters

You can display a peak meter, VU/PPM, a phase scope, and mono-compatibility meter for each channel. To toggle the display of each meter, right-click the Channel Meters or Hardware Meters window and choose a command from the shortcut menu.

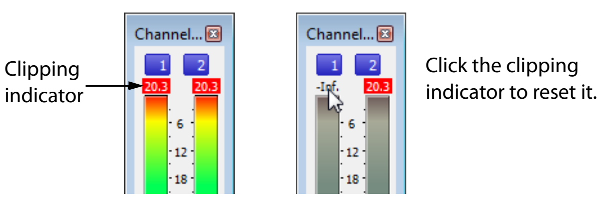
A check mark appears to indicate which meters are currently visible.

- *For more information about VU/PPM meters, see VU and Peak Program Meters on page 138.*
- *For more information about phase scopes, see Phase Scope on page 139.*
- *For more information about mono-compatibility meters, see Mono-Compatibility Meter on page 139.*

Resetting clipping indicators

When audio levels are too high, clipping can occur. A red indicator appears at the top of the meter to show when audio is clipping. Do any of the following to reset the indicator:

- From the **Options** menu, choose **Channel Meters**, and then choose **Reset Clip** from the submenu.
- Click to reset the indicator, or right-click the meters and choose **Reset Clip** from the shortcut menu.



You can also detect and mark clipped audio using the detect clipping tool. *For more information, see Detecting and marking clipping on page 120.*

Changing the meters' display resolution

The peak meters display levels in dB FS. To change the resolution of the meters, do either of the following:

- From the **Options** menu, choose **Channel Meters**, choose **Peak Range** from the submenu, and then choose a display range.
- Right-click the channel meter, choose **Peak Range** from the submenu, and then choose a display range.

Note: *Choosing a wide range allows you to see low-level signals at the expense of precision display at high levels.*

Changing the meters' display options

You can choose whether labels, peaks, and valleys are displayed in the meters and whether the meters are displayed on top of other windows when they are not docked.

Do either of the following to change the meters' display options:

- From the **Options** menu, choose **Channel Meters**, and then choose a command from the submenu.
- Right-click the meters and choose a command from the shortcut menu.

Command	Description
Expand Meters	Toggles expanded-width meters. Turning off expanded meters can conserve screen space.
Interleave Meters	Toggles interleaved or stacked display of VU/PPM meters with the corresponding channel meters.
Show Labels	Toggles the meter level labels on and off.
Hold Peaks	When selected, the highest peak levels are represented by a thin line on the meter.
Hold Valleys	When selected, the lowest peak levels are represented by a thin line on the meter.

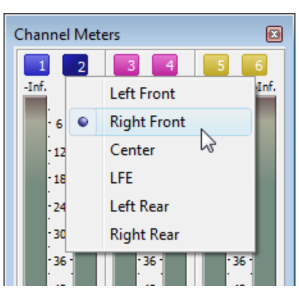
To change the layout of the meters in the Channel Meters window, right click the meters, choose **Layout** from the shortcut menu, and then choose a command from the submenu.

Command	Description
Horizontal/Vertical/Auto	Choose a command to change the orientation of the meters in the Channel Meters window.
Stretch to Fit	Stretches the meters to fit the window.
Narrow Width	Toggles narrow- or normal-width meters. Using narrow meters can conserve screen space.
Interleave Peak/VU	Toggles interleaved or stacked display of VU/PPM meters with the corresponding channel meters.

Routing channels to hardware outputs

You can change channel assignments from the **Audio** tab in the Preferences dialog or the Channel Meters window. Changing the setting in either location updates your preferences and affects all open data windows.

To change a channel's output device using the Channel Meters window, click the channel number and choose a new output port from the menu:



For more information about changing channel assignments in the Preferences dialog, see Audio tab on page 342.

VU and Peak Program Meters

You can display volume unit (VU) and peak program (PPM) meters in the Channel Meters and Hardware Meters windows to help you determine the perceived loudness of your audio signal (peak program meters provide faster response times to volume increases than VU meters). *For more information about channel meters, see Channel Meters on page 135. For more information about hardware meters, see Using the hardware meters on page 117.*

VU/PPM meters are especially helpful when you're mastering. Comparing two audio files' VU/PPM readings will help take the guesswork out of matching levels.

Right-click the Channel Meters or Hardware Meters window and choose **Show VU/PPM** from the shortcut menu to toggle the display of the VU/PPM meters.

VU/PPM readings should fall near the 0 (or reference) mark. 0 VU is merely a reference level, and your signal may exceed 0 VU. To prevent clipping, keep an eye on your peak meters. Peak levels should never exceed 0 dB. You can use the **Status** tab in the Preferences dialog to calibrate the VU/PPM meters to their associated levels on the peak meters and adjust the VU meters' sensitivity. *For more information, see Status tab on page 340.*

Choosing a VU or PPM scale

To change the scale of the meter, choose **Channel Meters** from the **Options** menu, choose **VU/PPM Scale**, and then choose a setting from the submenu (you can also right-click the meter to set its options).

VU and PPM scales are most useful for displaying the average volume of the signal. The meter represents the RMS average level during playback, and their attack and decay are not as sensitive as the peak meter.

PPM scales are useful for monitoring peak levels. The meters use a fixed integration time (5 or 10 ms) that is sensitive to increases in volume, but the meters are less sensitive to decreases in volume than the VU scales, which produces less meter activity and decreased eyestrain.

Item	Description
Traditional VU	The traditional VU meter is displayed with a scale of -10 dB to +2 dB. 0 dB on the VU meter equals 4 dBu.
Extended VU	The extended VU meter is displayed with a scale of -30 dB to +8 dB. 0 dB on the VU meter equals 4 dBu.
Logarithmic VU	Displays the meters in a logarithmic scale (like the Sound Forge peak meters) instead of the linear scales traditionally associated with VU meters.
UK PPM	The UK peak program meter (also known as a BBC meter) is a Type II meter and is displayed with a scale of 1 to 7, which corresponds to a range of -12 to 12 dBu: **UK Marks** — **dBu** 7 — 12 6 — 8 5 — 4 4 — 0 3 — -4 2 — -8 1 — -12
EBU PPM	The EBU peak program meter is a Type II meter and is displayed with a scale of -12 to +12, which corresponds to -12 dBu to 12 dBu. 0 on the EBU PPM equals 0 dBu. The EBU PPM and UK PPM respond identically to increases in volume, but the EBU PPM decays more slowly.
DIN PPM	The DIN peak program meter is a Type I meter and is displayed with a scale of -50 dB to +5 dB, which corresponds to -44 dBu to 11 dBu. 0 dB on the DIN PPM equals 6 dBu.
Nordic PPM	The Nordic peak program meter is a Type I meter and is displayed with a scale of -42 dB to +12 dB, which corresponds to -42 dBu to 12 dBu. 0 dB on the Nordic PPM equals 0 dBu.

Adjusting the VU/PPM meters' sensitivity

Unlike peak meters, which read instantaneous changes in your audio signal, the VU/PPM meters read a portion of the signal and calculate the average level. The size of the signal that the meters read is determined by the meters' integration time.

To set the amount of data surrounding the cursor that will be used to calculate levels in the VU meters, specify a value in the VU meter integration time box on the **Status** tab of the Preferences dialog.

The PPM scales use a fixed integration time:

Scale	Integration time
UK PPM	10 ms
EBU PPM	10 ms
DIN PPM	5 ms
Nordic PPM	5 ms

Phase Scope

You can display a phase scope in the Channel Meters and Hardware Meters windows to find phase cancellation among the channels in an audio file. *For more information on channel meters, see Channel Meters on page 135. For more information on hardware meters, see Using the hardware meters on page 117.*

Right-click the Channel Meters or Hardware Meters window and choose **Show Phase Scope** from the shortcut menu to toggle the display of the phase scope.

To change the display, right-click the Channel Meters or Hardware Meters window, choose **Phase Scope Style** from the shortcut menu, and then choose a setting from the submenu:

Style	Description
Lissajous - XY Plot	Displays the right and left channels plotted along the X and Y axes of the graph.
Lissajous - Rotated	Displays the right and left channels plotted along the X and Y axes of the graph. This setting is identical to the Lissajous - XY Plot setting, but the graph is rotated 45 degrees.
Polar - Linear Plot	Displays the right and left channels plotted vertically on the graph.
Polar - Circular Plot	Displays the right and left channels plotted on a circular graph.

Mono-Compatibility Meter

You can display a mono-compatibility meter in the Channel Meters and Hardware Meters windows to detect correlations or differences between the channels of a file that can cause phase cancellation when downmixing to mono. *For more information about channel meters, see Channel Meters on page 135. For more information about hardware meters, see Using the hardware meters on page 117.*

Right-click the Channel Meters or Hardware Meters window and choose **Show Mono-Compatibility Meter** from the shortcut menu to toggle the display of the mono-compatibility meter.

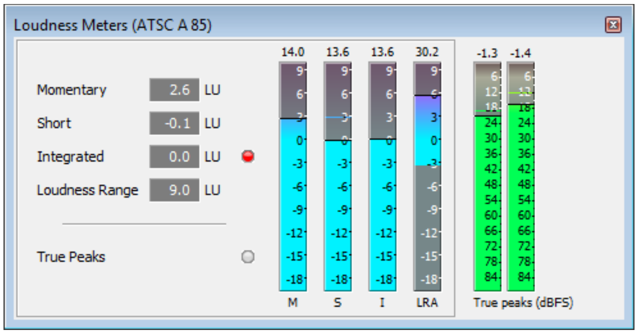
When the channels are similar, the right (or top) half of the meter is illuminated:

When the channels exhibit phase cancellations, the left (or bottom) half of the meter is illuminated:

Loudness Meters

From the View menu, choose **Loudness Meters** to display the Loudness Meters window.

The Loudness Meters tool provides data about an audio file's momentary loudness, short-term loudness, integrated (overall) loudness, and loudness range. You can use these values when mastering for broadcast to ensure compliance with loudness standards (such as the CALM Act).



The meters display real-time values for each of the following measurements:

- The **M** meter represents the momentary loudness — in loudness units (LU) — across all audio channels based on 400-millisecond integration windows. The **Momentary** box displays a numeric representation of the momentary loudness.
- The **S** meter represents the short-term loudness — in loudness units — across all audio channels based on 3-second integration windows. The **Short** box displays a numeric representation of the short-term loudness.
- The **I** meter represents the integrated loudness — in loudness units — across all audio channels over the duration of the program. The **Integrated** box displays a numeric representation of the integrated loudness and includes an over-target indicator.
- The **LRA** meter represents the loudness range — in loudness units — of the momentary and short-term levels. The Loudness Range measurement provides a standardized method of determining the dynamic range of the signal.
- The **True peaks** meter represents the peak levels in dB FS. True peaks are calculated using a higher sample rate than peaks in the Channel Meters window for increased accuracy.

  The **True Peaks** indicator shows you whether the target loudness has been exceeded. The indicator is reset when you restart playback, or you can right-click the Loudness Meters window and choose **Reset Clip** from the shortcut menu.

Tips:

- *Loudness is recalculated whenever you start, stop, seek, or change playback direction. If you want to force a recalculation, right-click the window and choose* **Reset Metering Engine** *from the shortcut menu.*
- *Select the* **Enable surround processing for files with 6 channels** *check box on the Status tab of the Preferences dialog if you want to treat audio with six or more channels as surround audio when measuring loudness (a gain of ~1.5 dB is applied to the left and right surround channels). When the check box is cleared, all channels contribute equally to the loudness measurement.*

Choosing a metering mode

To change the mode of the meters, choose **Loudness Meters** from the Options menu, and then choose **EBU R 128 Mode** or **ATSC A 85 Mode** from the submenu (you can also right-click the meter to set its options).

- When using **EBU R 128 Mode**, the target value of the **Integrated** meter is -23 LUFS, and the maximum **True peak** value is -1.0 dB FS. Use this mode when you're mastering to European Broadcasting Union (EBU) standards.
- When using **ATSC A 85 Mode**, the target value of the **Integrated** meter is -24 LUFS, and the maximum **True peak** value is -2.0 dB FS. Use this mode when you're mastering to North American Advanced Television Systems Committee (ATSC) standards.

The over-target indicators will be triggered if the target values for **Integrated** and **True peak** meters are exceeded.

Choosing a loudness scale

To change the scale of the meter, choose **Loudness Meters** from the Options menu, choose **Loudness Scale**, and then choose **EBU +9** or **EBU +18** from the submenu (you can also right-click the meter to set its options).

- When using **EBU +9**, the meters are displayed with a range of -18 to +9 LU.
- When using **EBU +18**, the meters are displayed with a range of -36 to +18 LU.

> **Note:** *Choosing a wide range allows you to see low-level signals at the expense of precision display at high levels.*

Select **Absolute (-23 LUFS)** if you want to display loudness values as Loudness Units Full Scale (LUFS). When **Absolute (-23 LUFS)** is not selected, all values are expressed as Loudness Units (LU) relative to the selected mode (**EBU R 128 Mode** or **ATSC A 85 Mode**).

Configuring peak meters

To toggle the True Peaks meters in the Loudness Meters window, choose **Loudness Meters** from the Options menu, and then choose **Show True Peak Meter** from the submenu (you can also right-click the meter to set its options).

> **Tip:** *Please note that true peaks are calculated using a higher sample rate than peaks in the Channel Meters for increased accuracy.*

Peak levels may be miscalculated if audio signals are asymmetrical or if a DC offset is present. To enable filtering, choose **Loudness Meters** from the Options menu, and then choose **True Peak Blocking Filter** from the submenu (you can also right-click the meter to set its options). When **True Peak Blocking Filter** is selected, peaks are calculated as the maximum of the filtered and unfiltered signals.

The True Peaks meters display levels in dB FS. To change the resolution of the meters, do either of the following:

- From the Options menu, choose **Loudness Meters**, choose **True Peak Range** from the submenu, and then choose a display range.
- Right-click the Loudness meter, choose **True Peak Range** from the submenu, and then choose a display range.

> **Note:** *Choosing a wide range allows you to see low-level signals at the expense of precision display at high levels.*

Generating a loudness log

A loudness log is a report of the loudness of an audio file and it allows you to provide documentation that your files adhere to loudness standards.

Generating a loudness log when saving a file

1. Use the Save As dialog to save your file.
2. Select the **Generate Loudness Log** check box if you want Sound Forge to analyze the loudness of your file and create a log file that summarizes its loudness values.

   The loudness log is created using the same folder and base name as your sound file with **_loud.txt** appended to the name.

   The log will record the file name, format, loudness metering mode, and loudness values throughout the file.

> **Important:** *Loudness logging is performed after the plug-in chain, but before any codec is applied to your rendered file. Because audio compression may affect audio levels, choose* **Tools** > **Generate Loudness Log** *to create a log after saving to a compressed format.*
>
> *Select the* **Enable surround processing for files with 6 channels** *check box on the Status tab of the Preferences dialog if you want to treat audio with six or more channels as surround audio when measuring loudness (a gain of ~1.5 dB is applied to the left and right surround channels). When the check box is cleared, all channels contribute equally to the loudness measurement.*

Generating a loudness log for a data window

1. Select the data window you want to analyze.

> **Note:** *If no data is selected, the entire file is analyzed.*

2. Choose **Tools** > **Generate Loudness Log**.

   The loudness log is created using the same folder and base name as your sound file with **_loud.txt** appended to the name.

   The log will record the file name, format, loudness metering mode, and loudness values throughout the selection or file.

   If the **Open editor when new loudness log is generated** check box is selected on the Status tab in the Preferences dialog, the log will be automatically opened in your default text editor.

> **Important:** *Select the* **Enable surround processing for files with 6 channels** *check box on the Status tab of the Preferences dialog if you want to treat audio with six or more channels as surround audio when measuring loudness (a gain of ~1.5 dB is applied to the left and right surround channels). When the check box is cleared, all channels contribute equally to the loudness measurement.*

Chapter 10

Recording

Sound Forge Pro can record audio into multiple audio channels while simultaneously playing back existing audio tracks. You are limited only by the performance of your computer system and audio hardware.

> **Tip:** *You can use the* **Stereo Recording** *window layout to optimize the Sound Forge Pro interface for recording. For more information, see* *Loading default window layouts on page 327*.

Creating a new recording

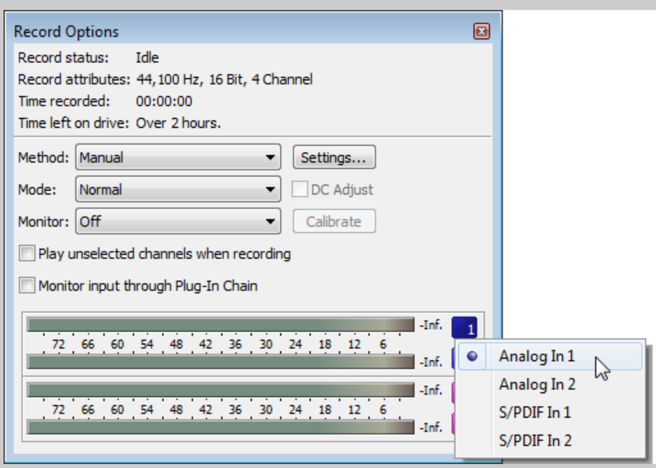
After you've connected an audio source and verified your recording setup, you're ready to start recording audio.

When you click the **Arm** ( ) or **Record** ( ) button when no data windows are open, Sound Forge Pro creates a new window automatically using the last-used new window settings. When **Create new windows** is selected in the **Mode** drop-down list in the Record Options window, a new window is each time you start recording.

If you want to recording into an existing sound file, see *Recording into an existing sound file on page 146*.

> **Tip:** *You can use the* **Stereo Recording** *window layout to optimize the Sound Forge Pro interface for recording. For more information, see* *Loading default window layouts on page 327*.

> **Note:** *The maximum number of channels recorded depends on the data window where you're recording. For example, if you enabled six inputs on the Record tab in Audio Preferences, you need to record into a six-channel data window to record all six inputs. If you record to a stereo data window, only two inputs will be recorded.*
>
> *To choose your recording input, use the Audio tab in the Preferences dialog or click a channel number in the Record Options window and choose a new input port from the menu.*



Creating a recording

> **Tip:** *When using the New Recording command, recording begins immediately after the new window is created. If you want to check your input levels before recording, click the* **Arm** *button ( ) in the main toolbar, check your levels using the meters in the Record Options window, and then click* **Arm** *again.*
>
> *The peak meters represent the volume of the recording input. For best results, the peak level should be somewhere in the yellow range with an occasional red segment: you want your input to be as loud as possible without clipping.*

1. Verify that your recording method is set to **Manual**.
   a. From the View menu, choose **Record Options**. The Record Options window is displayed.
   b. From the **Method** drop-down list, choose **Manual**.
   c. If you want to set up pre-roll, post-roll, or a prerecord buffer, click the **Settings** button. *For more information, see* *Setting up pre- and post-roll on page 144 or Setting a prerecord buffer on page 145*.
2. Choose **Transport** > **New Recording** (or press Ctrl+Shift+R).

> **Tip:** *If you want to check your input levels before or during recording, you can use the meters.*
>
> *The peak meters represent the volume of the recording input. For best results, the peak level should be somewhere in the yellow range with an occasional red segment: you want your input to be as loud as possible without clipping.*

3. Use the New Window dialog to specify the parameters for the new file:
   a. Choose a sample rate from the **Sample rate** drop-down list.
   b. Choose a setting from the **Bit depth** drop-down list to specify the number of bits that should be used to store each sample.
   c. Choose a setting from the **Channels** drop-down list to specify the number of channels that will be used in the window.

> **Tip:** *If you want to bypass the New Window dialog and use the last-used window settings, hold Shift while choosing* **Transport** > **New Recording***.*

4. Click **OK**. A new, untitled sound file is created, and recording begins immediately. During recording, the Time Display window and the data window's selection status bar will show the current record position.

> **Note:** *During recording, playback commands, the Preferences dialog, and commands that affect the recording data window are unavailable.*

5. Click the **Record** ( ) or **Stop** ( ) button to end recording, or click the **Pause** button ( ) to suspend recording and leave the recording device armed.

Setting up pre- and post-roll

Using pre- and post-roll can help you when recording voiceovers or overdubs:

- When you're performing punch-and-roll recording from the cursor, pre-roll allows you to hear the material before the cursor position.
- When you're recording into a selection, pre- and post-roll allow you to hear the material before and after the selection.

1. From the View menu, choose **Record Options**. The Record Options window is displayed.
2. From the **Method** drop-down list, choose **Manual**.
3. Click the **Settings** button in the Record Options window.
4. Select the **Pre-roll** check box and type a value in the edit box to set the amount of time before the selection that you want to play when recording.

5. Select the **Post-roll** check box and type a value in the edit box to set the amount of time after the selection that you want to play when recording.
6. Click the **OK** button.

   When performing punch-in recording, recording occurs underneath the pre- and post-roll. If your subject starts early, for example, you can adjust the event to uncover the recording. You can use the Event Tool ( ) to slip or trim the edges of the recorded event to expose the recorded pre- and post-roll. *For more information, see* *Editing events on page 173*.

   During recording, the **Record status** value in the Record Options window indicates that recording is armed, in pre-roll, recording, or in post-roll. The meters in the Record Options dialog monitor the level from your recording input.

Setting a prerecord buffer

A prerecording buffer helps to ensure you won't miss a perfect take when you're recording. When the prerecording buffer is enabled, sound data is written continuously to the buffer after you click the **Arm** button ( ). When you start recording, the sound data in the buffer is committed to disk.

After you finish recording, the buffer is not displayed in the window, but you can use the Event Tool ( ) to slip or trim the left edge of the recorded event to expose the buffer. *For more information, see* *Editing events on page 173*.

1. From the View menu, choose **Record Options**. The Record Options window is displayed.
2. From the **Method** drop-down list, choose **Manual**.
3. Click the **Settings** button in the Record Options window.
4. Select the **Prerecord buffer** check box and type a value in the edit box to set the duration of the buffer.
5. Click the **OK** button.

Reviewing recorded takes

Click the **Play** button ( ) to review your recording. Click the **Stop** button ( ) to end playback.

Recording into an existing sound file

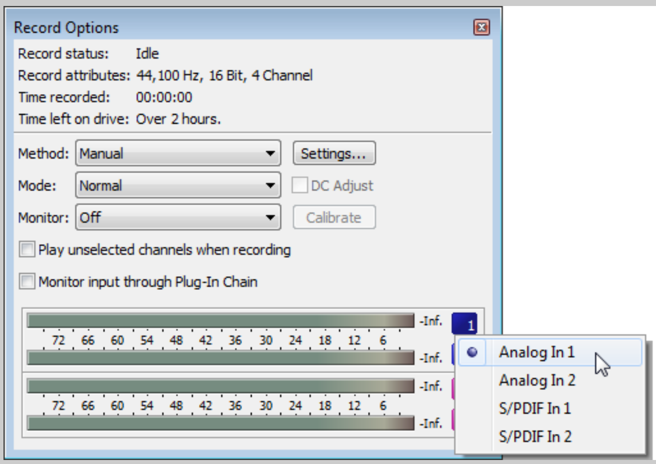
After you've connected an audio source and verified your recording setup, you're ready to start recording audio.

Click the **Record** button ( ) (or choose **Transport** > **Record**) to record into an existing sound file (also called punch-in recording).

If you want to record to a new file, see *Creating a new recording on page 143*.

> **Tip:** *You can use the* **Stereo Recording** *window layout to optimize the Sound Forge Pro interface for recording. For more information, see Loading default window layouts on page 327.*

> **Note:** *The maximum number of channels recorded depends on the data window where you're recording. For example, if you enabled six inputs on the Record tab in Audio Preferences, you need to record into a six-channel data window to record all six inputs. If you record to a stereo data window, only two inputs will be recorded.*
>
> *To choose your recording input, use the Audio tab in the Preferences dialog or click a channel number in the Record Options window and choose a new input port from the menu.*



Recording at the current cursor position or into a selection

1. Verify that your recording method is set to **Manual**.
   a. From the View menu, choose **Record Options**. The Record Options window is displayed.
   b. From the **Method** drop-down list, choose **Manual**.
   c. If you want to set up pre-roll, post-roll, or a prerecord buffer, click the **Settings** button. *For more information, see Setting up pre- and post-roll on page 144 or Setting a prerecord buffer on page 145.*
   d. From the **Mode** drop-down list, ensure **Normal** or **Create regions** is selected.
2. Select the sound data you want to replace, or click to position the cursor where you want to begin recording.

> **Tip:** *If you want to check your input levels before or during recording, you can use the meters.*
>
> *The peak meters represent the volume of the recording input. For best results, the peak level should be somewhere in the yellow range with an occasional red segment: you want your input to be as loud as possible without clipping.*

3. Click the **Arm** button ( ) if you want to begin recording as soon as possible after clicking the **Record** button ( ).

   The **Arm** button is optional, but can allow for more accurate takes. When you click **Arm**, the wave device is opened and all recording buffers are loaded in order to minimize the amount of time between clicking the **Record** button and when recording starts.

4. Click the **Record** button ( ) (or press Ctrl+R). Recording begins, and the Time Display window and the data window's selection status bar will show the current record position.

Note: *During recording, playback commands, the Preferences dialog, and commands that affect the recording data window are unavailable.*

5. Recording will stop automatically at the end of the selection.

   If you're recording without a selection, existing data is overwritten during recording, and you can click the **Record** ( ) or **Stop** ( ) button to end recording.

   Clicking **Pause** ( ) suspends recording, clearing the selection and moving the cursor to the end of the recorded data. When you pause recording, the recording device remains armed.

Recording multiple takes into a selection

1. Verify that your recording method is set to **Manual**.
   a. From the View menu, choose **Record Options**. The Record Options window is displayed.
   b. From the **Method** drop-down list, choose **Manual**.
   c. If you want to set up pre-roll, post-roll, or a prerecord buffer, click the **Settings** button. *For more information, see Setting up pre- and post-roll on page 144 or Setting a prerecord buffer on page 145*.
   d. From the **Mode** drop-down list, ensure **Normal** or **Create regions** is selected.
2. Select the sound data you want to replace, or click to position the cursor where you want to begin recording.

Tip: *If you want to check your input levels before or during recording, you can use the meters.*

The peak meters represent the volume of the recording input. For best results, the peak level should be somewhere in the yellow range with an occasional red segment: you want your input to be as loud as possible without clipping.

3. Select the **Loop Playback** button ( ).
4. Click the **Arm** button ( ) if you want to begin recording as soon as possible after clicking the **Record** button ( ).

   The **Arm** button is optional, but can allow for more accurate takes. When you click **Arm**, the wave device is opened and all recording buffers are loaded in order to minimize the amount of time between clicking the Record button and when recording starts.
5. Click the **Record** button ( ) (or press Ctrl+R). Recording begins, and the Time Display window and the data window's selection status bar will show the current record position.

Note: *During recording, playback commands, the Preferences dialog, and commands that affect the recording data window are unavailable.*

6. When recording reaches the end of the time selection, the cursor returns to the beginning of the selection, and a new take is recorded. Each take is added to the Undo/Redo History window.
7. Click the **Record** ( ) or **Stop** ( ) button to end recording.

   Clicking **Pause** ( ) suspends recording, clearing the selection and moving the cursor to the end of the recorded data. When you pause recording, the recording device remains armed.
8. You can click the **Play** button ( ) in the Undo/Redo History window to preview individual takes, or you can use the **Undo** and **Redo** commands to cycle through your recorded takes while previewing in the data window.

Reviewing recorded takes

Click the **Play** button ( ) to review your recording. Click the **Stop** button ( ) to end playback.

If you've recorded multiple takes, you can click the **Play** button ( ) in the Undo/Redo History window to preview individual takes, or you can use the **Undo** and **Redo** commands to cycle through your recorded takes while previewing in the data window.

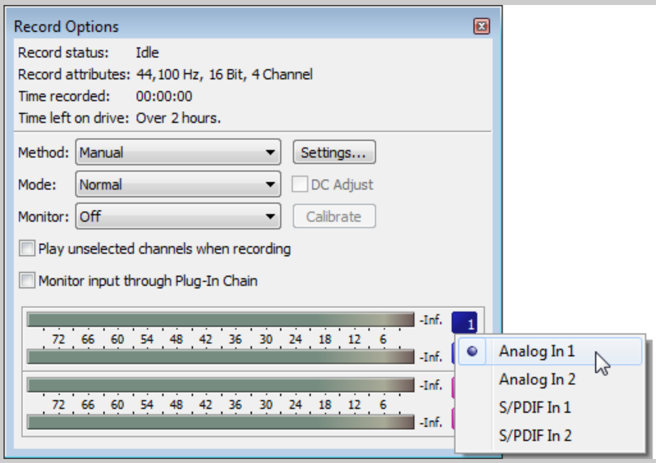
Recording audio automatically

You can set up recording to begin automatically from the selected input device using a timer, by detecting when audio exceeds a set threshold, or when MIDI timecode is detected.

When you're using threshold-triggered recording, you can choose to record continuously: set a buffer size, and the recorded audio will fill the buffer, discarding the oldest data as new data is recorded. If you want to save data from the buffer, you can save it to disk.

Note: *The maximum number of channels recorded depends on the data window where you're recording. For example, if you enabled six inputs on the Record tab in Audio Preferences, you need to record into a six-channel data window to record all six inputs. If you record to a stereo data window, only two inputs will be recorded.*

To choose your recording input, use the Audio tab in the Preferences dialog or click a channel number in the Record Options window and choose a new input port from the menu.



Recording audio over a set threshold

1. From the View menu, choose **Record Options**. The Record Options window is displayed.
2. Use the Record Options window to set the audio levels at which recording will start and stop:
   a. From the **Method** drop-down list, choose **Automatic: Threshold**.
   b. Choose a setting from the **Mode** drop-down list to choose whether to create regions or record to a new window when recording is suspended and resumed. *For more information, see Recording options on page 150*.
   c. Click the **Settings** button. The Threshold Settings tab in the Record Settings dialog is displayed.
   d. Drag the **Threshold** fader to set the audio level at which recording will begin.
   e. Drag the **Release** slider to set the amount of time the audio level should be below the **Threshold** setting before recording will stop.
   f. Select the **Automatically rearm after record** check box if you want to continue monitoring audio levels and recording until you click the **Stop** button ( ).
   g. Click **OK** to close the Record Settings dialog.

3. Click the **Arm** button ( ) in the data window where you want to record. The **Record status** value in the Record Options window indicates that recording is armed, and the meters in the Record Options window monitor the level from your recording input.

   Recording will begin when the audio signal meets the threshold level and will stop after the level falls below the threshold for the specified release time. Recording begins at the cursor position, and the Time Display window and the data window's selection status bar will show the current record position.

   If the destination window contains a selection that is shorter than the timer duration, recording will stop at the end of the selection. If the destination window contains a selection that is longer than the timer duration, recording will stop at the end of the timer duration.

> **Note:** *During recording, playback commands, the Preferences dialog, and commands that affect the recording data window are unavailable.*

Recording using MIDI timecode

1. From the Options menu, choose **MIDI In/Out**, and then choose **Trigger from MIDI Timecode** from the submenu.
2. From the View menu, choose **Record Options**. The Record Options window is displayed.
3. Use the Record Options window to set the MIDI timecode interval you want to record:
   a. From the **Method** drop-down list, choose **Automatic: MIDI Timecode**.
   b. Choose a setting from the **Mode** drop-down list to choose whether to create regions or record to a new window when recording is suspended and resumed. *For more information, see* *Recording options on page 150*.
   c. Click the **Settings** button. The MIDI Timecode Settings tab in the Record Settings dialog is displayed.
   d. From the **Input** drop-down list, choose the trigger device. Changing the setting here will also update the Input setting on the MIDI/Sync tab in the Preferences dialog.
   e. Select the **Timecode start** check box and type a value in the edit box to indicate the timecode location when recording will begin.
   f. Select the **Timecode stop** check box and type a value in the edit box to indicate the timecode location when recording will end. If you don’t indicate a stop time, recording will continue until you click the **Stop** button ( ).
   g. Select the **Bound record length on timecode loss** check box if you want to prevent recording beyond the specified end time. This ensures that your record length is exact regardless of any inaccurate timecode.
   h. Click **OK** to close the Record Settings dialog.
4. Click the **Arm** button ( ) in the data window where you want to record. The **Record status** value in the Record Options window indicates the timecode when recording will begin, and the meters in the Record Options window monitor the level from your recording input.

   Recording will begin when Sound Forge Pro detects the specified **Timecode start** value and will stop at the specified **Timecode stop** value. Recording begins at the cursor position, and the Time Display window and the data window’s selection status bar will show the current record position.

   If the destination window contains a selection that is shorter than the specified timecode range, recording will stop at the end of the selection. If the destination window contains a selection that is longer than the timecode duration, recording will stop at the **Timecode stop** value.

> **Note:** *During recording, playback commands, the Preferences dialog, and commands that affect the recording data window are unavailable.*

Recording using a timer

1. From the View menu, choose **Record Options**. The Record Options window is displayed.
2. Use the Record Options window to specify when you want to record:
    a. From the **Method** drop-down list, choose **Automatic: Time**.
    b. Choose a setting from the **Mode** drop-down list to choose whether to create regions or record to a new window when recording is suspended and resumed. *For more information, see* *Recording options on page 150*.
    c. Click the **Settings** button. The Time Settings tab in the Record Settings dialog is displayed.
    d. Click the **Add** button ( ) to create a timer setting (or click the **Edit** button ( ) to edit an existing setting). The Record Timer Event dialog is displayed.

Tips:

- *If you want to remove a timer setting, select it and click the* **Delete** *button ( ).*
- *If you want to remove all past timer settings, click the R***emove All Past Events from List** *button ( ).*

    e. Type a name in the **Name** box to create a name to identify the preset.
    f. Choose a setting from the **Recurrence** drop-down list to indicate whether you want to record one time only or repeat the timed recording at a regular interval.
    g. Use the **Start date**, **Start time**, and **Duration** boxes to indicate when you want to start and stop recording.
    h. Click **OK** to close the Record Timer Event dialog.
    i. Click **OK** to close the Record Settings dialog.
3. Click the **Arm** button ( ) in the data window where you want to record. The **Record status** value in the Record Options window will display a countdown to show you when recording will begin.

    When the timer is activated, recording begins at the cursor position, and the Time Display window and the data window's selection status bar will show the current record position.

    If the destination window contains a selection that is shorter than the timer duration, recording will stop at the end of the selection. If the destination window contains a selection that is longer than the timer duration, recording will stop at the end of the timer duration.

Note: *During recording, playback commands, the Preferences dialog, and commands that affect the recording data window are unavailable.*

Recording options

From the View menu, choose **Record Options** to open the Record Options window. You can use this window to configure various options for recording in Sound Forge Pro.

Tip: *The top of the Record Options window displays the current record status, attributes, time recorded, and time left on your hard drive. You can also display the record status in the Time Display window by right-clicking the Time Display window and choosing* **Record Status** *from the shortcut menu. For more information, see* *Customizing the Time Display window on page 329*.

Choosing a recording method

Choose a setting from the **Method** drop-down list to choose what happens when you start recording:

- **Manual**: Recording starts at the cursor position or selection. Use this mode for general-purpose recording, punch-in recording, or voiceover work.

    For more information, see Creating a new recording on page 143 or Recording into an existing sound file on page 146.

- **Automatic: Threshold**: Recording starts when the audio reaches a specified level and stops when the audio falls below that level for a specified duration.

    For more information, see Recording audio over a set threshold on page 148.

- **Automatic: MIDI Timecode**: Recording starts when the specified timecode is received from the MIDI input device and ends at the specified timecode.

  For more information, see Recording using MIDI timecode on page 149.

- **Automatic: Time**: Recording starts at the date and time you specify and stops after the specified duration.

  For more information, see Recording using a timer on page 150.

Choosing a recording mode

Choose a setting from the **Mode** drop-down list to choose what happens after you stop (or pause) and restart recording:

- **Normal**: Click the **Record** ( ) or **Stop** button ( ) to end recording, or click the **Pause** button ( ) to suspend recording and leave the recording device armed.
- **Create regions**: A new region is created each time you restart or resume recording. If you're recording into a time selection with **Loop Playback** enabled, Sound Forge Pro does not create a new region for each loop.
- **Create new windows**: A new window is created each time you restart or resume recording.

Notes:

- *When **Create new windows** is selected, the **Arm** ( ) and **Record** ( ) buttons on the main toolbar are enabled even when no data windows are open. When **Normal** or **Create regions** is selected, the **Arm** ( ) and **Record** ( ) buttons are not available until you create a data window or open a file.*
- *When **Create new windows** is selected, punch-in recording is not available. For more information, see Recording into an existing sound file on page 146.*

Setting up input monitoring

Turning input monitoring on or off

Choose **On**, **Off** , or **Auto** from the **Monitor** drop-down list to toggle record input monitoring.

When **Auto** is selected, the input is monitored during pre/post-roll and during recording only.

Monitoring your recording input through the Plug-In Chain

Select the **Monitor input through Plug-In Chain** check box if you want to monitor the selected channels from your recording input through the Plug-In Chain.

Notes:

- ***On** or **Auto** must be selected in the **Monitor** drop-down list to enable input monitoring.*
- *This setting is used for monitoring only; the Plug-In Chain is not applied to the recorded data.*

Monitoring unselected channels when recording

Select the **Play unselected channels when recording** check box to if you want to monitor additional channels while recording.

- When the check box is selected, all unselected audio channels will play when you're recording into the selected channels.

  This setting is useful when you're recording to backing tracks. For example, if you have a four-channel audio file, you could place backing tracks on channels 1 and 2 and record into channels 3 and 4.

  Select channels 3 and 4, and select the **Play unselected channels when recording** check box. When you start recording, your audio is recorded into channels 3 and 4, and you'll be able to monitor channels 1 and 2 while recording.

- When the check box is cleared, unselected audio channels will play only during pre/post roll.

Adjusting for DC offset

Sound Forge Pro software can automatically adjust for any DC offset produced by your audio hardware during the recording process.

1. Set up your hardware. *For more information, see* *Recording setup on page 152*.
2. From the View menu, choose **Record Options** to open the Record Options window.
3. Select the **DC adjust** check box.
4. Click the **Calibrate** button.

> **Note:** *If you change sound cards or are recording from different digital sources or at different sample rates, you should recalibrate the DC offset before recording.*

Recording setup

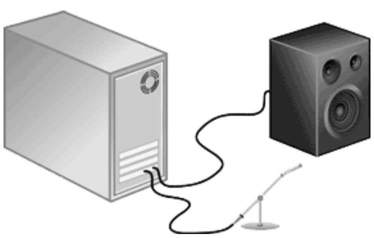
This section provides general guidelines to help you record sound from an external source using Sound Forge Pro software. Your specific hardware may vary. Please refer to your hardware documentation for more information.

> **Tip:** *If you're recording from a turntable, use a phono preamplifier between your turntable's output and your sound card's line input. Most turntables' outputs are phono-level (rather than line-level) outputs. Phono-level outputs are quieter than line-level outputs and have special equalization applied. A phono preamplifier will convert the phono-level signal to a line-level signal that you can record.*

Connecting an audio source to your sound card's input

Basic setup

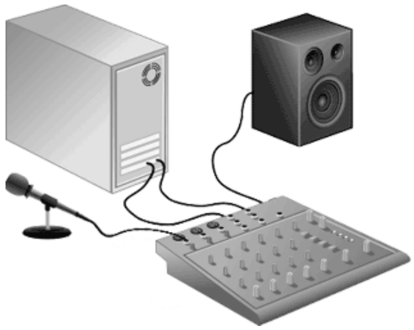
In this setup, an audio source is connected to an input on your sound card, and your powered speakers are connected to a **Line Out** output. You could connect a computer microphone to your sound card's **Mic In** input, or you can connect line-level outputs from a tape deck or other source to a **Line In** input.



Basic setup with mixer/preamplifier

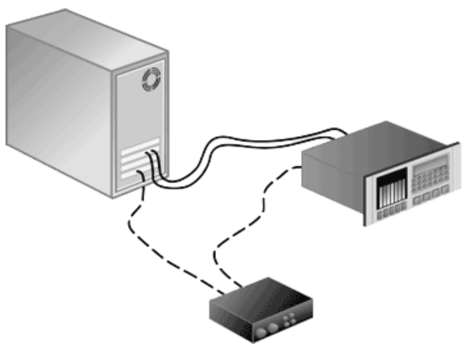
In this setup, your speakers and audio source are connected to a mixer or preamplifier. The mixer/preamplifier is then connected to **Line In** and **Line Out** connections on your sound card.

> **Tip:** *If you're recording from a turntable, use a phono preamplifier between your turntable's output and your sound card's line input. Most turntables' outputs are phono-level (rather than line-level) outputs. Phono-level outputs are quieter than line-level outputs and have special equalization applied. A phono preamplifier will convert the phono-level signal to a line-level signal that you can record.*



Digital input/output with MIDI synchronization

In this setup, an audio source with digital input/output is connected to a sound card with digital input and outputs. Dashed lines represent a sync connection from your audio source to a MIDI timecode converter to a MIDI card.

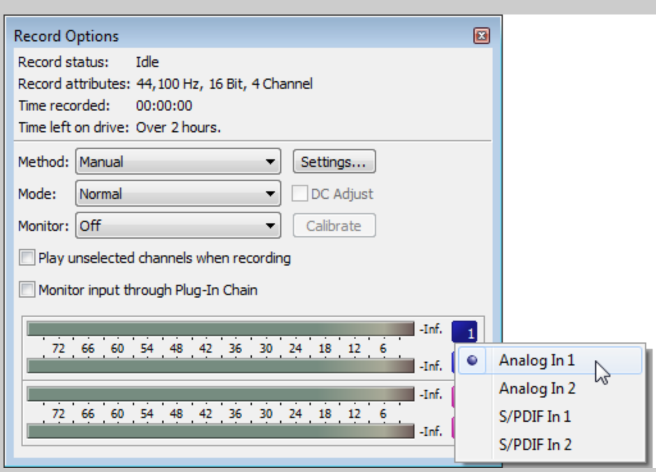


Choosing an input device and adjusting levels

The Record tab in the Audio Preferences page allows you to choose the audio inputs from which you want to record. *For more information, see Audio tab on page 342*. Before recording, you'll need to verify that your sound card's recording inputs are active.

Note: *The maximum number of channels recorded depends on the data window where you're recording. For example, if you enabled six inputs on the Record tab in Audio Preferences, you need to record into a six-channel data window to record all six inputs. If you record to a stereo data window, only two inputs will be recorded.*

To choose your recording input, use the Audio tab in the Preferences dialog or click a channel number in the Record Options window and choose a new input port from the menu.



1. Ensure all cables are connected and that your audio source is generating a signal.
2. Adjust your recording levels:
   - If your audio device provides a console application to adjust levels, open the application and adjust its gain controls while monitoring the peak meters on the Meters tab in the recording dialog. Adjust the gain controls in the console application so Sound Forge receives a strong signal with no clipping.

     For more information about using your sound card and its console application, please refer to the manufacturer's documentation.
   - If you're using your Windows sound card, perform the following steps to open the recording controls:
     a. Double-click the speaker icon ( ) in your system tray to open the Volume Control window.
     b. From the Options menu, choose **Properties**.
     c. Click the **Recording** radio button and click **OK**.
     d. Select (or unmute) the device from which you want to record.
     e. Adjust the **Volume** faders for the selected device and for the Master Record level while monitoring the recording meters in the Sound Forge Record dialog.

        For example, if you want to record from an audio CD in your CD-ROM drive, the CD **Mute** check box should not be selected, and the CD and Master Record **Volume** faders must be adjusted so Sound Forge receives a strong signal with no clipping.

Adjusting for DC offset

Sound Forge Pro software can automatically adjust for any DC offset produced by your audio hardware during the recording process. Perform the following steps before you start recording.

1. From the View menu, choose **Record Options** to open the Record Options window.
2. Select the **DC adjust** check box.
3. Click the **Calibrate** button.

Note: *If you change sound cards or are recording from different digital sources or at different sample rates, you should recalibrate the DC offset before recording.*

Recording multichannel audio

If you have an audio device that supports multiple inputs, you can use Sound Forge to perform multichannel recording.

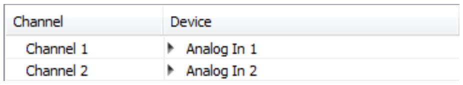
Tips:

- *Sound Forge is not a multitrack editor — check out our Vegas and ACID family of products for full multitrack recording and editing. You can use multichannel recording to create surround audio or capture field recordings.*
- *If you experience gapping or glitching when recording multichannel audio, try increasing your buffer size. You can increase the Record buffering setting on the Audio tab of the Preferences dialog or click the Advanced button on the Audio tab of the Preferences dialog to increase your device's buffers.* *For more information, see Audio tab on page 342.*

1. Connect your audio sources to your sound card's inputs. *For more information, see Recording setup on page 152.*
2. Enable your recording inputs:
   a. From the Options menu, choose **Preferences**, and click the Audio tab.
   b. Choose your recording device from the **Audio device type** drop-down list.
   c. Click the Record tab.

d. To assign a channel to an input, click the Device entry and choose an input from the drop-down list.

In the following example, the signal from Analog In 1 is recorded to channel 1 and Analog In 2 is recorded to channel 2.



e. Click **OK** to close the Preferences dialog and save your changes.

3. Verify that your recording method is set to **Manual**.

   a. From the View menu, choose **Record Options**. The Record Options window is displayed.

   b. From the **Method** drop-down list, choose **Manual**.

   c. If you want to set up pre-roll, post-roll, or a prerecord buffer, click the **Settings** button. F*or more information about other recording modes and methods, see* *Recording options on page 150*.

4. Create a data window for your recording.

   a. From the File menu, choose **New**.

   b. Choose a sample rate from the **Sample rate** drop-down list, or type a custom value in the edit box.

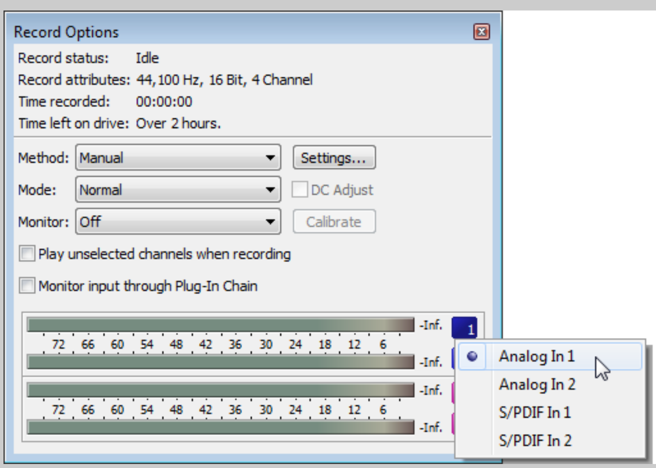
   c. Choose a setting from the **Bit depth** drop-down list to specify the number of bits that should be used to store each sample.

   d. Choose a setting from the **Channels** drop-down list to specify the number of channels that will be used in the window.

   e. Click **OK**. A new, untitled sound file is created.

Note: *The maximum number of channels recorded depends on the data window where you're recording. For example, if you enabled six inputs on the Record tab in Audio Preferences, you need to record into a six-channel data window to record all six inputs. If you record to a stereo data window, only two inputs will be recorded.*

To choose your recording input, use the Audio tab in the Preferences dialog or click a channel number in the Record Options window and choose a new input port from the menu.



5. Select the sound data you want to replace, or click to position the cursor where you want to begin recording.

Tip: *If you want to check your input levels before or during recording, you can use the meters.*

The peak meters represent the volume of the recording input. For best results, the peak level should be somewhere in the yellow range with an occasional red segment: you want your input to be as loud as possible without clipping.

6. Click the **Arm** button ( ) if you want to begin recording as soon as possible after clicking the **Record** button ( ).

   The **Arm** button is optional, but can allow for more accurate takes. When you click **Arm**, the wave device is opened and all recording buffers are loaded in order to minimize the amount of time between clicking the **Record** button and when recording starts.

7. Click the **Record** button ( ) (or press Ctrl+R). Recording begins, and the Time Display window and the data window's selection status bar will show the current record position.

> **Note:** *During recording, playback commands, the Preferences dialog, and commands that affect the recording data window are unavailable.*

8. Recording will stop automatically at the end of the selection.

   If you're recording without a selection, existing data is overwritten during recording, and you can click the **Record** ( ) or **Stop** ( ) button to end recording.

   Clicking **Pause** ( ) suspends recording, clearing the selection and moving the cursor to the end of the recorded data. When you pause recording, the recording device remains armed.

Generating MTC/SMPTE synchronization during recording

Sound Forge software can generate MTC/SMPTE synchronization while recording.

1. From the Options menu, choose **Preferences**, and click the MIDI/Sync tab.
2. On the MIDI/Sync tab, choose the trigger device from the **Input** drop-down list and click the **OK** button.
3. From the Options menu, choose **MIDI In/Out**, and then choose **Generate MIDI Timecode** from the submenu to enable MIDI timecode output.
4. Click the **Record** button ( ).
5. Click the Advanced tab at the bottom of the Record dialog.
6. Select the **Enable MTC/SMPTE Output Synchronization** check box.
7. Select the **Start** check box and specify the time you want to start recording.
8. Select the **Pre Roll** check box and type a value in the edit box to begin SMPTE output at a specified time before recording.
9. Click the **Close** button.

Chapter 11

Editing, Repairing, and Synthesizing Audio

This chapter introduces some of the Sound Forge® Pro advanced editing, repair, and synthesis features.

Overwriting and replicating

Earlier in this manual, paste and mix were described as ways of adding clipboard contents to the current data window. As your audio editing projects become more elaborate, you may discover the need for two more sophisticated paste operations: overwrite and replicate.

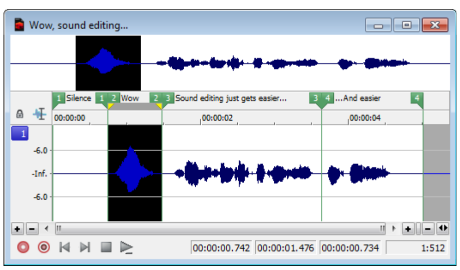
Overwriting

Overwriting allows you to replace the current selection with the contents of the clipboard and has two basic guidelines:

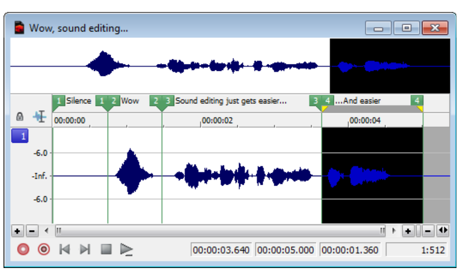
If	Then
The selection is longer than the clipboard contents	Data is overwritten from the beginning of the selection for the length of the clipboard contents only.
The selection is shorter than or equal to the clipboard contents	Data is overwritten for the length of the selection only.

Overwriting a selection

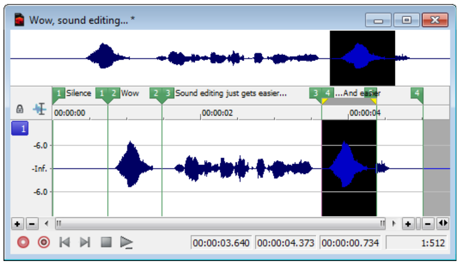
1. Open the Voiceover.pca file.
2. Create a selection containing “Wow.”



3. Copy the selection. The data is placed on the clipboard.
4. Create a selection of approximately the same length containing the final “...and easier.”



5. From the **Edit** menu, choose **Paste Special**, and choose **Overwrite** from the submenu or right-click the data window and choose **Overwrite** from the shortcut menu. The selection is overwritten with the clipboard contents.



Note: *If any of the selection data remains, it is because the length of the clipboard contents was less than the length of the selection.*

Replicating

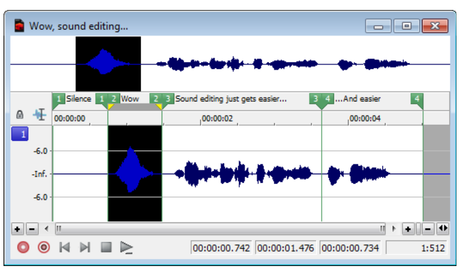
Replicating allows you to overwrite your current data window selection with several copies of the clipboard contents. When replicating, you must specify whether you want to use partial copies of the clipboard contents or only complete copies.

- Using partial copies of the clipboard content completely overwrites the selected data window area.
- Using complete copies of the clipboard content prevents a portion of the data window selection from being overwritten unless the selection length is an exact multiple of the length of the clipboard contents.

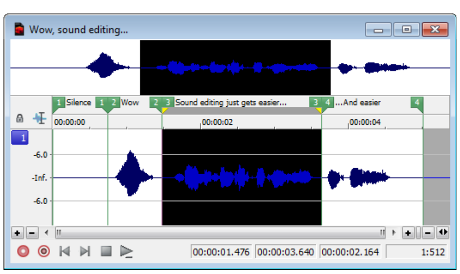
Note: *The* **Replicate** *command will paste as many copies of the clipboard as will fit in the current selection. If no selection exists in the data window, the command is not available.*

Replicating a selection

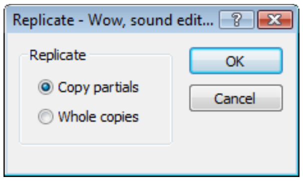
1. Open the Voiceover.pca file.
2. Create a selection containing “Wow.”



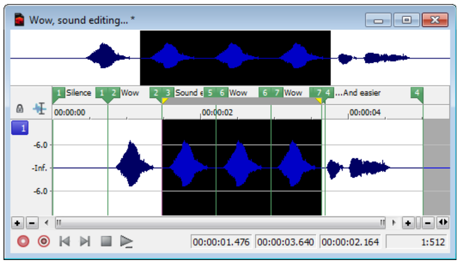
3. Copy the selection. The data is placed on the clipboard.
4. Create a selection containing “Sound editing just gets easier.”



5. From the **Edit** menu, choose **Paste Special** and choose **Replicate** from the submenu. The Replicate dialog is displayed.



6. Select the **Copy partials** radio button and click **OK**. The selection is overwritten with multiple copies of the clipboard contents. A partial copy of the clipboard contents is used where appropriate.



Repeating an operation

Once you perform an operation on an audio file, you can quickly repeat it with the same parameters by choosing **Repeat** from the **Edit** menu. This allows you to reapply the same effect, process, or function to a different section of audio using the same parameters.

> **Note:** *In the* **Edit** *menu, the* **Repeat** *command is displayed in conjunction with the name of the previous function.*

You can also repeat an operation by doing any of the following actions:

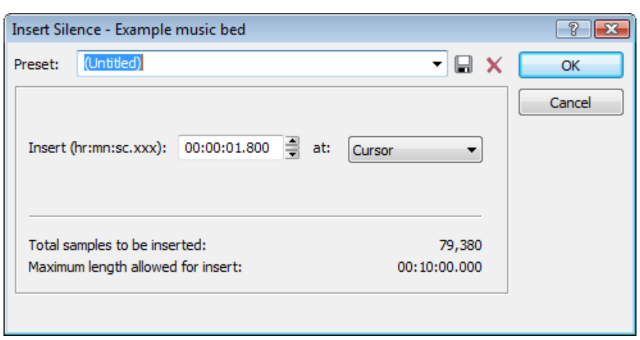
- Hold Shift while choosing the command from its menu.
- Press Ctrl+Y.
- Click the **Repeat** button ( ) on the Standard toolbar.

Inserting silence

The **Insert Silence** command allows you to place sections of silence in audio files.

1. Open the Musicbed.pca file.
2. From the **Insert** menu, choose **Silence**. The Insert Silence dialog is displayed.

> **Tip:** *You can also click the* **Insert Silence** *button ( ) on the Insert toolbar.*



3. Perform one of the following actions:
   - From the **Preset** drop-down list, choose a preset that has been stored for the plug-in.
   - Specify the length of silence that you want to add in the **Insert** box and choose a setting from the **at** drop-down list to specify where the silence should be inserted.

Setting	Description
Cursor	Inserts silence at the current cursor position.
Start of file	Inserts silence at the beginning of the file.
End of file	Inserts silence at the end of the file.

4. Click the **OK** button.

Using drag-and-drop operations

You can take advantage of using drag-and-drop operations to perform many common tasks. Drag-and-drop operations make controlling the Sound Forge software faster and more intuitive and allow for increased editing power. The major drag-and-drop editing operations are paste, mix, and create CD tracks.

Dragging mono selections into multichannel destinations

When pasting or mixing a mono selection into a multichannel file, you can mix the selection to both channels by dropping it on the destination data window's center line. Otherwise, the selection is mixed into the left or right channel exclusively.

Snapping to points in drag-and-drop operations

A major advantage of drag-and-drop editing is the ability to snap to markers, regions, time increments, or other points in the destination window. All drag-and-drop operations can be configured to snap (or align) to points established within the destination file.

The following table describes all points that drag-and-drop selections snap to in the destination file.

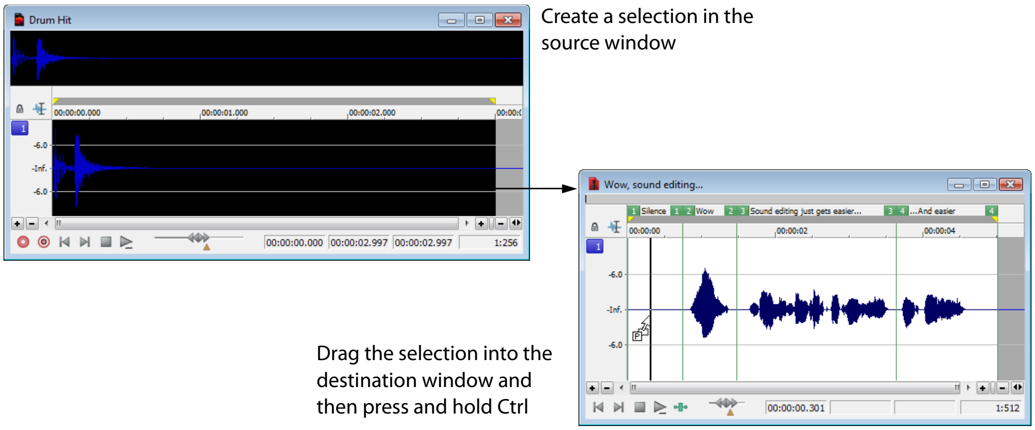
Points	Description
Cursor	Start of block snaps to cursor position.
Selection	Start of block snaps to start or end points of a selection.
Start	Start of block snaps to start of file.
End	Start of block snaps to end of file.
Markers	Start of block snaps to marker.
Regions Start and End Markers	Start of block snaps to region start or end.
Time, Measures, etc.	Start of block snaps to labeled divisions on time ruler.
Video Frames	Start of block snaps to the start of video frames appearing in the video strip.
Events	Start of block snaps to the start or end of an event.

Pasting, mixing, and creating CD tracks with drag-and-drop

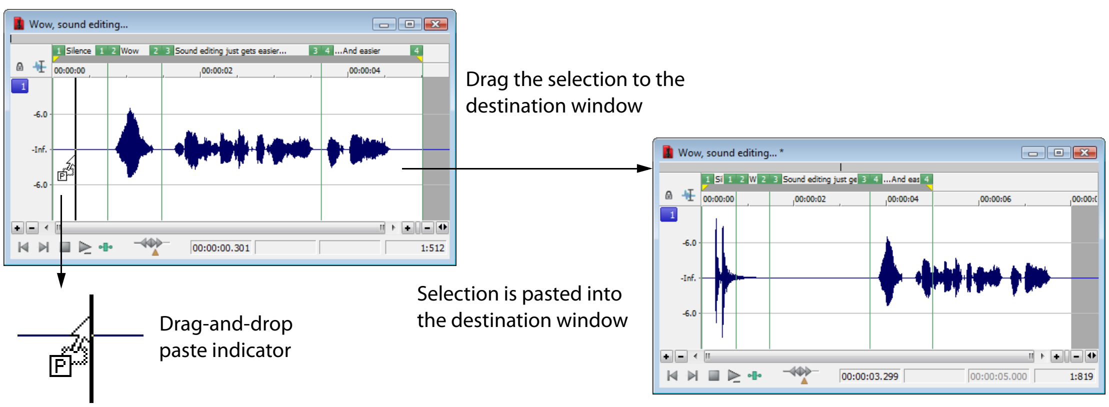
You can drag an audio selection and paste, mix, or create a CD track in another data window.

Pasting

1. Open the Voiceover.pca and Drumhit.pca files.
2. Select all audio data in Drumhit.pca.



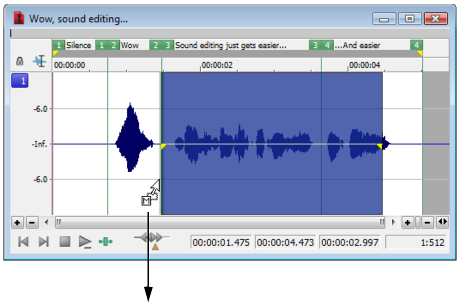
3. Hold the Ctrl key and drag the selection to the Voiceover data window.
   - A vertical line representing the leading edge of the source selection appears in the destination window.
   - The letter "P" appears in the box adjacent to the pointer.
4. Use the mouse to position the line in the destination window where the source data will be pasted.

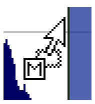


5. Release the mouse button. The selection is pasted into the destination window.

Mixing

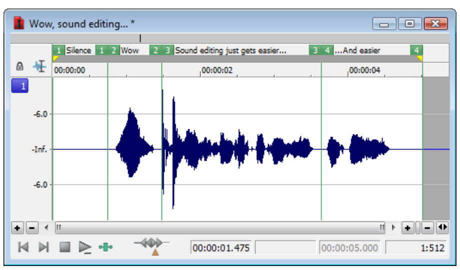
1. Open the Voiceover.pca and Drumhit.pca audio files.
2. Select all audio data in the Drumhit data window.
3. Drag the selection to the Voiceover data window.
    - A shaded region representing the source selection appears in the destination window.
    - An “M” appears in the box adjacent to the pointer.





Drag-and-drop mix indicator

4. Position the leading edge of the shaded region in the Voiceover data window where the mixing of the selection will begin.
5. Release the mouse button. The Mix/Replace dialog appears.
6. Verify that both **Volume** levels are set to 0 dB and click **OK**.



Selection is mixed into the destination window

Creating CD tracks

You can create CD tracks by dragging selections or files from the Explorer window to a data window. *For more information, see Adding files to a data window and creating tracks on page 312*.

Toggling the Mix, Paste, and CD Track functions

An alternate way of specifying a mix, paste, or CD track is the mouse toggle method.

1. Open the Voiceover.pca and Drumhit.pca files.
2. Select all audio data in the Drumhit data window.
3. Drag the selection to the Voiceover data window. A shaded region representing the source selection appears in the destination window and a letter or CD icon appears in the box adjacent to the pointer.
4. Continue holding the left mouse button while clicking the right mouse button. The mouse icon and the appearance of the selection region change to indicate the current drag-and-drop mode.
5. Release the left mouse button. The source audio data is pasted, mixed, or inserted as a CD track.

Creating new windows by dragging and dropping a selection

Drag-and-drop also allows you to create a new data window from a selection.

1. Open the Voiceover.pca file.
2. Create a selection containing “Wow.”
3. Drag the selection to an empty area of the Sound Forge workspace and drop it. A new data window is created containing the selection data with the attributes of the original file.

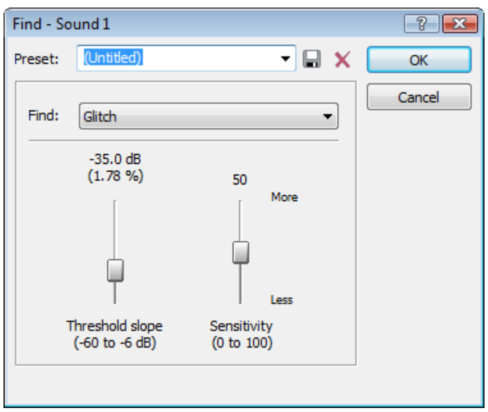
Finding and repairing audio glitches

Glitches are commonly the result of analog audio editing, analog to digital transfer, or electronic noise. Sound Forge software provides you with a tool for locating audio glitches and three distinct tools for repairing them: channel, interpolate, and replace. In addition, you can repair audio glitches manually using the Pencil tool.

Locating glitches

The Find tool allows you to quickly locate glitches, specific volume levels, or silence in a file. The Find tool’s glitch algorithm locates glitches by examining the file for instances where the waveform matches the specified threshold slope and sensitivity criteria. The cursor then moves to the location of the glitch to allow you to repair it. This tool locates only one glitch at a time. Therefore, it may be necessary to execute this command several times on a file to locate all glitches.

1. Open any audio file containing glitches.
2. From the **Tools** menu, choose **Find**. The Find dialog is displayed.



3. From the **Find** drop-down list, choose **Glitch**.
4. Adjust the **Threshold slope** fader to configure the minimum slope that constitutes a glitch.
   - A high value detects only glitches with steep slopes.
   - A lower value detects glitches with both steep and more gradual slopes.
5. Adjust the **Sensitivity** fader to determine the sensitivity of the detection algorithm.
   - A high value results in any part of the waveform with a slope greater than the **Threshold slope** being detected as a glitch.
   - A lower value forces the algorithm to verify that the slope is indeed a glitch, and not simply a portion of the smooth waveform.
6. Click **OK**. The first glitch in the file is found and its location is marked with the cursor.

Tip: *If you can hear glitches that the* **Find** *tool does not locate, decrease the* **Threshold slope** *and increase the* **Sensitivity**.

Locating additional glitches using the same settings

Once you have configured the settings in the Find dialog, you can find the next glitch in the file without viewing the Find dialog. To find the next glitch using the current settings, hold Shift while choosing **Find** from the **Tools** menu or hold Shift while clicking the **Find** button ( ) on the Tools toolbar.

Using the Shift key in this way is not limited to finding glitches. You can hold Shift and choose any command from a menu to repeat the command with the same settings. *For more information, see Repeating an operation on page 159.*

Repairing audio

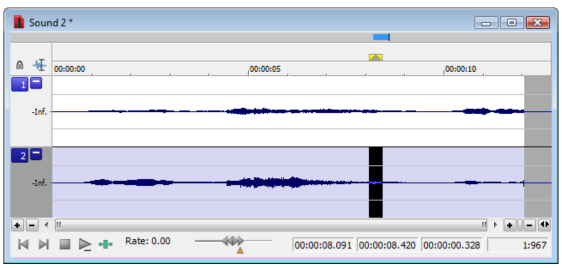
There are several ways to repair audio glitches.

Copying the other channel

For glitches in a single channel of a multichannel file, you can replace the glitched section of the damaged channel with the corresponding data from a "good" channel.

Note: *This method works only if the channels contain similar audio.*

1. Open the file containing the glitch.
2. Create a selection in the channel containing the glitch, three or four times longer (maximum 50 ms) than the glitch itself.



3. From the **Tools** menu, choose **Repair**, and choose **Copy Other Channel** from the submenu. The selected data is replaced with the corresponding data from the "good" channel. In addition, rapid crossfades are created at the beginning and end of the replacement selection to prevent a new glitch from being created.

Tip: *If this method fails to repair the glitch, undo it and apply* **Copy Other Channel** *again, this time using a longer selection.*

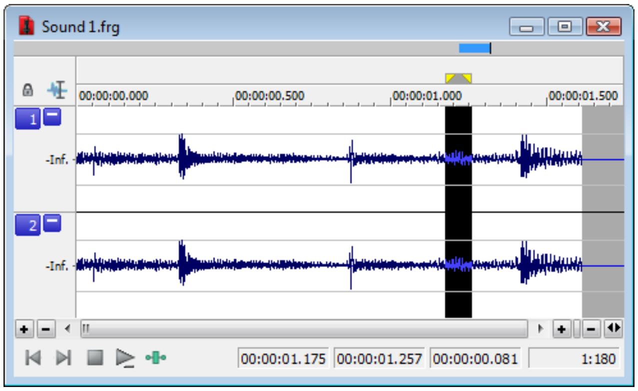
Interpolating new audio

This is the most basic method of repairing glitches. New audio data is simply interpolated based on the data at the beginning and end of the selection. This method results in a straight line connecting the beginning and end of the selection. Interpolation should be used to repair only small (less than 2 ms) glitches.

1. Open the file containing the glitch.
2. Right-click the data window and choose **Zoom** from the shortcut menu, and choose **In Full** from the submenu. The file is displayed at a 24:1 zoom ratio.
3. Create a selection containing the glitch.

Tip: *To improve the accuracy of this feature, the selection should be as small as possible while still containing the glitch.*

4. From the **Tools** menu, choose **Repair**, and choose **Interpolate** from the submenu. The glitch data is replaced with interpolated data.



Data is interpolated within the selection

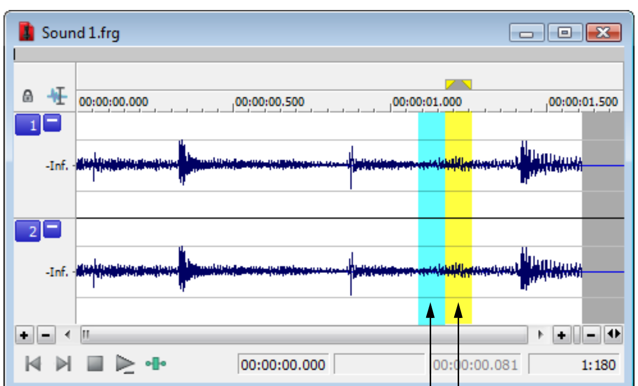
Replacing audio with preceding data

The Replace tool allows you to repair audio files by replacing the damaged data with the data immediately preceding it. This repair method is useful for repairing longer glitches such as needle drops and scratches.

1. Open the file containing the glitch.
2. Create a 5 to 50 ms selection containing the damaged audio.

> **Note:** *The maximum allowed replace time is 0.5 seconds.*

3. From the **Tools** menu, choose **Repair**, and choose **Replace** from the submenu. The selection is replaced with the selection of identical length immediately preceding the damaged data. In addition, rapid crossfades are created at the beginning and end of the replacement selection to prevent a new glitch from being created.



Replacement data
Selection data

Repairing audio glitches manually with the Pencil tool

The Pencil tool is for users who prefer to repair their audio glitches manually. This tool allows you to repair waveform glitches by redrawing the damaged waveform section. However, the Pencil tool can only be used when a file's waveform displays at a zoom ratio of 1:32 or lower.

1. Open the file containing the glitch.
2. Zoom in tightly on the glitch.
3. Select the Pencil tool using any of the following methods:
   - From the **Edit** menu, choose **Tool**, and choose **Pencil** from the submenu.
   - Click the **Pencil Tool** button ( ) in the Standard toolbar.
   - Click the Edit Tool Selector in the top-left corner of the data window until the **Pencil** tool is displayed.

4. Drag to draw a new waveform section. The new section is integrated into the original waveform, replacing the section containing the glitch.

Repairing audio using Audio Restoration plug-in

Sound Forge software includes an DirectX plug-in (part of the Noise Reduction plug-in) called Audio Restoration that you can use to remove surface noise from old recordings. For more information on this plug-in, see the Sound Forge online help file (accessible from the **Help** menu by choosing **Contents and Index**).

Synthesizing audio

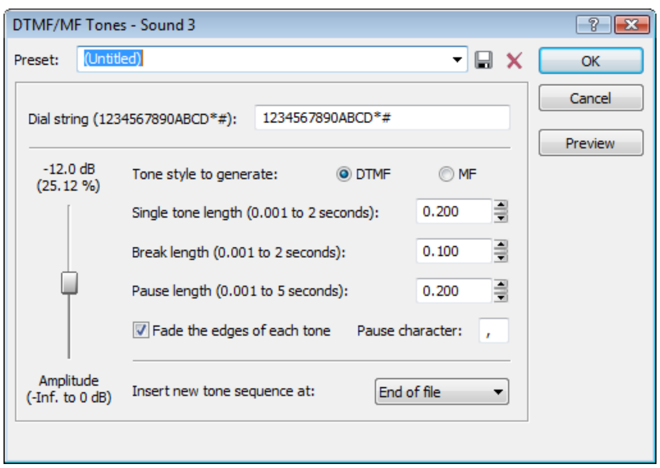
You can generate custom tones and waveforms for use in your audio projects.

Generating DTMF/MF tones

You can generate standard dial tones used by telephone companies.

1. From the **Insert** menu, choose **Synthesis**, and choose **DTMF/MF Tones** from the submenu. The DTMF/MF Tones dialog appears.

> **Tip:** *You can also click the* **DTMF/MF Tones Synthesis** *button ( ) on the Insert toolbar.*



2. Enter the phone number to be generated in the **Dial string** edit box, including pause characters.

> **Note:** *Unknown characters are ignored.*

3. Use the **Amplitude** fader to set the peak level of the waveform.
4. Select the **Tone style to generate** radio button corresponding to the tone to be generated.
   - DTMF (Dual Tone Multi-Frequency) signals are used by standard push-button telephones and are generated using combinations of 679, 770, 852, 941, 1209, 1336, 1477, and 1633 Hz sine waves.
   - MF signals are used internally by the telephone networks and are generated with a combination of 700, 900, 1100, 1300, 1500, and 1700 Hz sine waves.
5. Specify the output length (in seconds) of each tone in the **Single tone length** box.
6. Specify the length (in seconds) of silence between tones in the **Break length** box.
7. Specify the pause length (in seconds) to be inserted for a pause character in the **Pause length** box.
8. Select the **Fade the edges of each tone** check box to help prevent glitching.
9. Specify the pause character in the **Pause character** box.
10. Use the **Insert new tone sequence at** drop-down list to specify where the generated tone is placed in the audio file.

11. Click **OK**.

Generating audio with frequency modulation

The Sound Forge FM Synthesis feature can be used to create complex sounds from simple waveforms using frequency modulation (FM).

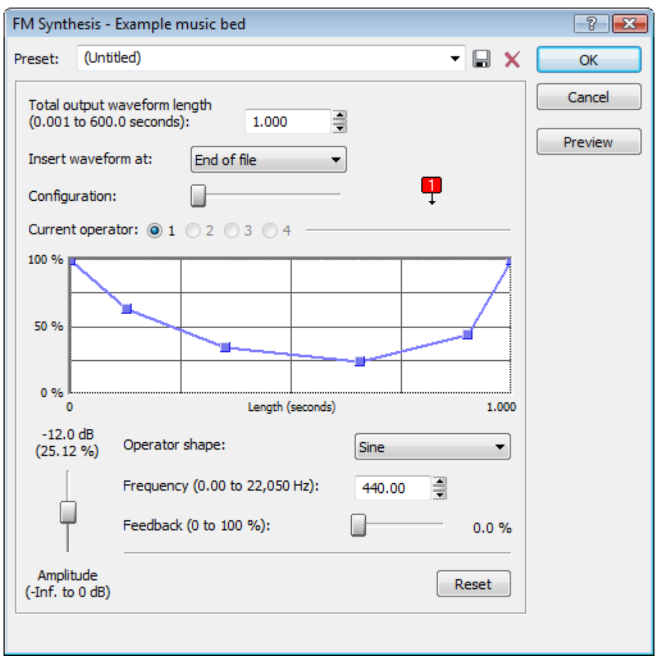
In frequency modulation, the frequency of a waveform (carrier) is modulated by the output of another waveform (modulator) to create a new waveform. If the frequency of the modulator is low, the carrier is detuned slowly over time. If the frequency of the modulator is high, the carrier is modulated so quickly that numerous additional frequencies (or sidebands) will be created.

Using the **FM Synthesis** tool, up to four waveforms (operators) can be used in a variety of configurations. Depending on the configuration, a waveform can be a carrier, a modulator, or a simple, unmodulated waveform.

Generating a waveform

1. From the **Insert** menu, choose **Synthesis**, and choose **FM** from the submenu. The FM Synthesis dialog appears.

Tip: *You can also click the* **FM Synthesis** *button ( ) on the Insert toolbar.*



2. Specify the length (in seconds) of the generated waveform in the **Total output waveform length** box.
3. Use the **Configuration** slider to configure the arrangement and number of operators used to generate the waveform. *For more information, see Specifying the number and arrangement of operators on page 168.*
4. Modify individual operators as needed. *For more information, see Modifying an operator on page 168.*
5. From the **Insert waveform at** drop-down list, choose a position to determine where the generated waveform is placed in the file.
6. Click **OK**.

Specifying the number and arrangement of operators

Dragging the **Configuration** slider changes the graphical representation of the arrangement and number of operators used to generate the waveform. When configuring your waveform, keep the following guidelines in mind:

- The outputs of horizontally joined operators are simply mixed. The outputs of the bottom operators are mixed to form the final output. Mixing unique simple waveforms is referred to as additive synthesis.
- Operators joined vertically are FM carrier-modulator pairs. The bottom operator is the carrier and the top operator is the modulator.
- Operators without other operators directly above are simple waveform generators.
- When three or more operators are stacked, the top operator modulates the operator below it, which modulates the following operator, and so on.

Modifying an operator

1. Select the **Current operator** radio button corresponding to the operator to be modified.
2. Use the envelope graph to modify the amplitude of the operator over time. *For more information, see Envelope graphs on page 49*.
3. From the **Operator shape** drop-down list, choose a waveform shape.
4. Specify the frequency of the operator in the **Frequency** box.

> **Notes:**
>
> - *If* **Frequency** *is set to 0.00, a DC (zero-frequency) waveform is produced regardless of the waveform specified.*
> - *When you choose* **Filtered Noise** *from the* **Operator shape** *drop-down list,* **Frequency** *determines the high-frequency content of the noise.*

5. Use the **Feedback** slider to determine the amount of the operator's output that is used to modulate itself. If the operator is also being modulated by another waveform, the feedback path and the modulator output are mixed together to modulate the carrier.
6. Use the **Amplitude** fader to determine the output gain that is applied to the current operator after the amplitude envelope.

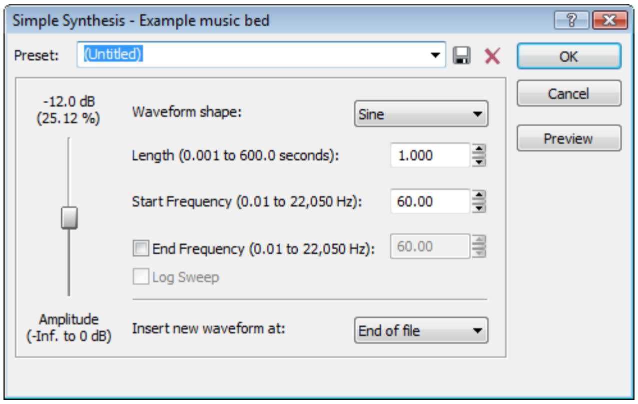
> **Note:** *If the operator is a modulator, this control (along with the envelope) determines the amount of frequency modulation applied to the carrier. If the amplitude of a modulator is high, harsh audio may result.*

Generating simple waveforms

The **Simple Synthesis** tool is used to generate simple waveforms of a given shape, pitch, and length.

1. From the **Insert** menu, choose **Synthesis**, and choose **Simple** from the submenu. The Simple Synthesis dialog appears.

> **Tip:** *You can also click the* **Simple Synthesis** *button ( ) on the Insert toolbar.*



2. From the **Waveform shape** drop-down list, choose a shape to specify the shape of a single period of the current operator's waveform.
3. In the **Length** box, specify the length (in seconds) of the generated waveform.
4. In the **Start Frequency** box, specify the frequency of the waveform.
5. If you want to sweep a range of frequencies, select the **End Frequency** check box and specify an ending frequency in the box.

   Select the **Log Sweep** check box if you want to sweep the range logarithmically; when the check box is cleared, the sweep is linear.
6. Use the **Amplitude** fader to set the peak level of the waveform.

> **Note:** *When you choose* **Noise** *in the* **Waveform shape** *drop-down list, the amplitude is affected by the specified cutoff frequency.*

7. From the **Insert new waveform at** drop-down list, choose a position to determine where the waveform is placed in the data window.
8. Click **OK**.

Chapter 12

Using the Event Tool

The Event tool ( ) in Sound Forge® Pro software allows you to edit multiple audio events in a single data window, which can be an easier way to edit edges and fades and lay out tracks for disc-at-once CDs.

Notes:

- *Events do not loop and cannot exceed the start, end, or channels of the underlying media. For example, you cannot trim an event past its right edge to insert silence.*
- *To preserve events in a file after saving, use a Sound Forge Pro project file. For more information, see* *Working with projects on page 77.*

Creating events

Events are created when you cut/copy/paste sound data, split events, drag sound data to existing data windows, or process selections.

Creating events by cutting, copying, or pasting sound data

Cutting, copying, pasting, or mixing sound data will create events in a data window.

After performing one of these edits, select the Event tool to work with the new event.

Splitting events

From the Edit menu, choose **Event**, and then choose **Split** from the submenu (or press S) to split one or more selected events at the current cursor position.

For more information, see Splitting events on page 174.

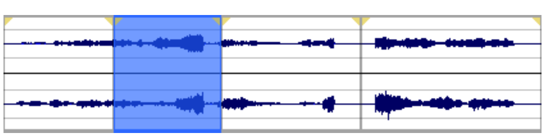
Creating events by dragging files from the Explorer

You can create events in a data window by dragging files from the Explorer window or Windows Explorer.

Tip: *The* **Always open dropped files in new window** *check box on the General page of the Preferences dialog must be cleared if you want to create events with drag-and-drop operations. When the check box is selected, dropping a file on the Sound Forge workspace always creates a new data window.*

1. Drag a file from the Explorer window to a data window. The cursor indicates where the sound data will be added.
2. When you release the mouse button to drop the selection, the selection is pasted, and an event is created.
3. From the Edit menu, choose **Tool**, and then choose **Event** from the submenu.
4. When you click in the data window with the Event tool, events are selected. You can hold Ctrl or Shift to select multiple events.

   In the following example, the data window contains four events, and the second event is highlighted to indicate that it is selected.

Creating events by dragging selections

You can create events in a data window by dragging sound data within a data window or to another data window.

> **Tip:** *The* **Always open dropped files in new window** *check box on the General page of the Preferences dialog must be cleared if you want to create events with drag-and-drop operations. When the check box is selected, dropping a file on the Sound Forge workspace always creates a new data window.*

1. Select the Edit tool ( ).
2. Create a time selection in a data window.
3. Drag the selection to the location where you want to mix or paste it.
4. When you release the mouse button to drop the selection, the selection is pasted or mixed, and an event is created.
5. Select the Event tool ( ). You can then use the Event tool to work with the new event.

Creating events by processing a selection

When you apply processes or effects to a portion of a sound file, Sound Forge creates an event from the selection.

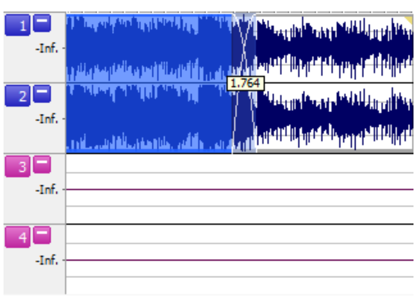
1. Select the Edit tool ( ).
2. Create a time selection in a data window.
3. Apply a process or effect to your time selection. Sound Forge creates an event from the selection.
4. Select the Event tool ( ). You can then use the Event tool to work with the new event.

Moving events

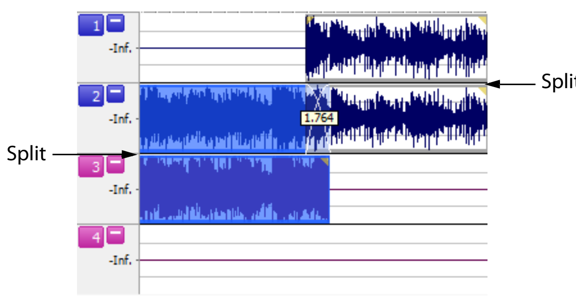
You can move events horizontally along the timeline by dragging them. Moving an event past the end of the current file inserts time into the data window. You must remove any unwanted silence when you are finished editing the file.

When dragging events vertically across channels, you are limited to the number of channels in the current file. You cannot drag an event past the top or bottom channel in a data window to create more channels in the file.

Events can cross channel boundaries, but overlapping events must lie on the same channels. Sound Forge automatically splits or merges event channels as you move events.



Crossfaded events before channel split.



The channels in both events are split when the first event is moved to channels 2 and 3 because overlapping events must lie on the same channels.

For more information on dragging channel boundaries, see Dragging channel boundaries on page 174.

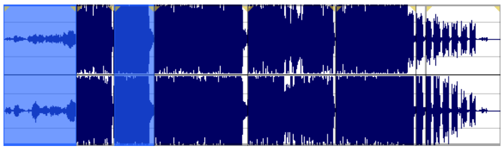
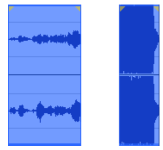
You can also Ctrl+drag events to other data windows or to the workspace to create new data windows.

> **Tip:** *Regions, markers, and envelope points are moved with an event. To turn this feature off, turn off the* **Lock to Selection** > **Markers/Regions** *and* **Envelope Points** *commands on the Options menu.*

Editing events

Copying events

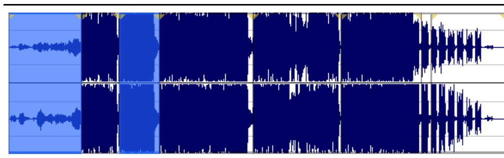
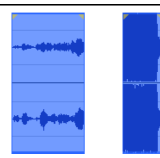
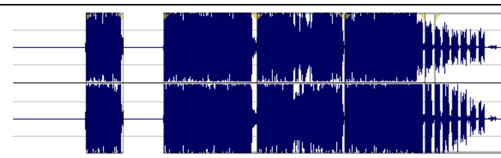
You can copy events to the clipboard and paste them into any data window. You can copy a single event or multiple events. Copying preserves the original event information, edits, and other modifications.

Events before copy	Clipboard contents	Events after copy
		The original events are not affected and do not change.

1. Select the events to be copied. *For more information, see Selecting events on page 179.*
2. From the **Edit** menu, choose **Copy**, or click the **Copy** button ( ) on the Standard toolbar. The selected events are copied to the clipboard, and the waveform is unchanged.

Cutting events

Cutting events removes them from the file, but places the cut information on the clipboard. After the events are on the clipboard, you can paste them into any data window.

Events before cut	Clipboard contents	Events after cut
		

1. Select the events to be cut. *For more information, see Selecting events on page 179.*
2. From the **Edit** menu, choose **Cut**, or click the **Cut** button ( ) on the Standard toolbar. The selected events are removed from the data window and placed on the clipboard.

Pasting events

After events are copied or cut to the clipboard, you can paste them to a different place in the file or to a different data window.

Notes:

- *If any regions or markers are present in the original sound data, they will also be pasted into the destination sound file. To turn this feature off, turn off the* **Lock to Selection** > **Markers/Regions** *command on the Options menu.*
- *Envelope points from the original sound data are not pasted into the destination sound file.*

1. Move the cursor to the desired location on the timeline.
2. From the **Edit** menu, choose **Paste**, or click the **Paste** button ( ) on the Standard toolbar. The clipboard events are inserted into the data window and existing events on the selected channels are moved down the data window by the total length of the pasted audio. If no channels are selected, the clipboard events are pasted to all channels.

Mixing events

You can mix files, events, and selections when using the Event tool by dragging and dropping audio from the current data window, other data windows, the Sound Forge Explorer window, or Windows Explorer.

Mixed audio is inserted as a new events over the existing events in a data window. Crossfades are created, but the Mix/Replace dialog does not appear.

Deleting events

Deleting an event removes the event's audio from the data window. You can delete multiple events at once. Deleting operates like a cutting operation, but the removed information is not placed on the clipboard. *For more information, see Cutting events on page 173*.

1. Select the events to be deleted. *For more information, see Selecting events on page 179*.
2. From the **Edit** menu, choose **Delete (Clear)**, or press Delete.

Tip: *Regions, markers, and envelope points are deleted with an event. To turn this feature off, turn off the* **Lock to Selection** > **Markers/Regions** *and* **Lock Envelope Points** *commands on the Options menu.*

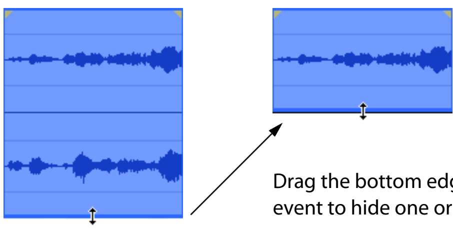
Trimming events

1. Move the cursor over the edge of an event. The cursor changes when properly positioned ().
2. Drag the edge of the event to trim it.

Note: *You cannot trim an event beyond the edges of the underlying media.*

Dragging channel boundaries

You can drag the top and bottom edges of events if you want to use only a subset of the channels in a multichannel file. This method is also useful for duplicating or rearranging channels without using the Channel Converter, which always processes the entire file.



Drag the bottom edge of a multichannel event to hide one or more channels.

You can quickly restore contiguous channels by dragging the event edge back instead of stepping through undo operations.

Splitting events

Splitting an event allows you to adjust a small part of an event or break a single event into multiple sections that you can edit independently.

Splitting events at the cursor

1. Select the events you want to split. *For more information, see Selecting events on page 179*.
2. Position the cursor where you want to split the events.
3. From the Edit menu, choose **Event**, and then choose **Split** from the submenu (or press S):
   - If no events are selected, the events located at the current cursor position will be split throughout all of the channels.
   - If events are selected, only the selected events will be split at the current cursor position.

Splitting events at region boundaries

1. Select the events you want to split. *For more information, see Selecting events on page 179*.
2. From the Edit menu, choose **Event**, and then choose **Split Regions** from the submenu (or press Ctrl+Alt+T):
   - If no events are selected, the events located at the current cursor position will be split throughout all of the channels at region boundaries.
   - If events are selected, only the selected events will be split at region boundaries.

Slipping events

Press Alt while dragging an event. The slip cursor appears ().

As you drag the event, the contents of the event shift, but the event does not move. You can use this technique when you want to maintain an event's length and position, but have the event play a different section of the source audio file.

Tips:

- *Hold the Shift key to temporarily override snapping.*
- *Regions, markers, and envelope points are moved with the contents of the event. To turn this feature off, turn off the* **Lock to Selection** > **Markers/Regions** *and* **Envelope Points** *commands on the Options menu.*

Slip-trimming events

Press Alt while dragging the right or left edge of an event. The slip-trim cursor appears ().

As you drag the event edge, the media moves with the event edge.

Tips:

- *Press Alt+Shift while dragging any portion of an event to slip-trim the right edge. The left edge of the event remains fixed, and the media is slipped past the left edge of the event. This slip mode is useful when you want to slip an event without changing its last frame.*
- *Hold the Shift key to temporarily override snapping.*
- *Regions and markers are moved with the contents of the event. To turn this feature off, turn off the* **Lock to Selection** > **Markers/Regions** *and* **Envelope Points** *commands on the Options menu.*

Auto ripple events

You can ripple the contents of the data window following an edit after performing the following tasks:

- Adjusting an event's length by trimming, slip-trimming, or slipping.
- Moving events.
- Cutting events.
- Pasting events.
- Mixing events.
- Deleting events.

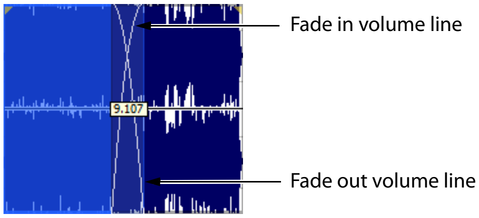
1. Select **Options** > **Event** > **Auto Ripple** (or press Ctrl+Shift+R) to turn on auto ripple.
2. Perform one of the edits listed above.

   The contents of the data window are rippled after the edit.

Note: *If no channels are selected, events are rippled across all channels. If channels are selected, only events on the selected channels are rippled.*

Crossfading events

You can crossfade between two events on the same channel. Crossfading fades out one event’s volume while another event’s volume fades in.

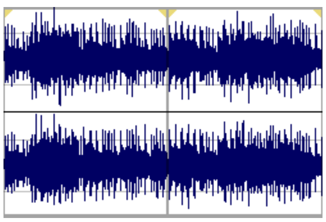


> **Note:** *You can show or hide the crossfade length ToolTip by selecting* **Event** *and then* **Show Crossfade Lengths** *from the* **Options** *menu or pressing Ctrl+Shift+T.*

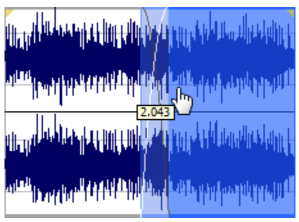
Using automatic crossfades

The automatic crossfades features turns the overlapping portions of two events into a crossfade. From the **Options** menu, choose **Event**, and then choose **Automatic Crossfades** or press Ctrl+Shift+X to turn automatic crossfades on and off.

Events before crossfade



Events after crossfade



Drag one event to overlap the other.

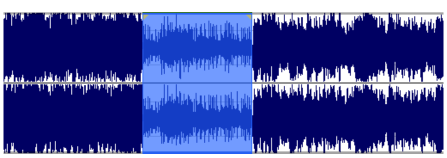
When **Automatic Crossfades** is turned off, overlapping events punch in and out with no fades.

Manually setting a crossfade

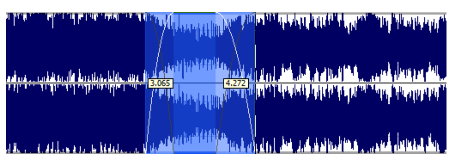
An automatic crossfade is not inserted if a shorter event is placed within the same time frame of a longer event. In this case, the longer event begins playing, then the shorter event plays (punch in), and then the longer event resumes playing. You can manually create a crossfade to fade in or out of the shorter event.

This is a fast and effective method of inserting a voiceover on top of background music (although the music fades out completely) or to replace a bad section of audio.

1. Place the mouse pointer on one of the shorter event’s handles. The fade cursor () appears.
2. Drag the handle to the desired position.



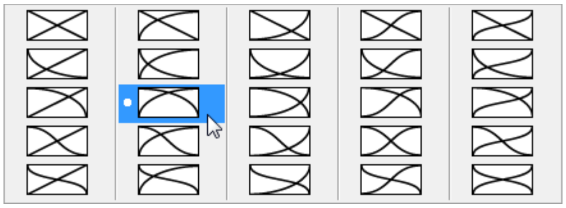
Events without crossfade



Events with manual crossfade

Changing crossfade curves

You can change the crossfade curves that are used to fade in and out between two events.



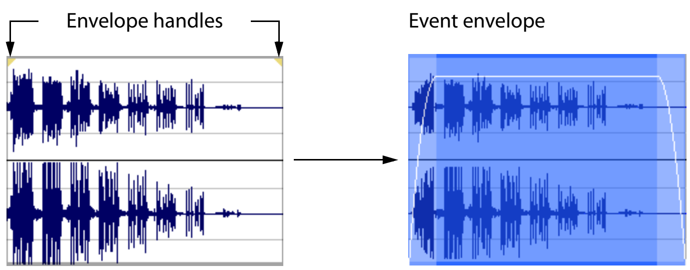
Right-click a crossfade to choose a different crossfade curve.

1. Right-click anywhere in the crossfade region to display a shortcut menu.
2. From the shortcut menu, choose **Fade Type**, and then choose the desired fade type from the submenu.

Using event envelopes (ASR)

You can apply envelopes to individual events. Envelopes, also known as ASRs (attack, sustain, and release), give you the ability to control an event's fade-in, fade-out, and overall volume level.

When you create an event, handles are added that are used to set the envelope. As you drag these handles on events, a volume envelope appears indicating how the event is being affected.

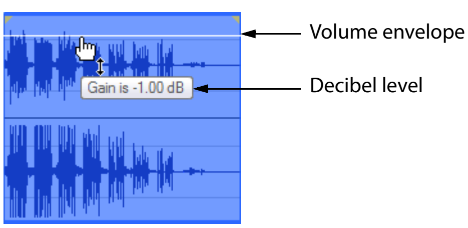


Setting an event's volume

When you place the mouse pointer at the top of an event, the pointer changes to a hand cursor (☝) that you can use to lower the event's overall volume.

1. Place the mouse pointer at the top of the event.
2. When you see the envelope cursor (☝), drag the volume envelope to the desired level. As you drag, the event's decibel level is displayed in a ToolTip.

   You can make fine adjustments by holding Ctrl or clicking the right mouse button while dragging the envelope.



Note: *When you have multiple events selected, the gain of all selected events is adjusted simultaneously.*

Setting an event's fade in and fade out

The event handles allow you to change an event's fade in and out volume. You can also change the type of curve that the event uses to control the volume's fade in or fade out.

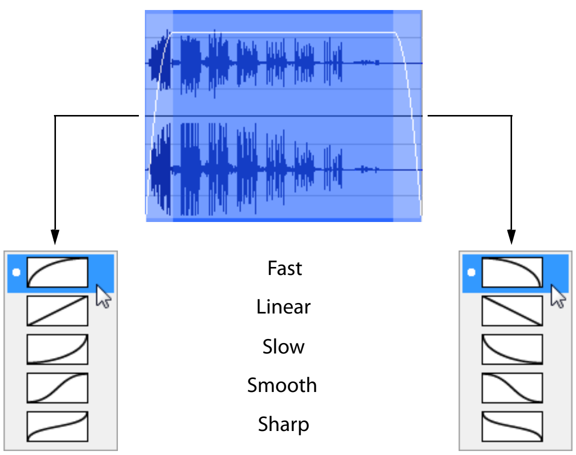
1. Place the mouse pointer on a handle (upper corners of the event). The pointer changes to the fade cursor ( ).
2. Click the corner of the event and drag to create a fade.

To remove a fade, drag the end of the fade curve back to the edge of the event.

Changing an event's fade curve

You can set the shape of the fade curve (fast, linear, slow, smooth, or sharp) that an event uses to raise or lower the volume over time. To access the different fade curves, right-click anywhere in the event's fade-in or fade-out region and choose **Fade Type** from the shortcut menu.

Right-click to select the fade curve type



Applying processes and effects to events

When using the Event tool, processes and effects are applied to each event individually. The processed result is only the length and number of channels in the event at the time of processing, not the entire length and number of channels of the underlying media. You can process multiple selected events at once, but Sound Forge software processes each event separately.

Note: *You cannot use format-changing processes such as Bit-Depth Converter and Resample when using the Event tool.*

Processing multiple events creates a single operation in the Undo/Redo History window. *For more information, see Using the Undo/ Redo History window on page 85.*

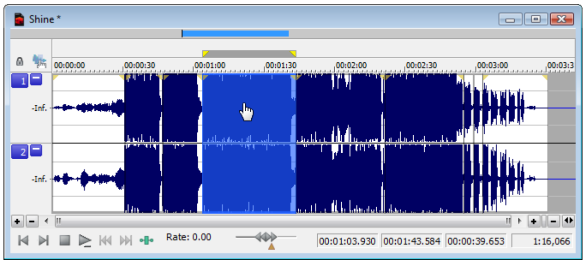
For more information about applying effects, see Applying an effect on page 205.

Selecting events

When working with the Event tool (), you can select one or more events, but you cannot create time selections.

To select an event, click it.

Double-click an event to select it and adjust the loop bar to the length of the event.



Selecting events using menu commands

From the **Edit** menu, choose **Event**, and then choose **Select Next Event** or **Select Previous Event** from the submenu to select the next or previous event in the data window.

Choose **Select First Event** or **Select Last Event** to select the first or last event in the data window.

Choose **Extend to Next Event** or **Extend to Previous Event** to extend the current selection to the next or previous event in the data window.

Choose **Extend to First Event** or **Extend to Last Event** to extend the current selection to the first or last event in the data window.

Selecting events using keyboard shortcuts

You can use the following keyboard shortcuts to select events while using the Event tool:

Command	Keyboard shortcut
Select the next event.	Shift+Right Arrow
Select the previous event.	Shift+Left Arrow
Select the first event.	Shift+Home
Select the last event.	Shift+End
Extend the selection to the next event.	Shift+Ctrl+Right Arrow
Extend the selection to the previous event.	Shift+Ctrl+Left Arrow
Extend the selection to the first event.	Shift+Ctrl+Home
Extend the selection to the last event.	Shift+Ctrl+End

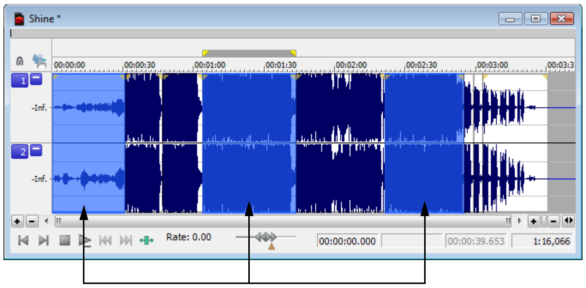
Selecting multiple events

You can select multiple events in your project using two methods.

Selecting nonadjacent events

1. Hold the Ctrl key.
2. Select the events by clicking them.

   To deselect an event, click it again to toggle the event selection on or off.



Selected events

Selecting a range of events

1. Hold the Shift key.
2. Click the first event that you want to select.
3. Click the last event that you want to select.

   All events between the first and last selected events are selected.

Zooming events

To zoom an event, right-click the event and choose **Zoom Event** from the shortcut menu, or press Ctrl+Up Arrow. The event is zoomed to fit the width of the data window.

Snapping to events

When snapping is enabled and you're using the Event tool ( ), you can choose to have events snap to other events in the data window.

From the **Options** menu, choose **Snapping**, and then choose **Events** from the submenu to toggle snapping to event edges.

Snap point indicator

For more information about snapping, see Enable Snapping on page 100.

Tips:

- *To turn snapping to events on and off, press Ctrl+Shift+F8.*
- *Hold the Shift key to temporarily override snapping.*

Chapter 13

Processing Audio

This chapter provides descriptions of processing presets and previews as well as an overview of all functions in the Sound Forge® Pro **Process** menu.

Applying presets

Many Sound Forge dialogs contain drop-down lists of presets used to quickly apply processes and effects. Presets are especially useful when you are learning the application, as they allow you to hear the results of processing as well as view the control settings used to produce these results.

> **Note:** *All information regarding presets in this chapter is applicable to DirectX® Plug-Ins (effects) from Sony Creative Software Inc.*

Using presets

1. Open the Voiceover.pca file.
2. From the **Process** menu, choose **Fade**, and then choose **Graphic** from the submenu. The Graphic Fade dialog is displayed.
3. From the **Preset** drop-down list, choose the **-20 dB exponential fade out** preset. Notice that the dialog's controls change to reflect the -20 dB exponential fade out.
4. Click the **Preview** button. The following actions occur:
   - The **Preview** button changes to a **Stop** button.
   - The effect previews on a brief selection of audio.

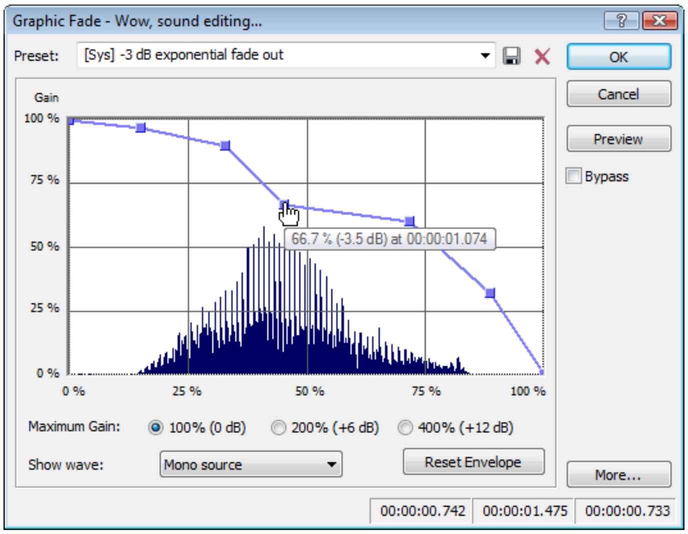
   For more information, see Previewing processed audio on page 183.
5. From the **Preset** drop-down list, choose the **-3 dB exponential fade out** preset. Notice that the dialog's controls update to reflect the new preset and the effect automatically previews.
6. Select the **Bypass** check box. The original audio previews with no effects. *For more information, see Bypassing a process while previewing on page 183.*
7. Clear the **Bypass** check box and click **OK**. The **-3 dB exponential fade out** preset is applied to the audio file.

> **Note:** *An effect or process is not applied to the audio data until you click* **OK**.

Creating presets

You can also create custom effects and save them as presets.

1. Open the Voiceover.pca file.
2. From the **Process** menu, choose **Fade**, and then choose **Graphic** from the submenu. The Graphic Fade dialog is displayed.
3. From the **Preset** drop-down list, choose the **-3 dB exponential fade out** preset. The dialog's controls change to reflect the preset.
4. Drag any of the graphic fade points to a new position.



5. Click the **Save Preset** button ( ). The Save Preset dialog appears.
6. Enter a name for the preset and click **OK**. The new preset is saved and added to the dialog's drop-down list.

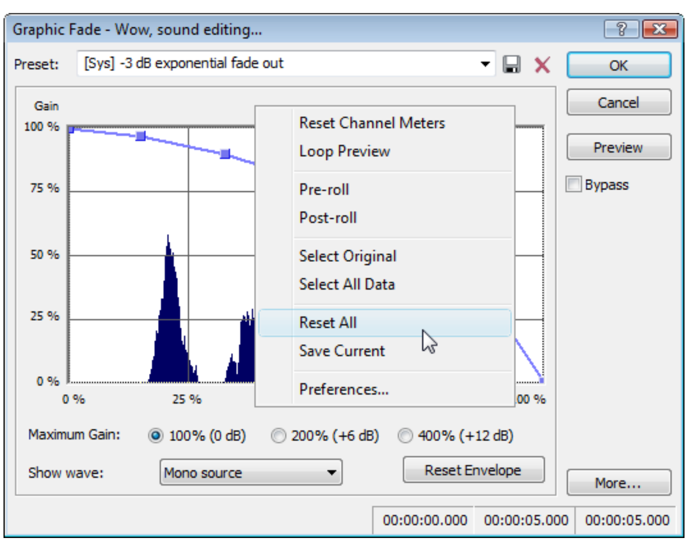
Deleting presets

To delete a preset, choose it from the **Preset** drop-down list and click the **Delete Preset** button ( ).

Note: *Built-in presets cannot be deleted.*

Resetting parameters

To reset all dialog controls to their default settings, right-click the dialog and choose **Reset All** from the shortcut menu.



Managing presets

After you have created custom presets, you can use the Sound Forge Preset Manager to back up, transfer, or delete custom presets from any of the installed effects, processes, tools and plug-ins. You can also use the Preset Manager to manage your ACID® and Vegas® presets. *For more information, see Using the Preset Manager on page 226.*

Previewing processed audio

You can preview the effect that a process has on a file by using the **Preview** button found in most audio processing dialogs. You can use previews to fine-tune effect parameters without leaving the dialog. More importantly, using previews reduces wasted processing time.

Setting custom preview parameters

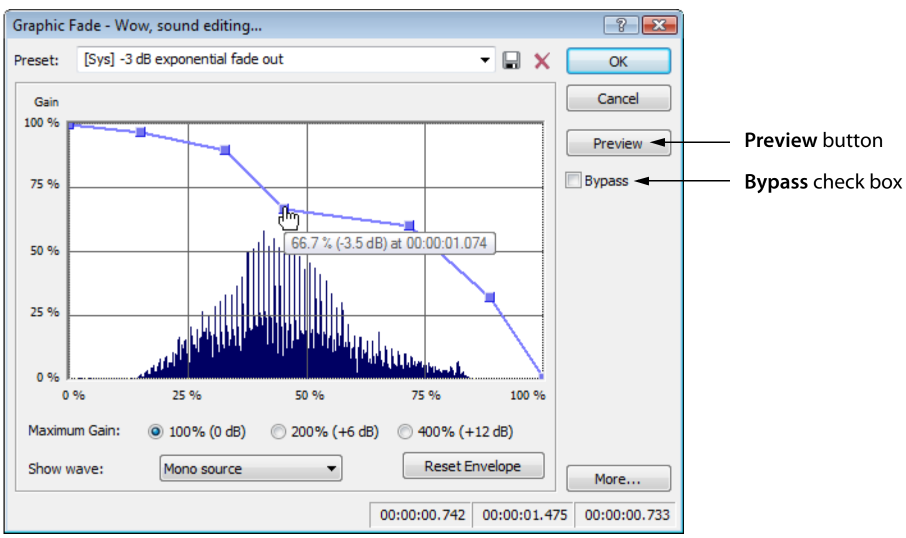
You can customize the preview parameters to satisfy your editing preferences. You can save custom previewing settings for the current process alone or for all processes.

1. From the **Options** menu, choose **Preferences**. The Preferences dialog is displayed.
2. Click the **Previews** tab.
3. Edit the preview parameters as desired. *For more information, see Previews tab on page 339.*
4. Click **OK**. The new preview parameters are updated and saved for all effects.

Bypassing a process while previewing

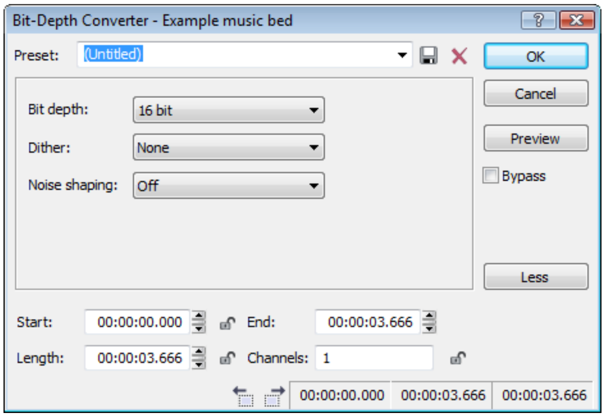
You are also able to A/B test an effect by using the **Bypass** check box to switch between previewing the processed and unprocessed audio file.

- If you select the **Bypass** check box, the unprocessed audio file is played when you click the **Preview** button.
- If you clear the **Bypass** check box, the processed audio file is played when you click the **Preview** button.



Adjusting the data window selection

You can easily adjust your data window selection from within most processing dialogs by clicking the **More** button on the right side of the dialog and specifying the selection parameters explained below.



Clicking the **More** button displays additional information you can use to adjust your data window selection. To hide this information, click the **Less** button.

Control	Description
Start	Determines the starting point for your selection in the data window. The **Length** field adjusts automatically according to your input in this box. Click the **Lock Start** button ( ) if you want to preserve the selection start when adjusting the **Length** or **End** settings. **Note:** *When working with the Event tool ( ), the* **Start** *box is automatically locked.*
End	Determines the ending point for your selection in the data window. The **Length** field adjusts automatically according to your input in this box. **Note:** *When working with the Event tool ( ), the* **End** *box is not available.*
Length	Determines the length of your selection in the data window. Click the **Lock Length** button ( ) to lock or unlock the current selection length. When the selection length is locked, Sound Forge will adjust the values in the **Start** or **End** boxes to retain the specified selection length. **Note:** *When working with the Event tool ( ), the* **Length** *box is automatically locked.*
Channels	Determines the channel(s) included in the data window selection. Type a number in the box to change the channel selection while retaining the start, end, and length selections. Click the **Lock Channels** button ( ) to preserve to **Channels** setting when adjusting the time selection in the data window. **Note:** *When working in event-editing mode, the* **Channels** *box is automatically locked.*
Go to Previous Event **Go to Next Event**	If you're working with the Event tool ( ) and have multiple events selected, you can click the **Go to Previous Event** ( ) or **Go to Next Event** ( ) buttons at the bottom of the processing dialog to navigate events.

Sound Forge processes

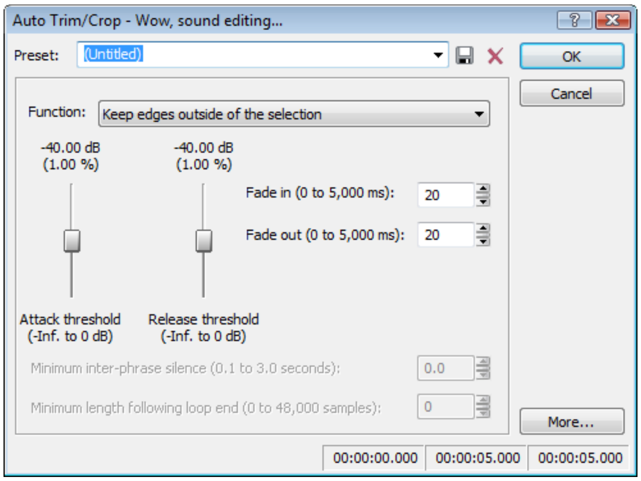
The remainder of this chapter describes the functions located in the **Process** menu.

Auto Trim/Crop

Auto Trim/Crop removes silence from an audio file. In addition, this function automatically fades the endpoints of a phrase.

Using Auto Trim/Crop

1. Open the Voiceover.pca file.
2. From the **Process** menu, choose **Auto Trim/Crop**. The Auto Trim/Crop dialog is displayed.



3. From the **Preset** drop-down list, choose **Phrase Concatenator 1** and click **OK**. The **Auto Trim/Crop** function deletes silence in the file and creates new regions based on the preset's parameters. *For more information, see Auto Trim/Crop controls on page 185.*

Auto Trim/Crop controls

The following controls are located in the Auto Trim/Crop dialog.

Control	Description
Function	This drop-down list contains five modes: • **Keep edges outside of the selection** Removes silence within the selection, but retains all data outside of the selection. • **Remove edges outside of the selection** Removes silence within the selection and deletes all data beyond the selection. • **Remove silence between phrases (creates regions)** Removes silence within the selection and creates regions from individual phrases. *For more information, see Minimum inter-phrase silence on page 186.* • **Remove data beyond loop points** Removes all data beyond the selected loop. *For more information, see Minimum length following loop end on page 186.* • **Remove data from start and limit file length** Allows you to specify an amount of sound to be deleted from the beginning of each file and specify a maximum length for converted files. If a file is longer than this length, it is trimmed. This preset is useful for creating sample clips.
Attack threshold	Determines the threshold level for detection of the trim/crop start point: -Inf. indicates complete silence, and 0 dB indicates maximum amplitude level.
Release threshold	Determines the threshold level for detection of the trim/crop end point: -Inf. indicates complete silence, and 0 dB indicates maximum amplitude level.
Fade in	Determines the length (in milliseconds) of the fade applied to a section of audio prior to the detected trim/crop start point.

Control	Description
Fade out	Determines the length (in milliseconds) of the fade applied to a section of audio following the detected trim/crop end point.
Minimum inter-phrase silence	When you choose the **Remove silence between phrases** mode, the **Minimum inter-phrase silence** value determines the minimum amount of silence needed between phrases for a new region to be created.
Minimum length following loop end	When you choose the **Remove data beyond loop points** mode, the **Minimum length following loop end** value determines the number of samples that must follow a loop.
More	Click this button to view additional options that you can use to adjust your data window selection. *For more information, see Adjusting the data window selection on page 184.*

Bit-Depth Converter

The Bit-Depth Converter is used to increase/decrease the bit depth of a file while concealing the resulting quantization noise.

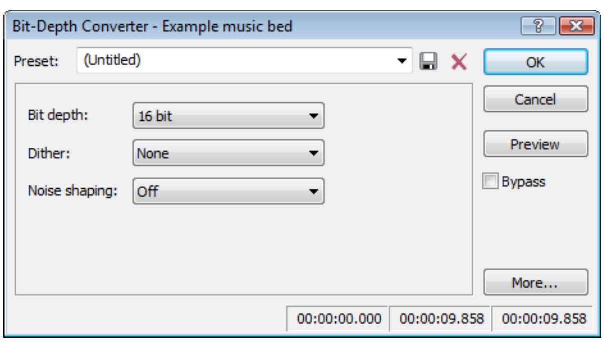
- Decreasing a file's bit depth decreases the overall size of the file, but results in added quantization noise, which can be masked using dither and noise shaping.
- Increasing a file's bit depth—while not improving the quality of the audio—allows subsequent audio processing to be performed with greater accuracy and resolution.

Prior to decreasing a file's bit depth, you should optimize the audio for conversion. *For more information, see Minimizing quantization error on page 109.*

> **Note:** *There are no rules regarding maintaining audio quality when decreasing bit- depth. For this reason, you should always experiment with the* **Dither** *and* **Noise shaping** *controls to determine the optimum settings for each file.*

Converting a file's bit depth

1. Open the Musicbed.pca file.
2. From the **Process** menu, choose **Bit Depth**, and then choose **Bit-Depth Converter** from the submenu. The Bit-Depth Converter dialog is displayed.



3. From the **Bit depth** drop-down list, choose the desired bit depth.
4. If necessary, use the **Dither** drop-down list to specify the type of dither used to mask the quantization noise results from lowering a file's bit depth. *For more information, see Dither on page 187.*
5. If desired, use the **Noise shaping** drop-down list to specify any noise shaping to be applied to the file. *For more information, see Noise shaping on page 187.*

> **Note:** *When increasing a file's bit depth, set the* **Dither** *and* **Noise shaping** *controls to* **None** *and* **Off** *respectively.*

Bit-Depth Converter controls

The following controls are located in the Bit-Depth Converter dialog.

Control	Description
Bit depth	Choose a setting to specify the number of bits that should be used to store each sample.
Dither	This control allows you to specify the randomness of the dither (generated noise) used to mask quantization distortion resulting from conversion to a lower bit depth. You can select from several shapes, each roughly describing the pattern that would be produced if you plotted a graph with the dither amplitude on the X-axis and the probability of the dither values on the Y-axis. As is frequently the case when working with audio, experimentation with dither values yields the best results; however, keep the following information in mind: • **Half Rectangular** Eliminates distortion resulting from conversion to a lower bit depth, but the noise level is more likely to be dependent on the signal. This setting uses a maximum dither noise amplitude of 0.5 LSB (least significant bit). • **Rectangular** Identical to **Half Rectangular**, but with a maximum dither noise amplitude of 1 LSB (least significant bit). • **Triangular** Eliminates distortion products as well as any noise floor modulation, but results in a slightly higher noise level. The option typically works well in conjunction with noise shaping. • **Highpass Triangular** Behaves like triangular dither, but shifts its noise into higher frequencies. This is typically the best option when used in conjunction with noise shaping. • **Gaussian** Does not perform as well as **Rectangular** and **Triangular** dither, but may be suitable for certain audio.
Noise shaping	Determines the aural positioning of quantization noise. Using this control, you can shift the noise into audio registers that are less perceptible to human hearing. This lowers the perceived noise floor and creates the illusion of cleaner audio. **High-pass contour** noise shaping attempts to push all quantization noise and error into high frequencies. **Equal loudness contour** noise shaping attempts to push the noise under an equal loudness-type of curve.
More	Click this button to view additional options that you can use to adjust your data window selection. *For more information, see Adjusting the data window selection on page 184*.

Noise shaping dangers

Noise shaping places quantization noise near the audio's Nyquist frequency, a value equal to one-half of the file's sample rate. Consider the following information:

- A file with a sample rate of 44.1 kHz has a Nyquist frequency of 22.05 kHz (at the high end of human hearing). Applying noise shaping to this file results in audio perceived to be cleaner than it actually is.
- A file with a sample rate of 22 kHz has a Nyquist frequency of 11 kHz (well within the sensitive range of human hearing). Applying noise shaping to this file results in audio that is perceived to be noisier than it actually is. Ironically, this defeats the entire purpose of the **Noise shape** control.

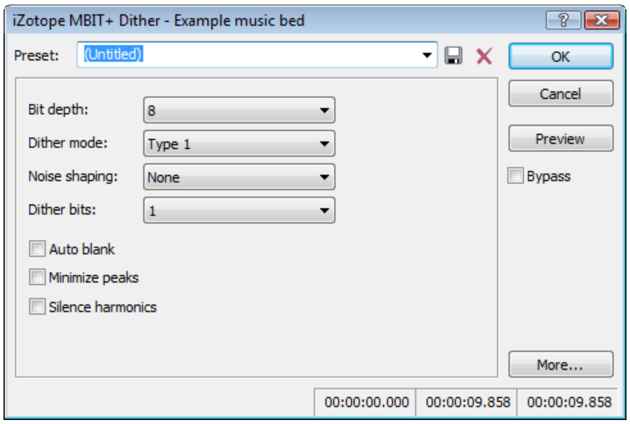
For this reason, we do not recommend using noise shaping on files with sample rates less than 44.1 kHz.

iZotope MBIT+ Dither

From the **Process** menu, choose **Bit Depth**, and then choose **iZotope MBIT+ Dither** from the submenu to convert sound files to different bit depths and apply dithering.

> **Tip:** *Because the signal-to-noise ratio decreases when you decrease the bit depth of a file, you should maximize the volume of the sound file using the Volume or Normalize functions before performing the conversion.*

1. From the **Process** menu, choose **Bit Depth**, and then choose **iZotope MBIT+ Dither** from the submenu. The iZotope MBIT+ Dither dialog is displayed.



2. Choose a setting from the **Preset** drop-down list, or adjust the controls as needed.

Item	Description
Bit depth	Choose the desired bit depth from the drop-down list. **Note:** *Increasing a file's bit depth cannot improve the quality of the existing audio, but does allow higher resolution for processing.*
Dither mode	Choose a setting from the drop-down to choose the type of dithering that will be applied to mask quantization noise. • **Type 1** Uses a traditional rectangular probability distribution function. • **Type 2** Uses a traditional rectangular probability distribution function. • **MBIT+** Uses a proprietary algorithm to offer superior results for all types of source material.
Noise shaping	Choose a setting from the drop-down list to control the amount of noise shaping that will be applied. When **Type 1** or **Type 2** is selected in the **Dither mode** drop-down list, the following settings are available: • **None** No noise shaping is applied. • **Simple** A high-pass filter is applied to the dither noise. • **Clear** Aggressively moves dither noise toward the Nyquist frequency. • **Psych5** Uses a fifth-order filter to move dither noise away from audible frequency bands. • **Psych9** Uses a ninth-order filter to move dither noise away from audible frequency bands. When **MBIT+** is selected in the **Dither mode** drop-down list, you can choose a setting from the **Noise shaping** drop-down list to control the amount of noise shaping applied. Increased settings will provide more audible noise suppression at the expense of a higher noise floor.

Item	Description
Dither bits/amount	When **Type 1** or **Type 2** is selected in the **Dither mode** drop-down list, you can choose a setting from the **Dither bits** drop-down list to choose whether you want to use 1 or 2 dither bits. 1 works well for most applications. When **MBIT+** is selected in the **Dither mode** drop-down list, you can choose a setting from the **Dither amount** drop-down list to control the amount of dithering applied. Normal works well for most applications. The **None** and **Low** settings can leave some nonlinear quantization distortion or dither noise modulation behind. The **High** setting can eliminate nonlinear quantization distortion at the expense of a higher noise floor.
Auto blank	Select this check box if you want the plug-in to suppress dithering noise during silent portions of your audio.
Minimize peaks	Select this check box if you want to suppress peaks in the dither noise signal.
Silence harmonics	Select this check box if dithering distorts the timbre of your audio. When the check box is selected, harmonic quantization distortion is moved away from the overtones of audible frequencies.

3. Click **OK**.

Channel Converter

The Channel Converter is used to change the number of channels in an audio file. The Channel Converter dialog can also be used to reverse the channels of a stereo file or intermix the channels of a multichannel file to create interesting panning effects.

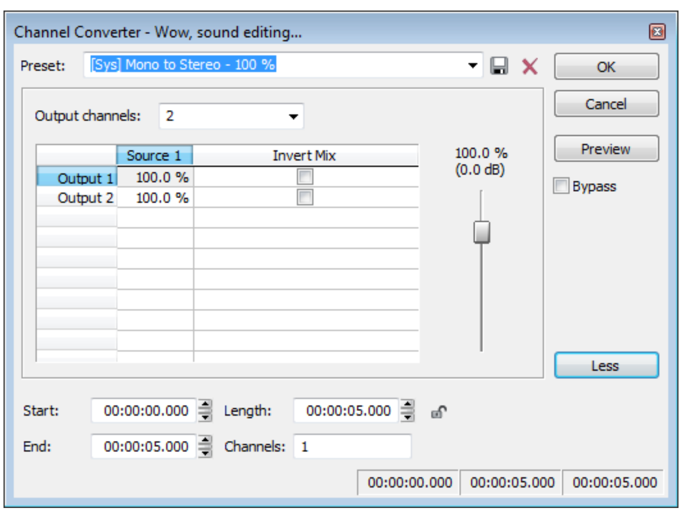
Tip: *To perform quick channel conversion without specifying the mix, use the* ***Audio channels*** *box on the File Properties window or right-click the* ***Channels*** *box in the status bar and choose a setting from the shortcut menu. For more information, see Editing file properties on page 105.*

Notes:

- *If you want to apply a panning envelope to a mono file, use the Channel Converter to convert the file to stereo first.*
- *You can use mono files in the Pan/Expand dialog (accessible from the* ***Process*** *menu by choosing* ***Pan/Expand****) if you choose* ***Pan (preserve stereo separation)*** *or* ***Pan (mix channels before panning)*** *from the* ***Process mode*** *drop-down list. When you click* ***OK*** *to apply your changes, the file will be converted to stereo and your panning settings will be applied.*

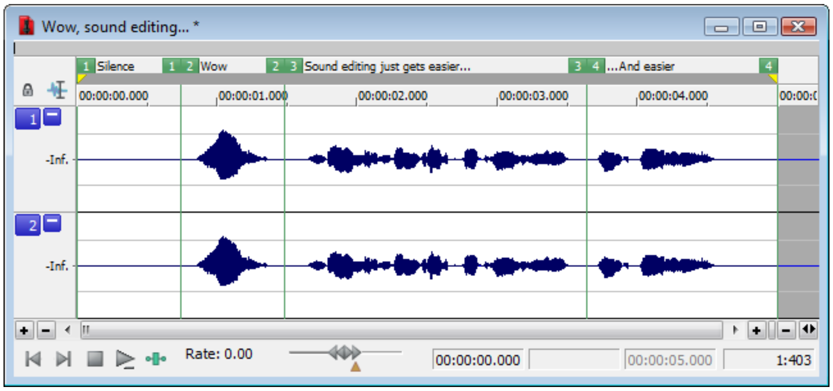
Converting a mono file to stereo (or multichannel)

1. Open the Voiceover.pca file. Notice that this is a mono file.
2. From the **Process** menu, choose **Channel Converter**. The Channel Converter dialog appears.



3. Choose a setting from the **Preset** drop-down list, or adjust the controls as needed:
    a. Choose a setting from the **Output channels** drop-down list to indicate the number of channels in the converted file.
    b. Click in the **Output** box for each output channel and type a gain value (or drag the fader) to adjust the amount of the original mono file that will be mixed to the new channel.
    c. Select the **Invert Mix** check box if you want to reverse the phase of the new channel's content.
4. Click the **OK** button.

    The file is converted to stereo.



Converting a stereo file to mono

1. Open the saxriff.wav file. Notice that this is a stereo file.
2. From the **Process** menu, choose **Channel Converter**.
3. Choose a setting from the **Preset** drop-down list, or adjust the controls as needed:
    a. Choose 1 from the **Output channels** drop-down list to create a mono file.
    b. Click in the **Source 1** box and type a gain value (or drag the fader) to adjust the amount of the original left channel that will be mixed to the new mono file.
    c. Click in the **Source 2** box and type a gain value (or drag the fader) to adjust the amount of the original right channel that will be mixed to the new mono file.
    d. Select the **Invert Mix** check box if you want to reverse the phase of the new left-channel mix.
4. Click the **OK** button.

Intermixing channels in a file

1. From the **Process** menu, choose **Channel Converter**.
2. Adjust the controls as needed:
    a. Choose a setting from the **Output channels** drop-down list to indicate the number of channels in the converted file.
    b. Click in the **Source** box for each output channel and type a gain value (or drag the fader) to adjust the amount of the original channel that will be mixed to the new channel.
    c. Select the **Invert Mix** check box if you want to reverse the phase of the new channel's content.
3. Click the **OK** button.

Swapping stereo channels

1. From the **Process** menu, choose **Channel Converter**.
2. From the **Preset** drop-down list, choose the **Stereo to Stereo - Swap Channels** preset.
3. Click the **OK** button.

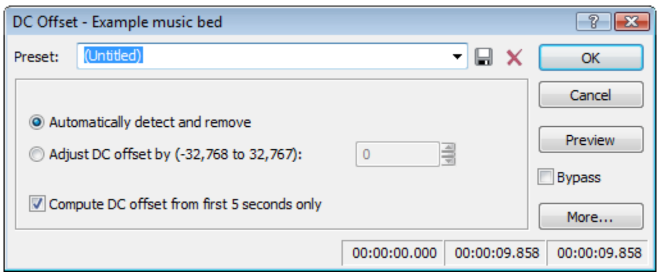
Channel Converter controls

The following controls are located in the Channel Converter dialog.

Control	Description
Output channels	This drop-down determines the number of channels in the output file.
Source	Determines the amount of the original channel data that will be mixed to the new file.
Invert Mix	Select this check box to reverse the polarity of the new channel.
More	Click this button to view additional options that you can use to adjust your data window selection. *For more information, see Adjusting the data window selection on page 184*.

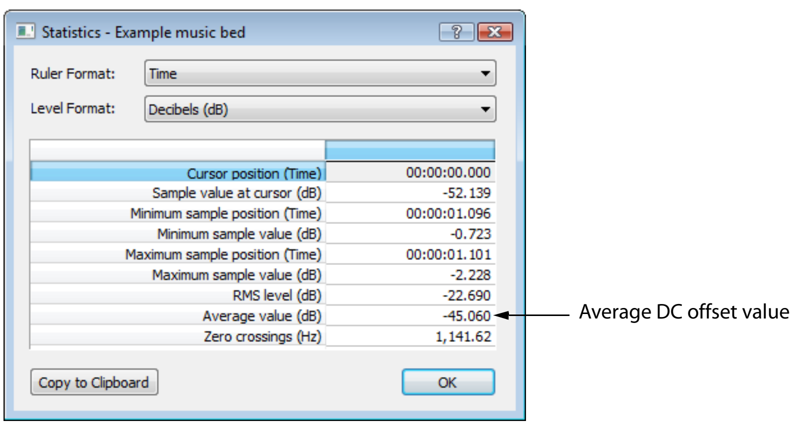
DC Offset

Audio that is not centered around the zero baseline in the waveform display is said to have a DC offset. DC offsets are typically caused by electrical conflicts between the sound card and input device. The **DC Offset** function (located on the **Process** menu) is used to change the baseline of an audio file by adding a constant value to each sample to compensate for offsets.



Estimating DC Offset

You can estimate the DC offset of an audio file by choosing **Statistics** from the **Tools** menu.



DC Offset controls

Choose **DC Offset** from the **Process** menu to display the DC Offset dialog. The following controls are located in the DC Offset dialog.

Control	Description
Automatically detect and remove	Calculates and corrects the DC offset for each channel individually.
Adjust DC offset by	Allows you to specify a DC offset value manually. • -2,147,483,648 to 2,147,483,647 for 32-bit data • -8,388,608 to 8,388,607 for 24-bit data • -32,768 to 32,767 for 16-bit data • -128 to 127 for 8-bit data
Compute DC offset from first 5 seconds only	Selecting this check box specifies that only the first five seconds of a file are analyzed when measuring the DC offset. Be aware that five seconds is not sufficient if the beginning of a file has a long fade-in or mute.

EQ

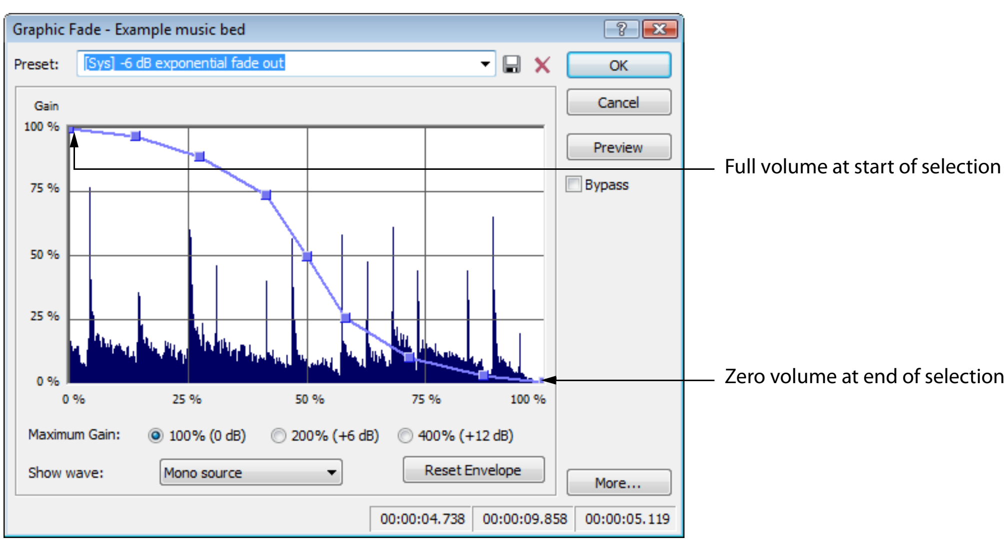
Three **EQ** options are available in the **Process** menu: **Graphic**, **Paragraphic**, and **Parametric**. Each of these options launch the appropriate XFX effect. For more information on using the XFX EQ effects, click the **Help** button (?) in the process dialog.

Fade - Graphic Fade

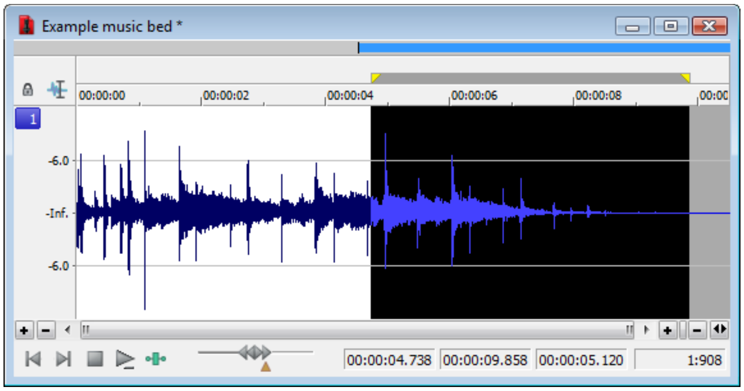
Graphic fade allows you to create custom fade envelopes to apply to audio data. You can use up to 16 envelope points to create complex graphic fades.

Creating a graphic fade

1. Open the Musicbed.pca file.
2. Select the last half of the audio (approximately five seconds).
3. From the **Process** menu, choose **Fade**, and choose **Graphic** from the submenu. The Graphic Fade dialog is displayed.
4. From the **Show wave** drop-down list, choose **Mono source**. The Musicbed.pca waveform is displayed in the graph. *For more information on the dialog controls, see* *Graphic Fade controls on page 193*.
5. From the **Preset** drop-down list, choose **-6 dB exponential fade out**. The fade's envelope is displayed in relation to the waveform in the graph.



6. Click **OK**. The specified fade is applied to the selection.



Creating a custom graphic fade

1. Open the Musicbed.pca file.
2. Select the first half of the audio (approximately five seconds).
3. From the **Process** menu, choose **Fade**, and choose **Graphic** from the submenu. The Graphic Fade dialog is displayed.
4. From the **Show wave** drop-down list, choose **Mono source**. The Musicbed.pca waveform displays in the graph. *For more information on the dialog controls, see Graphic Fade controls on page 193.*
5. Edit the fade envelope using the following controls:
   - Click the envelope to create a new point.
   - Drag a point to move it to a new position.
   - Double-click or right-click a point to delete it.
   - Right-click an envelope segment and choose a new fade type from the shortcut menu.

   For more information, see Envelope graphs on page 49.
6. Click **OK**. The custom graphic fade is applied to the selected audio.

Graphic Fade controls

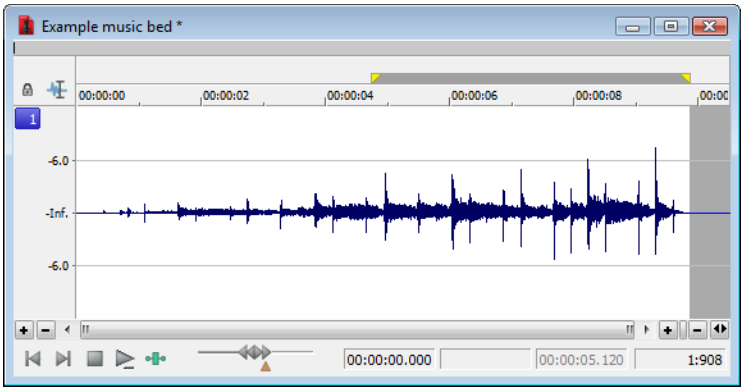
The following controls are located in the Graphic Fade dialog.

Control	Description
Maximum Gain	Select a radio button to adjust the range of the envelope graph.
Show wave	The **Show wave** drop-down list provides several settings for drawing the current selection's waveform on the envelope graph. This function is available only for small selections.
Reset Envelope	Clicking the **Reset Envelope** button clears the envelope of all points except the original two.

Fade - Fade In

The Fade In command is used to linearly fade a selection from a volume of -Inf. to a volume of 0 dB. The size of the selection determines the length of the fade.

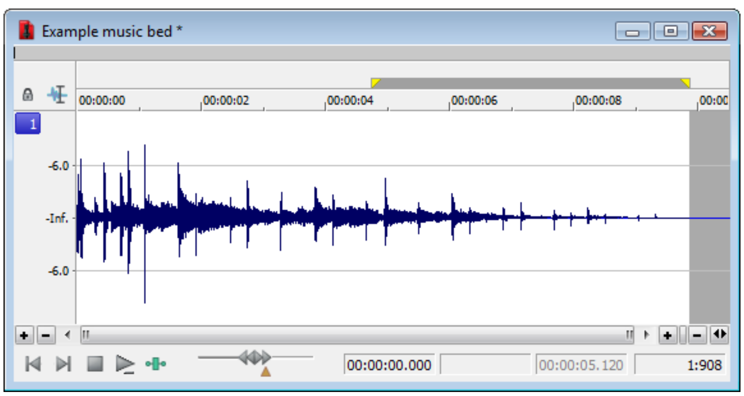
1. Open the Musicbed.pca file.
2. From the **Process** menu, choose **Fade,** and choose **In** from the submenu. The fade is applied, and volume increases over the length of the entire file.



Fade - Fade Out

The Fade Out command is used to linearly fade a selection from a volume of 0 dB to a volume of -Inf. The size of the selection determines the length of the fade.

1. Open the Musicbed.pca file and select all audio data.
2. From the **Process** menu, choose **Fade**, and choose **Out** from the submenu. The fade is applied, and the volume decreases over the length of the entire file.



Invert/Flip

The Invert/Flip command inverts the audio selection at its baseline, in effect reversing its polarity. Inverting a file, while creating no audible difference, is occasionally useful for matching sample transitions when executing certain pastes, mixes, or loops.

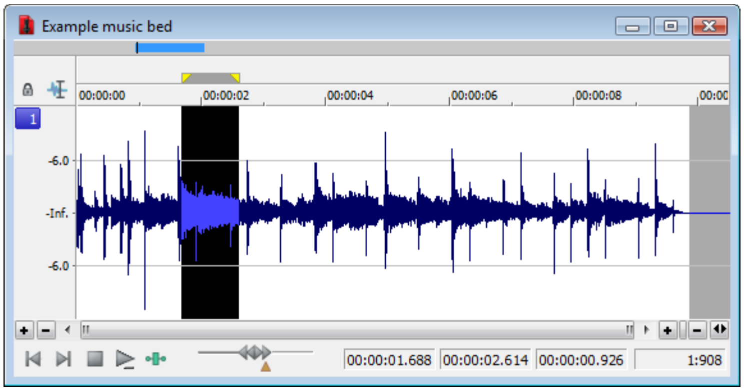
1. Create a selection in the data window.
2. From the **Process** menu, choose **Invert/Flip**. The selection is inverted.

Mute

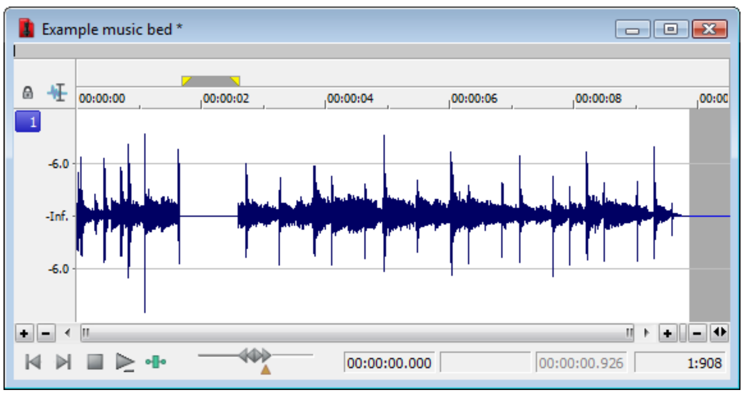
The Mute command forces the selection to a volume of -Inf. dB (silence).

Muting an audio selection

1. Create a selection in the data window.



2. From the **Process** menu, choose **Mute**. The selection is muted.

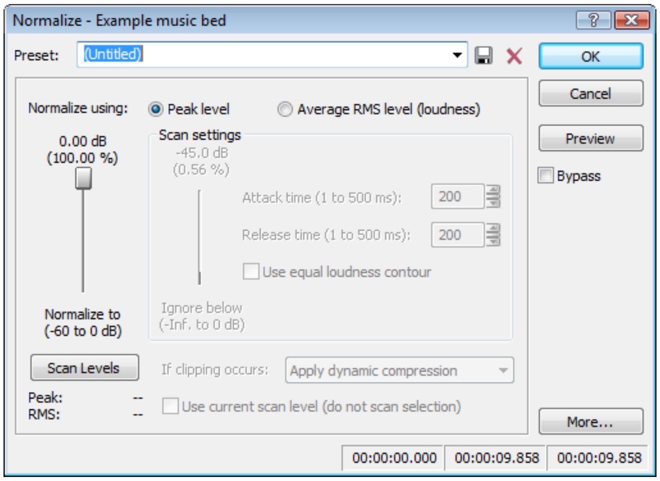


Normalize

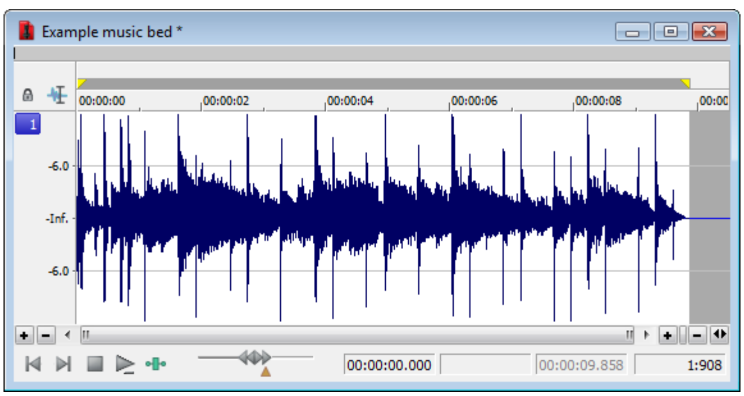
The Normalize command maximizes the overall volume of a file without introducing clipping. When you normalize a file, the entire file is scanned and a constant gain is applied to raise the file's level to a specified value.

Normalizing audio

1. Open the Musicbed.pca file.
2. From the **Process** menu, choose **Normalize**. The Normalize dialog is displayed.



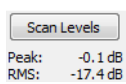
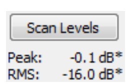
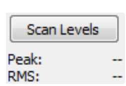
3. From the **Preset** drop-down list, choose **Normalize RMS to -16 dB (music)** and click **OK**. The file is normalized and its overall "loudness" is increased.



Normalize controls

The following controls are located in the Normalize dialog.

Control	Description
Normalize using Peak level	This radio button normalizes the audio file using the maximum (instantaneous) sample values detected. A constant gain is then applied to the audio.
Normalize using Average RMS level (loudness)	This radio button normalizes the audio file using the detected average RMS value of the audio file. This is helpful for matching the apparent loudness of a number of individual recordings.

Control	Description
Normalize to	This fader specifies the level to which the highest peak should be set. With **Peak level**, if the peak level is -10 dB and the **Normalize to** value is -3 dB, a constant boost of 7 dB is applied to the entire file. With **Average RMS level**, normalizing to 0 dB means boosting the signal until it has the same apparent loudness as a 0 dB square wave. This results in all the dynamic range of the signal being flattened and all peaks being either clipped or seriously compressed. **Note:** *As a rule, normalizing using* **Peak levels** *to 0 dB is acceptable, but normalizing using* **Average RMS level** *to anything above -6 dB is not recommended.*
Ignore below	Determines the level of audio data included in the RMS calculation. Data below the threshold is ignored, effectively eliminating silent sections from RMS calculation. The **Ignore below** fader should be set a few dB above perceived silence. If **Ignore below** is set to -Inf., all audio data is used. However, if the value is set too high (above -10 dB) the RMS value may never rise above the threshold. In this case, normalization cannot occur. For this reason, you should evaluate the threshold by clicking the **Scan Levels** button.
Attack time	Determines how quickly the scan responds to transient peaks.
Release time	Determines how quickly the scan should stop using transient peak material after it begins to drop in level. Slower release times result in more data being included in RMS calculation.
Use equal loudness contour	Allows the scan to compensate for the Fletcher-Munson Equal Loudness Contours. The Fletcher-Munson Equal Loudness Contours illustrate that very low- and high-frequency audio is less perceptible to the human ear than mid-range audio. Therefore, selecting this option forces the scan to factor this into RMS calculation.
Scan Levels	Clicking **Scan Levels** initiates Peak and RMS scans on the audio and displays the RMS level and the highest peak level detected. When previewing a normalize effect, the entire file must be scanned to preview even a small selection. Clicking **Scan Levels** saves the current Peak and RMS values and allows you to preview different **Normalize to** settings without re-scanning the entire file.  Current Peak and RMS levels An asterisk adjacent to a level value indicates that the value is not current. This occurs when the selection is updated or the dialog is initially opened. To update values, click **Scan Levels**.  Noncurrent Peak and RMS levels If values have never been calculated, two dashes display. Click Scan Levels to calculate values.  Nonexistent Peak and RMS levels **Note:** *If the RMS level never reaches the Ignore below threshold, a value of -96 dB displays. If this occurs, decrease the Ignore below threshold level and rescan.*

Control	Description
If clipping occurs	The **If clipping occurs** drop-down list is used to specify how the normalize function handles clipping that may occur when an audio file is processed using the RMS option. This list provides four options: • **Apply dynamic compression** Audio peaks that will result in clipping are limited below 0 dB using non-zero attack and release times to minimize distortion. This mode is useful for getting loud and clear audio during mastering. • **Normalize peak value to 0 dB** The selection's peak amplitude level is normalized to 0 dB, thereby allowing the maximum possible constant gain without clipping the selection. However, less gain is applied than would be necessary to achieve the Normalize to RMS level. • **Ignore (saturate)** Audio is permitted to clip and distort. • **Stop processing** Audio peaks that will result in clipping force the normalize function to cease processing and alert you that clipping will occur at the current level. **Note:** *When normalizing multichannel audio, normalization is computed on the loudest sample value found in a channel and identical gain is applied to all channels. If a single channel is selected in a multichannel file, normalization processes only that channel.*
Use current scan level (do not scan selection)	When you select the **Use current scan level** check box, the current scan levels are used without initiating a new scan. This is useful when applying scan levels from a different selection or file to the current selection, thereby allowing identical gains to be applied to multiple files. This option can also be used to scan a selection of an audio file containing the loudest or most constant levels and then apply that scan to normalize the entire file.

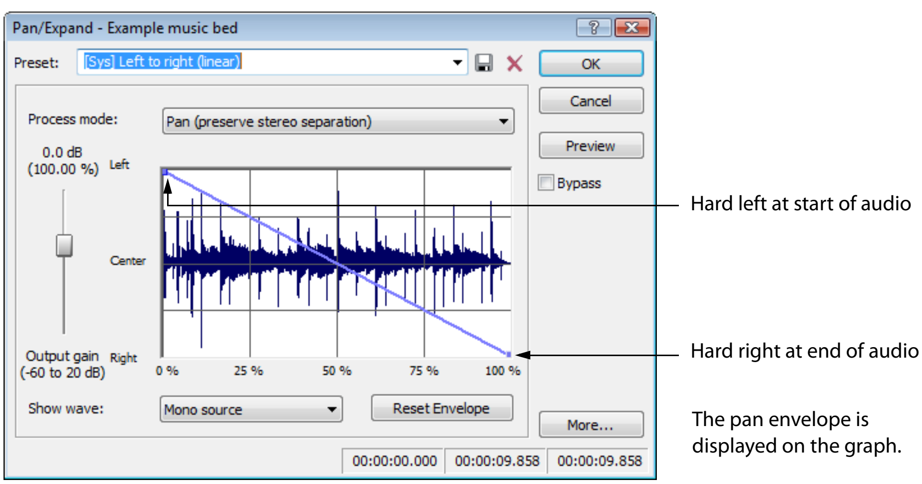
Pan/Expand

Pan/Expand allows you to create panning effects and stereo compression/expansion in selections.

Creating a pan

A pan is used to control the apparent position of a sound between the left and right channels of a stereo file.

1. Open the Musicbed.pca file.
2. From the **Process** menu, choose **Pan/Expand**. The Pan/Expand dialog is displayed.
3. From the **Preset** drop-down list, choose **Left to right (linear)**. The pan envelope is displayed on the graph.



4. Click **OK**. The file is converted to stereo and a left-to-right pan is added.

5. Play the file. The audio source seems to move from the left channel to the right channel during playback.

Note: *A pan, by nature, cannot be created in a mono file.*

Creating a custom pan

You can create complex custom panning effects using up to 16 envelope points.

1. Open the Musicbed.pca file.
2. From the **Process** menu, choose **Pan/Expand**. The Pan/Expand dialog is displayed.
3. Configure the pan envelope using the following controls:
   - Click the envelope to create a new point.
   - Drag a point to move it to a new position.
   - Double-click or right-click a point to delete it.
   - Right-click an envelope segment and choose a new fade type from the shortcut menu.

   For more information, see Envelope graphs on page 49.
4. Click **OK**. The custom pan is applied to the file.

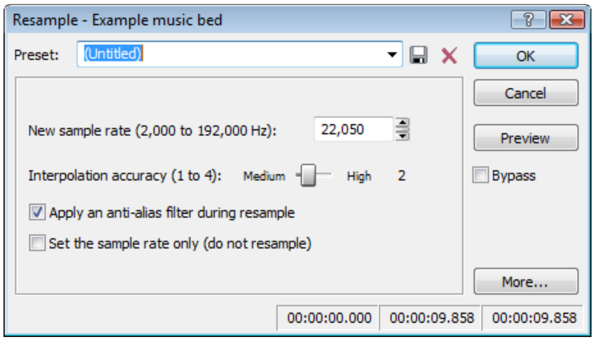
Pan/Expand controls

The following controls are located in the Pan/Expand dialog.

Control	Description
Process mode	The **Process mode** drop-down list contains the following options: • **Pan (preserve stereo separation)** Applies the pan effect without mixing the channels, thereby simulating the spectral positioning of stereo recordings. • **Pan (mix channels before panning)** Mixes the left and right channels prior to applying panning effects. • **Stereo expand** Allows you to contract or expand the image of stereo audio from dead center (mono) to completely panned wide (no center channel). • **Mix mid-side (MS) recording to left and right channels** Simulates a recording technique in which one microphone is pointed directly at the source and used to record the center (mid) channel, and a second microphone is pointed 90 degrees away from the source (side) and used to record the stereo image. For proper playback on most systems, MS recordings must be converted to standard left/right orientation. To convert an MS-recorded track to a left/right track, first ensure that the center channel is in the left track and the side channel on the right. The MS mix function is then used to set the width of the stereo image for the converted track.
Output gain	Determines the amount of gain applied to the signal following pan/expand processing.
Show wave	The **Show wave** drop-down list provides several settings for drawing the current selection's waveform on the envelope graph. This function is available only for small selections.
Reset Envelope	Clicking the **Reset Envelope** button clears all but the two original envelope points. • For the **Pan** modes, these two points prevent unintended panning. • For the **Stereo expand** and **Mix Mid-Side** modes, these two points prevent unintended expansion.

Resample

The **Resample** command allows you to change the sampling rate of a file without altering its pitch or duration.



- Resampling to a lower sample rate results in less frequent samples and a decreased file size, but adds aliasing noise to the audio. *For more information, see Apply an anti-alias filter during resample on page 201.*
- Resampling to a higher sample rate results in extra samples being created through interpolation and an increased file size. Like increasing bit depth, up-sampling does not improve the quality of an audio file, but permits subsequent audio processing to be performed with greater precision.

Downsampling audio

1. Open the Musicbed.pca file.
2. Right-click the data window and choose **File Properties** from the shortcut menu. The File Properties window is displayed. Notice that this file has 44,100 Hz sample rate and a file size of 0.48 MB.
3. Click **OK**.
4. From the **Process** menu, choose **Resample**, and then choose **Resample** from the submenu. The Resample dialog is displayed.
5. From the **Preset** drop-down list, choose **Resample to 8,000 Hz with anti-alias filter** and click **OK**. The audio is resampled at 8,000 Hz.
6. From the **File** menu, choose **Save As.** Save the resampled file with a new name and close it.
7. Open the resampled file and view its File Properties window. The sample rate is lower (8,000 Hz) and the file size is smaller.
8. Play the file. Notice the obvious decrease in audio quality.

Note: *Use this new file to perform the following up-sampling procedure.*

Upsampling audio

1. Verify that the file created in the previous procedure is the active data window.
2. From the **Process** menu, choose **Resample**, and then choose **Resample** from the submenu. The Resample dialog is displayed.
3. From the **Preset** drop-down list, choose **Resample to 48,000 Hz with anti-alias filter** and click **OK**. The audio is resampled at 48,000 Hz.
4. From the **File** menu, choose **Save As.** Save the resampled file with a new name and close it.
5. Open the new file and view its File Properties window. Notice that the sample rate is higher (48,000 Hz) and the file size is larger.
6. Play the file. Notice that resampling to a higher sample rate produces an audio quality at 48,000 Hz that is indistinguishable from the quality at 8,000 Hz.

Resample controls

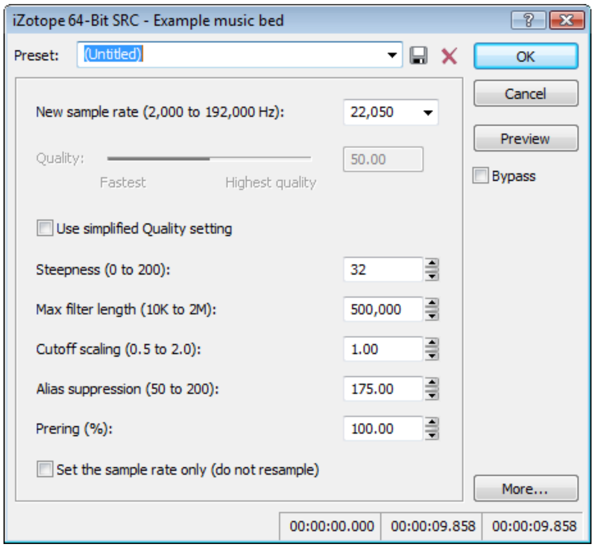
The following controls are located in the Resample dialog.

Control	Description
New sample rate	Determines the sample rate (in Hz) at which the file is resampled. **Tip:** *Processing is quicker when downsampling by an even multiple (such as when going from 44 kHz to 22 kHz).*
Interpolation accuracy	The **Interpolation accuracy** value determines the complexity of the interpolation method used during resampling. Interpolation accuracy is most apparent in high frequencies, but the audible difference between the values is subtle and often undetectable without the use of test tones. A value of 1 is suitable for general-purpose audio. A value of 2 or 3 is good for high-end audio applications. A value of 4 results in professional-quality audio, but requires substantial processing.
Apply an anti-alias filter during resample	Selecting this check box applies an anti-aliasing filter during the resampling process. Remember that the maximum frequency that can be represented by a sample rate is one-half of the sampling rate (the Nyquist frequency). Therefore, high frequencies cannot be accurately represented when downsampling. The anti-aliasing filter prevents high frequencies from becoming low-frequency distortion. **Tip:** *It is also advisable to apply an anti-aliasing low-pass filter to an audio file prior to resampling to a lower sample rate.*
Set the sample rate only (do not resample)	If this check box is selected, the playback rate is changed without resampling the data. This means that the pitch of the original file is not preserved. For this reason, this option is only useful for quickly converting between two similar sample rates.

iZotope 64-Bit SRC

You can use the iZotope 64-Bit SRC process to change the sample rate of an existing sound file.

1. From the **Process** menu, choose **Resample**, and then choose **iZotope 64-Bit SRC** from the submenu. The iZotope 64-Bit DSRC dialog is displayed.



2. Choose a setting from the **Preset** drop-down list, or adjust the controls as needed.

Item	Description
New sample rate	Specify the sample rate to which the sound file will be converted. **Note:** *Increasing a file's sample rate cannot improve the quality of the existing audio, but does allow higher resolution for processing.*
Quality	When the **Use simplified Quality setting** check box is selected, you can drag the **Quality** slider to adjust the plug-in's controls automatically. Dragging the slider sets the balance of audio quality vs. processing speed. A setting of 50% works well for most applications. Increasing the setting improves the quality of the plug-in but requires more processing power. When the **Use simplified Quality setting** check box is cleared, you can adjust the plug-in's controls manually.
Steepness	The plug-in uses a low-pass filter to discard frequencies that cannot be represented or are undesirable in your audio output. This setting establishes the steepness of the transition band of the low-pass filter. Higher settings will reject unwanted frequencies, but can cause more ringing in the time domain and a higher CPU load.
Max filter length	Sets the maximum length of the filters used for resampling. The default setting will work well for most applications, but you can increase the setting if very high-quality output is desired for uncommon source or destination sampling rates.
Cutoff scaling	Allows you to scale the cutoff frequency of the plug-in's low-pass filter from the Nyquist frequency. Typical values are near 1. Higher values will offer a flatter pass-band, and lower values will offer better aliasing suppression.
Alias suppression	Sets the amount of suppression in the low-pass filter's stop-band. Frequencies in the stop-band that are not fully attenuated will result in aliasing.Higher settings will result in better quality, and lower settings can minimize CPU load.

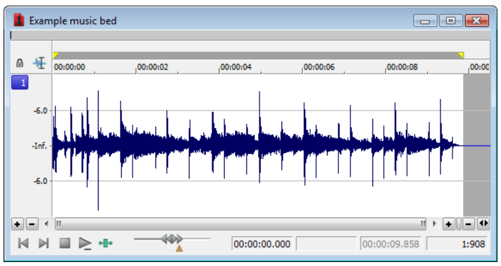
Item	Description
Prering	Low-pass filters are characterized by the amount of ringing they introduce into their output. Higher **Steepness** settings produce increased ringing. A setting of 100% produces a linear phase filter with equal pre and post ringing. A setting of 0% produces a minimum phase filter that offers no preringing but has nonlinear phase distortion. Intermediate settings allow a tradeoff between preringing and postringing and allows you to linearize phase in the pass-band.
Set the sample rate only (do not resample)	Select this check box to change the playback rate without resampling the data. This means that the original pitch of the file is not preserved.

3. Click **OK**.

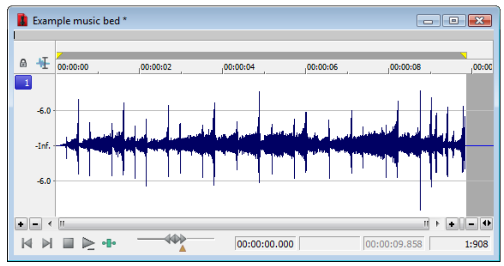
Reverse

The Reverse command reverses the audio selection.

1. Open the Musicbed.pca file.
2. From the **Process** menu, choose **Reverse**. The reversed audio data displays in the data window.



Original audio data



Reversed audio

Rotate Audio

You can move the beginning of a loop to the end, or the end of a loop to the beginning by rotating audio. *For more information, see Rotating audio on page 286.*

Smooth/Enhance

To start the XFX Smooth/Enhance plug-in, choose **Smooth/Enhance** from the **Process** menu. For more information on using the XFX Smooth/Enhance plug-in, click the **Help** button () in the Smooth/Enhance dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Time - Time Stretch

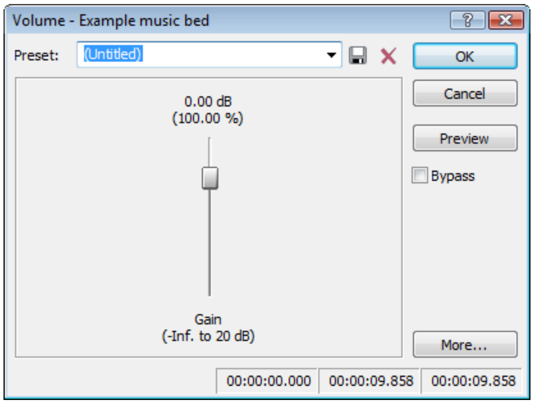
To start the XFX Time Stretch plug-in, choose **Time Stretch** from the **Process** menu. For more information on using the XFX Time Stretch plug-in, click the **Help** button () in the Time Stretch dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Time - élastique Timestretch

To start the élastique Timestretch plug-in, choose **Time** from the **Process** menu and then choose **élastique Timestretch** from the submenu. For information about using the élastique Timestretch plug-in, click the **Help** button () in the élastique Timestretch dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Volume

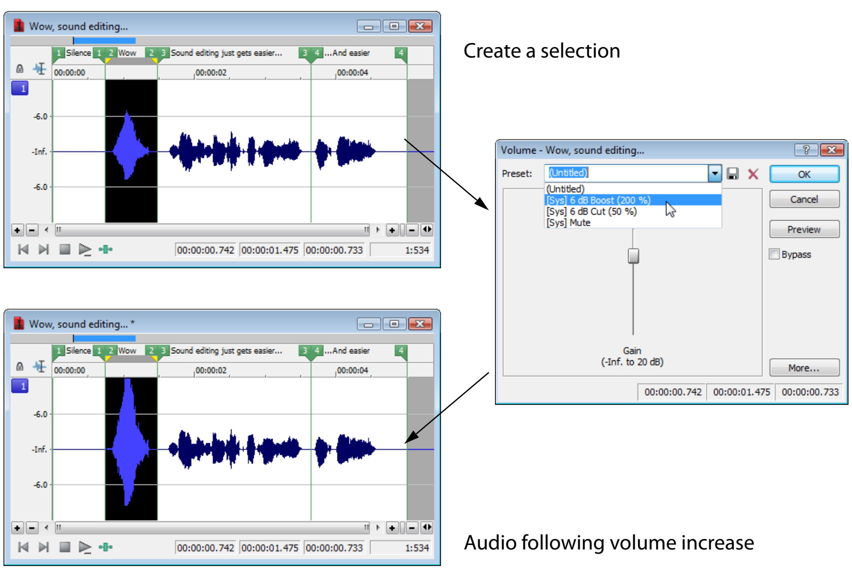
The Volume command alters the volume of an audio selection.



Increasing the volume of a selection

1. Open the Voiceover.pca file.
2. Create a selection containing the word "Wow."
3. From the **Process** menu, choose **Volume**. The Volume dialog is displayed.
4. From the **Preset** drop-down list, choose **6 dB boost (200%)** and click **OK**. The specified boost is applied to the selection.
5. Play the file. The "Wow" data clips and distorts upon playback.

   Exercise caution when using the **Volume** command. Unlike Normalize, Volume performs no pre-processing scans and offers no options for clipping audio data.



Create a selection

Audio following volume increase

> **Note:** *When audio data is clipped, it cannot be restored by performing a second Volume operation. The initial Volume operation must be undone.*

Volume control

The Volume dialog contains only one control: **Gain**. The Gain fader determines the new volume of a selection. Negative decibel values decrease the selection's volume, while positive decibel values increase the selection's volume.

> **Note:** *A value of -Inf. corresponds to mute (0%).*

Chapter 14

Working with Effects

Effects, or plug-ins, can be used to improve the quality of the audio or to create special artistic effects. Additional DirectX® and VST plug-in effects, both from Sony and other third-party vendors, can also be used.

Adding effects

You can choose an effect from the **Effects** menu to apply to a file or just a portion of a file. If you've added an effect to the **FX Favorites** menu, you can select it from that location as well. *For more information, see Organizing effects in the FX Favorites menu on page 208.*

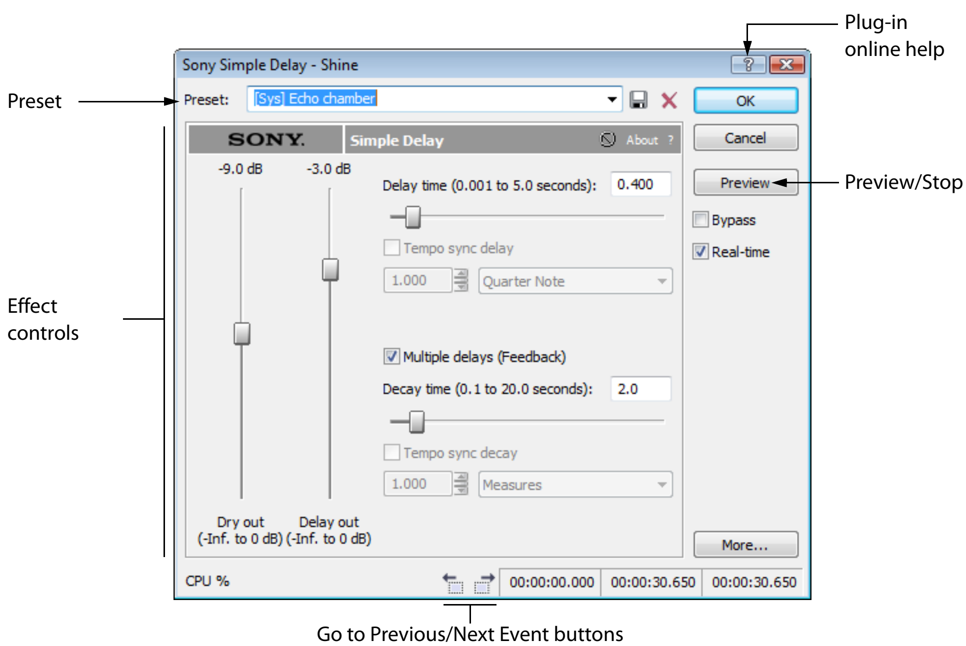
Applying an effect

1. Select the data you want to process. If no data is selected, the effect is applied to the entire file.

   If you're using the Event tool ( ), select the events you want to process. *For more information, see Selecting events on page 179.*

> **Note:** *When you're working with multichannel files, only the selected region in the selected channel is processed. Most functions can be applied to an individual channel or all channels. However, because the channels in a multichannel file must be equal in length, functions that affect the length of the data cannot be performed on individual channels. These functions include Insert Silence, Resample, Time Stretch, Gapper/Snipper, Pitch Bend, and Pitch Shift (without preserving duration).*
>
> *If you want to apply one of these processes in a single channel, convert the file into separate mono files (you can select a channel and drag it to the Sound Forge workspace to create a new file quickly), apply the process, and merge the files into a new multichannel file.*

2. Choose a command from the **Process**, **Effects**, or **FX Favorites** menu. The dialog for the selected effect is displayed.



3. Choose a preset from the **Preset** drop-down list and adjust the parameters in the dialog to achieve the effect you want. For help on the different controls in the effect dialog, click the **Help** button ( ).

4. Click the **Preview** button to test out the effect. Adjust the settings as needed and click **Stop** to end the preview.

> **Tips:**
>
> - *If the selection you made in the data window needs to be adjusted, click the* **Selection** *button to adjust the selection.*
> - *When using the Event tool ( ), click the* **Go to Previous Event** *( ) and* **Go to Next Event** *( ) buttons to navigate to the events you selected and preview and modify the effect for each event.*

5. Click **OK**. During processing, a progress meter is displayed at the bottom of the data window. You can cancel the operation at any time by clicking the **Cancel** button to the left of the progress meter, or you can press the Escape key.

Saving effect settings as a custom preset

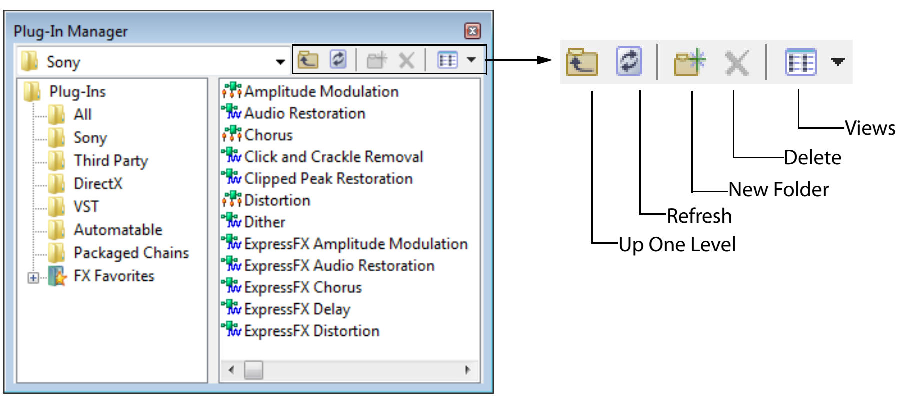
After you have adjusted the parameters in the effect dialog, you may want to save your settings as a custom preset for later use. You can select the preset from the **Preset** drop-down list to apply the same settings at a later time.

1. Adjust the parameters in the effect dialog to achieve the effect you want.
2. Click the **Save Preset** button ( ). The Save Preset dialog is displayed.
3. Enter a new preset name and click **OK**. The new preset is added to the **Preset** drop-down list.

Using the Plug-In Manager

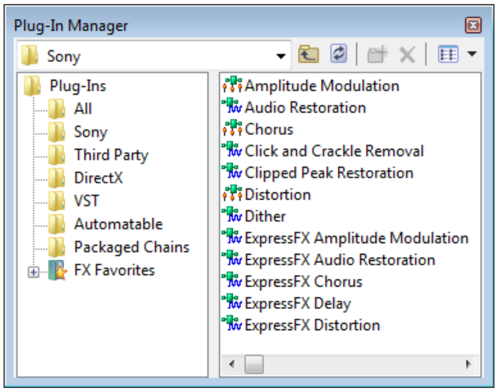
You have several tools to help you manage your plug-ins, including the Plug-In Manager, **FX Favorites** menu, and the Preset Manager.

The Plug-In Manager window not only allows you to add plug-ins and saved plug-in chains, but it also provides a way to manage your plug-in files—to rename plug-ins, hide plug-ins, create folders, add plug-ins to an FX Favorites folder, and perform other standard file management tasks.



Applying a plug-in or chain to a media file

You can add a plug-in to a chain in the Plug-In Chain by dragging an effect from the Plug-In Manager window.



1. From the **View** menu, choose **Plug-In Manager**. The Plug-In Manager window is displayed.
2. Select the data you want to process. If no data is selected, processing will be applied to the entire file.

> **Note:** *When you're working with multichannel files, only the selected region in the selected channel is processed. Most functions can be applied to individual or all channels. However, since all channels in a multichannel file must be equal in length, functions that affect the length of the data cannot be performed on individual channels. These functions include Insert Silence, Resample, Time Stretch, Gapper/Snipper, Pitch Bend, and Pitch Shift (without preserving duration).*
>
> *If you want to apply one of these processes in a single channel, convert the file into separate mono files (you can select a channel and drag it to the Sound Forge workspace to create a new file quickly), apply the process, and merge the files into a new multichannel file.*

3. Select the desired plug-ins.

> **Note:** *Effects chains—including packages created in Vegas or ACID—appear in the DirectX Chains folder in the Plug-In Manager.*

4. Drag the plug-ins from the Plug-In Manager window to the Plug-In Chain window. The selected plug-ins are added to the chain.

> **Tip:** *You can also drag plug-ins or a plug-in chain from the Plug-In Manager window to a data window. The Plug-In Chain window is opened with the selected effects in a new chain.*

5. Use the Plug-In Chain to preview your effects and adjust settings as needed. You can select the **Bypass** button ( ) to hear the original, unprocessed audio.
6. When you are satisfied with the chain, click the **Process Selection** button ( ) in the Plug-In Chain to apply the effect.

Renaming a plug-in

You can customize the names of plug-ins within the software.

1. Right-click a plug-in in the Plug-In Manager and choose **Rename** from the shortcut menu.
2. Type a new name and press Enter.

Hiding a plug-in

All DirectX plug-ins on your system are automatically available to you. You may want to hide a plug-in within the software without removing the plug-in from your system.

1. Right-click a plug-in in the Plug-In Manager and choose **Hide** from the shortcut menu. You are prompted to confirm that you want to permanently hide the plug-in.
2. Click **Yes**. The plug-in no longer appears in Sound Forge software.

> **Tip:** *To restore hidden plug-ins, you can force Sound Forge to rescan your system for plug-ins by deleting the HKEY_CURRENT_USER\Software\Sony Creative Software\Sound Forge Pro\11.0\DXCache key in the Windows Registry.*

Organizing effects in the FX Favorites menu

The **FX Favorites** menu provides easy access to the plug-ins you use most frequently. You can add and remove plug-ins and folders to organize the menu however you like. You can also automatically add all plug-ins on your system to the menu. *For more information, see Automatically adding and organizing plug-ins on page 208.*

Once you add a plug-in to the **FX Favorites** menu, you can apply the plug-in to a file by selecting it from the menu. *For more information, see Adding effects on page 205.*

1. From the **FX Favorites** menu, choose **Organize**. The Organize Favorites dialog is displayed.
2. Organize your plug-ins:
   - Drag plug-ins to the FX Favorites folder to add them to the **FX Favorites** menu.
   - Create submenus in the **FX Favorites** menu by clicking the FX Favorites folder and clicking the **Create New Folder** button (). After you have created a new folder, drag plug-ins to the folder to add them to the submenu in the **FX Favorites** menu.
   - Remove plug-ins or folders from the **FX Favorites** menu by selecting the plug-in or folder and clicking the **Delete** button (). Deleting a plug-in from the FX Favorites folder removes it from the **FX Favorites** menu but does not delete the plug-in from your system.
3. Close the Organize Favorites dialog. The new plug-ins and/or submenus appear in the **FX Favorites** menu.

> **Tip:** *You can also add plug-ins to the* **FX Favorites** *menu using the Plug-In Manager.*

Automatically adding and organizing plug-ins

You can automatically add all the plug-ins on your computer to your FX Favorites folder and organize them by the first word in the plug-in name (usually the company name). This replaces any menu structure you may have created with a rebuilt **FX Favorites** menu.

1. From the **FX Favorites** menu, choose **Recreate by Plug-In Name**. You are prompted to confirm the reorganization of the FX Favorites folder.
2. Click **Yes** to continue.

   Folders are created and the plug-ins are organized based on the first word in the names of the plug-ins.

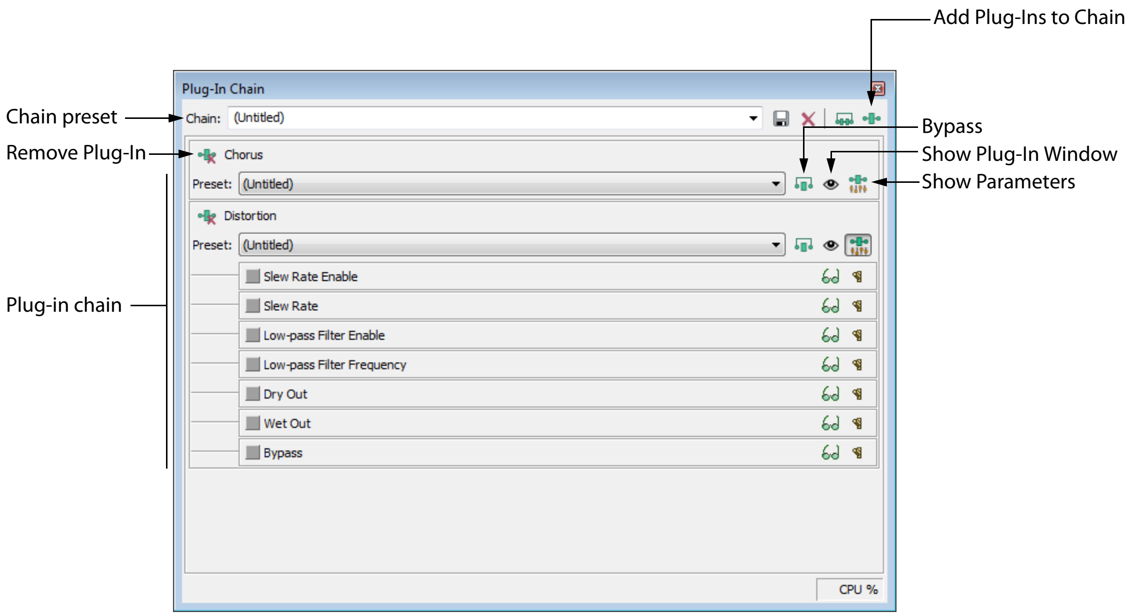
Using the Plug-In Chain

From the View menu, choose **Plug-In Chain** to open or close the Plug-In Chain for the current data window.

> **Tip:** *If you want to apply a single plug-in chain to a data window, use* **FX Favorites** > **Apply Plug-In Chain**. *For more information, see* *Applying a plug-in chain on page 214*.

The Plug-In Chain allows you to link up to 32 DirectX and VST plug-ins into a single processing chain. All of the plug-ins in the chain can be previewed simultaneously in real time (as long as your computer can process the preview information quickly enough).

The Plug-In Chain is modeless—you can navigate your data windows to change selections or apply effects to a different data window without closing the Plug-In Chain.



Creating a plug-in chain

1. Select the data you want to process.

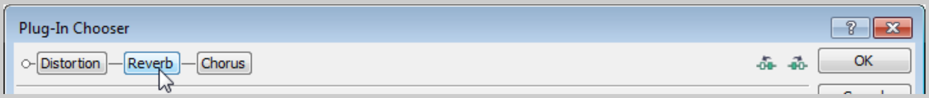
> **Tip:** *If you've used previous versions of Sound Forge Pro software, you'll notice that you can now click away from the Plug-In Chain at any time to adjust your selection or even to apply an effects chain to a different data window.*

2. From the View menu, choose **Plug-In Chain**.
3. Choose a preset from the **Chain** drop-down list to load an existing chain, or add the desired plug-ins to the chain. *For more information, see* *Adding a plug-in from the Plug-In Chooser on page 210*.
4. Choose which of the chain's plug-ins you want to apply:
   - Select an effect's **Bypass** button ( ) to prevent the audio signal from being sent through the plug-in. This is useful for isolating certain plug-ins without removing others from the chain.
   - Deselect the **Bypass** button for each plug-in you want to apply to your signal.
5. Adjust each plug-in's properties as desired. *For more information, see* *Setting a plug-in's properties on page 211*.

Adding, removing, or arranging plug-ins on a chain

Adding a plug-in from the Plug-In Chooser

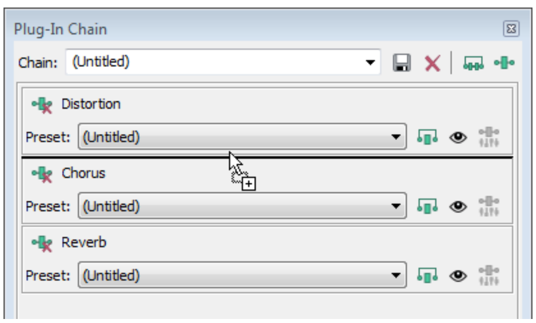
1. Click the **Add Plug-Ins to Chain** button ( ) in the Plug-In Chain window. The Plug-In Chooser window appears.
2. Select each plug-in you want to add, and then click the **Add** button, or browse to an effects package. The plug-ins appear at the top of the window in the order you added them.

> **Tips:**
>
> - *You can also double-click a plug-in in the Plug-In Chooser to add it to the chain.*
> - *To reorder the plug-ins within the chain, simply drag a plug-in button to a new location, or click the* **Shift Plug-In Left** *( ) or* **Shift Plug-In Right** *( ) buttons.*
>
> 
>

3. Once you have added all of the plug-ins and specified the plug-in chain order, click the **OK** button.

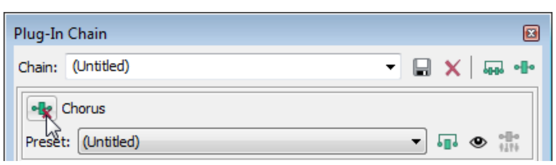
Adding a plug-in from the Plug-In Manager

You can add a plug-in to a chain quickly by dragging it from the Plug-In Manager to the Plug-In Chain window. Drop the plug-in at the position where you want to add it to the chain:



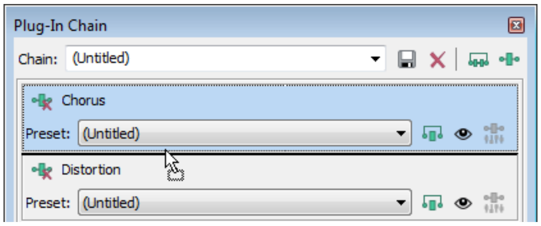
Removing a plug-in

Click the **Remove Plug-In** button ( ).



Editing a plug-in chain

1. From the View menu, choose **Plug-In Chain**.
2. Choose a preset from the **Chain** drop-down list.
3. To bypass a plug-in without removing it from the chain, select the effect's **Bypass** button ( ).
4. To reorder the plug-ins within the chain, simply drag a plug-in to a new location.



5. Click the **Show Plug-In Window** button ( ) to display the effect's parameters in a separate window. For more information about using specific plug-ins, click the **Plug-In Help** button (?) in the plug-in window.

Setting a plug-in's properties

Click the **Show Plug-In Window** button ( ) to display the effect's parameters in a separate window. For more information about using specific plug-ins, click the **Plug-In Help** button (?) in the plug-in window.

> **Tip:** *You can have multiple plug-in windows open in the Sound Forge Pro workspace. To quickly close all open plug-in windows, right-click a plug-in in the Plug-In Chain window and choose* **Close All Plug-In Windows** *from the shortcut menu.*

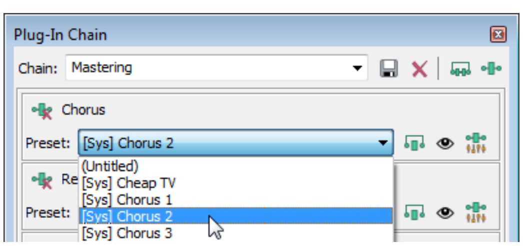
Loading an effects chain or plug-in preset

Loading a plug-in chain preset

Choose a setting from the **Chain** drop-down list. The preset chain is loaded using the saved settings for each DirectX and VST plug-in in the chain.

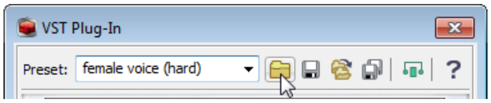
Loading a preset for an individual DirectX plug-in

Choose a setting from the **Preset** drop-down list. The plug-in settings stored in the preset are loaded.



Loading a preset for an individual VST plug-in

1. Click the **Show Plug-In Window** button ( ) to display the effect's parameters in a separate window.
2. Click the **Open VST Preset** button ( ).

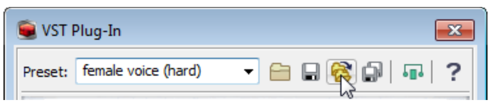
   

   The Open VST Preset dialog is displayed.
3. Browse to the .fxp file that you want to use.
4. Click the **Open** button.

   The current VST preset is replaced with the settings stored in the .fxp file.

Loading a bank of VST plug-in presets

1. Click the **Show Plug-In Window** button ( ) to display the effect's parameters in a separate window.
2. Click the **Open VST Bank** button ( ).

   The Open VST Preset Bank dialog is displayed.
3. Browse to the .fxb file that you want to use.
4. Click the **Open** button.

   All presets for the current VST plug-in are replaced with the settings stored in the .fxb file, and the first preset in the bank is loaded by default.

Previewing the effects chain

Effects from the Plug-In Chain are previewed in real time when you play back a file.

To hear the results of your effects chain without applying it to the sound file, click the **Play Normal** button ( ) in the data window's playbar.

You can select the **Bypass** button ( ) in the Plug-In Chain to bypass all effects in the chain, or select the **Bypass** button for an effect to bypass individual effects.

Applying the effects chain

Effects from the Plug-In Chain are applied when you save the file.

Saving the plug-in chain as a preset

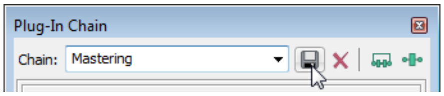
When you save a plug-in chain, the order of the effects in the chain and the settings for each plug-in are saved with the chain.

> **Note:** *Effect automation envelope points are not saved with presets. If you are using effect automation envelopes and save a preset during playback, the effect's settings at the playback cursor position are saved.*

1. Add plug-ins to the chain.
2. Adjust each plug-in's properties.
3. Type a name in the **Chain** box.

> **Note:** *You cannot modify default presets.*

4. Click the **Save Chain Preset** button ( ).

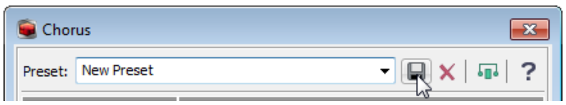


Saving the settings from an individual plug-in as a preset

> **Note:** *Effect automation envelope points are not saved with presets. If you are using effect automation envelopes and save a preset during playback, the effect's settings at the playback cursor position are saved.*

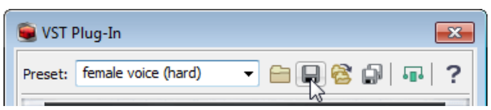
Saving a preset for an individual DirectX plug-in

1. Click the **Show Plug-In Window** button ( ) to display the effect's parameters in a separate window.
2. Type a name in the **Preset** box.
3. Click the **Save Preset** button ( ).



Saving a preset for an individual VST plug-in

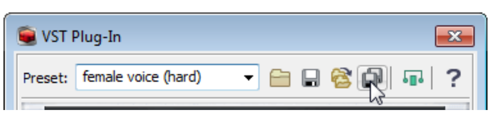
1. Click the **Show Plug-In Window** button ( ) to display the effect's parameters in a separate window.
2. Click the **Save VST Preset As** button ( ).



   The Save VST Preset dialog is displayed.
3. Browse to the folder where you want to save the .fxp file and type a name in the **File name** box.
4. Click the **Save** button. The current plug-in settings are saved in the .fxp file.

Saving a bank of VST plug-in presets

1. Click the **Show Plug-In Window** button ( ) to display the effect's parameters in a separate window.
2. Click the **Save VST Bank As** button ( ).



   The Save VST Preset Bank dialog is displayed.
3. Browse to the folder where you want to save the .fxb file and type a name in the **File name** box.
4. Click the **Save** button. All presets for the current plug-in are stored in the bank.

Plug-In Chain shortcuts

If you want to make your editing really fast, the Plug-In Chain can be controlled using keyboard shortcuts. *For more information, see Plug-In Chain shortcuts on page 355.*

Applying a plug-in chain

From the FX Favorites menu, choose **Apply Plug-In Chain** to apply a plug-in chain to a selection or data window.

> **Tip:** *If you want to work with each data window's active plug-in chain or use effect automation, choose* **View** > **Plug-In Chain**. *For more information, see Using the Plug-In Chain on page 209.*

Creating a plug-in chain

1. Select the data you want to process.

> **Tip:** *If you've used previous versions of Sound Forge software, you'll notice that you can now click away from the Apply Plug-In Chain at any time to adjust your selection.*

2. From the FX Favorites menu, choose **Apply Plug-In Chain**.
3. Choose a preset from the **Chain** drop-down list to load an existing chain, or add the desired plug-ins to the chain.
4. Choose which of the chain's plug-ins you want to apply:
   - Clear an effect's check box (Reverb) to prevent the audio signal from being sent through the plug-in. This is useful for isolating certain plug-ins without removing others from the chain.
   - Select the check box for each plug-in you want to apply to your signal.
5. If you want to choose a processing mode to determine how Sound Forge software handles the extra audio tail of plug-ins such as reverb or delay, right-click the processing dialog and choose a command from the shortcut menu:
   - Choose **Ignore Tail Data** to ignore the tail. The effect will end abruptly at the end of the selection.
   - Choose **Mix Tail Data** to mix the tail into the adjacent material. This is the most natural-sounding option.
   - Choose **Insert Tail Data** to insert the audio tail. All audio to the right of the tail will be moved over to accommodate the extra audio.
6. Adjust each plug-in's properties as desired.

Adding, removing, or arranging plug-ins on a chain

Adding a plug-in from the Plug-In Chooser

1. Click the **Add Plug-Ins to Chain** button ( ) in the Apply Plug-In Chain window. The Plug-In Chooser window appears.
2. Select each plug-in you want to add, and then click the **Add** button, or browse to an effects package. The plug-ins appear in the Apply Plug-In Chain window in the order you added them.

> **Tips:**
>
> - *You can also double-click a plug-in in the Plug-In Chooser to add it to the chain.*
> - *To reorder the plug-ins within the chain, simply drag a plug-in button to a new location, or click the* **Shift Plug-In Left** *( ) or* **Shift Plug-In Right** *( ) buttons.*

3. Once you have added all of the plug-ins and specified the plug-in chain order, click the **OK** button.

Removing a plug-in

Select a plug-in in the Apply Plug-In Chain window and click the **Remove Selected Plug-In** button ( ).

Editing a plug-in chain

1. From the FX Favorites menu, choose **Apply Plug-In Chain**.
2. Choose a preset from the **Chain** drop-down list.

3. To bypass a plug-in without removing it from the chain, clear an effect’s check box (Reverb).
4. To reorder the plug-ins within the chain, simply drag a plug-in button to a new location.
5. Click the plug-in’s button to select it, and use the bottom half of the dialog box to adjust the effect’s parameters. For more information about using specific plug-ins, click the **Plug-In Help** button (?).

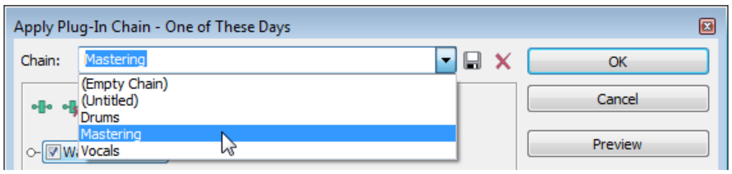
Setting a plug-in’s properties

Click the plug-in's button to select it, and use the bottom half of the dialog box to adjust the effect's parameters. For more information about using specific plug-ins, click the **Plug-In Help** button (?).

Loading an effects chain or plug-in preset

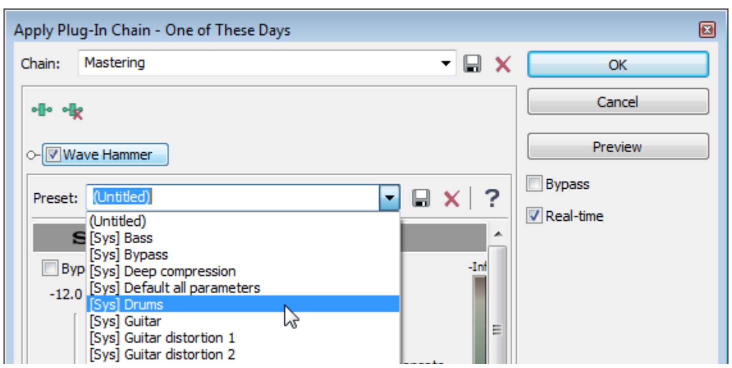
Loading a plug-in chain preset

Choose a setting from the **Chain** drop-down list. The preset chain is loaded using the saved settings for each DirectX and VST plug-in in the chain.



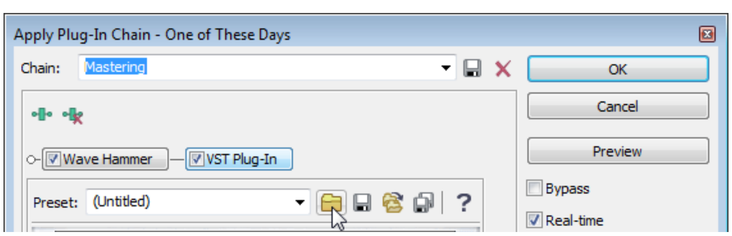
Loading a preset for an individual DirectX plug-in

Choose a setting from the **Preset** drop-down list. The plug-in settings stored in the preset are loaded.



Loading a preset for an individual VST plug-in

1. Click the VST effect’s button to display the effect’s parameters in the Apply Plug-In Chain window.
2. Click the **Open VST Preset** button ( ).



The Open VST Preset dialog is displayed.

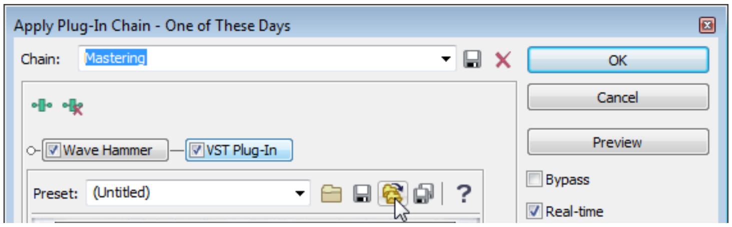
3. Browse to the .fxp file that you want to use.

4. Click the **Open** button.

   The current VST preset is replaced with the settings stored in the .fxp file.

Loading a bank of VST plug-in presets

1. Click the VST effect's button to display the effect's parameters in the Apply Plug-In Chain window.
2. Click the **Open VST Bank** button ( ).



   The Open VST Preset Bank dialog is displayed.

3. Browse to the .fxb file that you want to use.
4. Click the **Open** button.

   All presets for the current VST plug-in are replaced with the settings stored in the .fxb file, and the first preset in the bank is loaded by default.

Previewing the effects chain

To hear the results of your effects chain without applying it to the sound file, click the **Preview** button in the Apply Plug-In Chain window.

You can select the **Bypass** check box to bypass all effects in the chain, or clear an effect's check box (Reverb) to bypass individual effects.

You can use the buttons and shortcut menus in processing dialogs to set parameters for previewing and processing with plug-ins. Click the **More** button to display controls you can use to adjust the selection, wet and dry gain, and fade in/out settings.

For more information, see Using processing dialogs on page 218.

Applying the effects chain

To apply the effects chain to a data window, click the **OK** button in the Apply Plug-In Chain window.

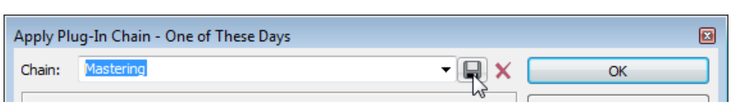
Saving the plug-in chain as a preset

When you save a plug-in chain, the order of the effects in the chain and the settings for each plug-in are saved with the chain.

1. Add plug-ins to the chain.
2. Adjust each plug-in's properties.
3. Type a name in the **Chain** box.

> **Note:** *You cannot modify default presets.*

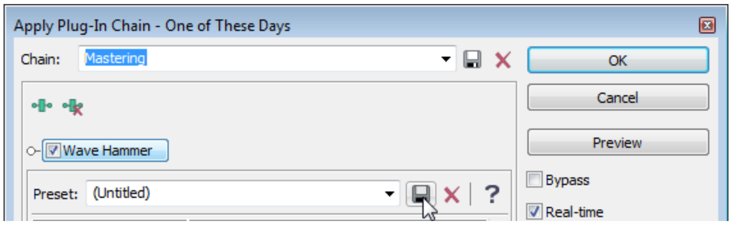
4. Click the **Save Chain Preset** button ( ).



Saving the settings from an individual plug-in as a preset

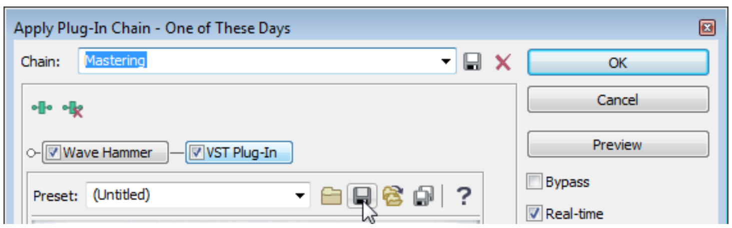
Saving a preset for an individual DirectX plug-in

1. Click the effect's button to display the effect's parameters in the Apply Plug-In Chain window.
2. Type a name in the **Preset** box.
3. Click the **Save Preset** button ( ).



Saving a preset for an individual VST plug-in

1. Click the VST effect's button to display the effect's parameters in the Apply Plug-In Chain window.
2. Click the **Save VST Preset As** button ( ).

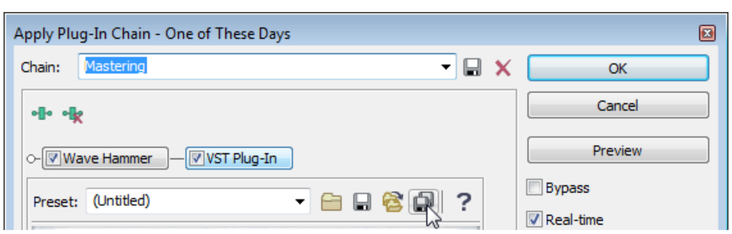


The Save VST Preset dialog is displayed.

3. Browse to the folder where you want to save the .fxp file and type a name in the **File name** box.
4. Click the Save button. The current plug-in settings are saved in the .fxp file.

Saving a bank of VST plug-in presets

1. Click the VST effect's button to display the effect's parameters in the Apply Plug-In Chain window.
2. Click the **Save VST Bank As** button ( ).



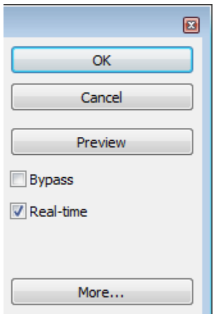
The Save VST Preset Bank dialog is displayed.

3. Browse to the folder where you want to save the .fxb file and type a name in the **File name** box.
4. Click the **Save** button. All presets for the current plug-in are stored in the bank.

Using processing dialogs

In previous versions of Sound Forge, processing dialogs were modal: after opening a processing dialog, you couldn't adjust the selection in the data window. You can move freely between processing dialogs and data windows to adjust your selection, fade-in and -out, and gain levels.

All processing dialogs share common controls along the right-hand side of the dialog. Use these controls to save and delete custom presets, preview your changes, and modify the selection.



Item	Description
OK	Closes the dialog and processes the dialog settings.
Cancel	Closes the dialog without making changes.
Preview	Click to begin previewing the processed sound file.
Bypass	Select this check box and click **Preview** to hear the unprocessed audio. This is a useful feature when comparing the affected and unprocessed signal.
Real-time	When this check box is selected, Sound Forge will try to preview the plug-in in real time. If your computer can not keep up, clear the check box. When the check box is cleared, the maximum preview length is determined by the **Limit non-realtime previews to** setting on the Previews tab of the Preferences dialog. *For more information, see Previews tab on page 339.*
More	Click to display additional controls at the bottom of the dialog that you can use to change the selection you want to process or adjust dry and wet gain and fades.

Previewing audio

Click the **Preview** button to hear the effect of the current dialog settings on the selected audio.

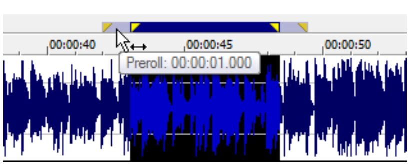
During playback, you can select the **Bypass** check box to hear the unprocessed audio. This is a useful feature when comparing the affected and unprocessed signal.

Previewing with pre- or post-roll

If you want to listen to the unprocessed audio before or after the current selection, right-click the processing dialog and choose **Pre-Roll** or **Post-Roll** from the shortcut menu. A check box is displayed next to the commands when they're selected.

Pre- and post-roll allow you to hear the transition from unprocessed to processed data.

When the **Pre-Roll** and **Post-Roll** buttons are selected, the pre/post roll regions are displayed next to the loop region in the data window:



You can use the Previews tab in the Preferences dialog to specify how many seconds of unprocessed audio will be played before and after the processed selection. *For more information, see Previews tab on page 339.*

Choosing a processing mode for tail data

If you want to specify how Sound Forge handles the extra audio tail of plug-ins such as reverb or delay, right-click the processing dialog and choose a command from the shortcut menu:

- Choose **Ignore Tail Data** to ignore the tail. The effect will end abruptly at the end of the selection.
- Choose **Mix Tail Data** to mix the tail into the adjacent material. This is the most natural-sounding option.
- Choose **Insert Tail Data** to insert the audio tail. All audio to the right of the tail will be moved over to accommodate the extra audio.

Creating a preset

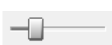
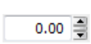
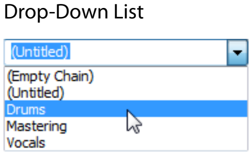
1. Choose a command from the Process, Effects, or FX Favorites menu.
2. Adjust the dialog controls to create the desired effect.
3. Type a name in the **Preset** box.
4. Click the **Save Preset** button ( ).

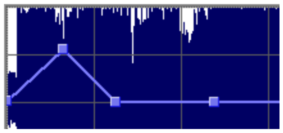
Deleting a preset

1. Choose a command from the Process, Effects, or FX Favorites menu.
2. Choose a preset from the **Preset** drop-down list.
3. Click the **Delete Preset** button ( ).

Note: *The* **Delete Preset** *button is available only for custom presets.*

Using Sound Forge controls

<table>
<tr><th>Control</th><th>Description</th></tr>
<tr><td>Slider/Fader
</td><td>Drag the handle to change the setting.
Tips:
• Hold the left and right mouse buttons (or hold the Control key) to fine-tune a control's value.
• You can use the Up, Down, Left and Right arrow keys to change the value in small increments or the Page Up and Page Down keys to change the value in larger increments. The Home and End keys change the parameter value to its maximum or minimum.
• Double-click a handle to return to the default value (usually 0%, 50%, or 100%). Left-clicking on the hash marks in a fader also changes the value by very small increments.</td></tr>
<tr><td>Spin Control

</td><td>Use the up/down buttons or type a value in the edit box to change the setting.
Tips:
• Click the button between the up/down buttons and drag the mouse to change the setting in large increments.
• Hold down both mouse buttons (or hold down the Control key) to fine-tune a control's value.
• You can use the Up/Down arrow and Page Up/Page Down keys to alter the value.</td></tr>
<tr><td>Drop-Down List

</td><td>Click the drop-down list and choose an item. If you have to scroll through a large list, click the scroll buttons or use the arrow keys.
Tip: If you scroll down the drop-down list for a preset, you can see all of the parameters change for each preset. This is useful for getting a feel for which parameters are used to create different effects.</td></tr>
</table>

Control	Description
Button	Click a button or press the spacebar while it is selected.
Radio Button	Radio buttons always come in groups of two or more where you can select only one option. When you select a radio button, the previously selected button is turned off.
Check Box	Click a check box to select it. You can click a selected check box to clear it.
Envelope Graph	An envelope allows you to change a sound over time.  • Drag the small boxes (envelope points) up or down. • To create a new envelope point, click the envelope. • To delete an envelope point, click it with the right mouse button, or double-click it with the left mouse button. • To move all envelope points, press Ctrl+A and drag when the envelope has focus (the cursor will be displayed as a). You can create up to 16 envelope points. **Note:** *Click the Reset Envelope button to reset the graph.*

Changing the data window selection

You can adjust the selection to be processed by changing the selection in the data window or using the controls at the bottom of the processing dialog.

> **Tip:** *Right-click the processing dialog and choose a command from the shortcut menu to change the selection quickly. Choose* **Select Original** *to restore the selection that existed when you first opened the processing dialog. Choose* **Select All** *to select all data in the current window.*

1. Click the **More** button in the processing dialog. Selection and gain controls are displayed at the bottom of the dialog.
2. Type a value in the **Start** box (or use the spinner) if you want to change the beginning of the selection.

   Click the **Lock Start** button ( ) if you want to preserve the selection start when adjusting the **Length** or **End** settings.

   > **Note:** *When using the Event tool, the* **Start** *box is automatically locked.*

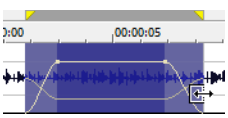
   Changing this setting has the same effect as dragging the beginning of the selection in the data window:

3. Type a value in the **End** box (or use the spinner) if you want to change the end of the selection.

   > **Note:** *When using the Event tool, the* **End** *box is not available.*

   Changing this setting has the same effect as dragging the end of the selection in the data window:

4. Type a value in the **Length** box (or use the spinner) if you want to specify the length of the selection. The beginning of the selection remains fixed.

   Click the **Lock Selection Length** button ( ) if you want to preserve the selection length when adjusting the **Start** or **End** settings.

   > **Note:** *When using the Event tool, the* **Length** *box is automatically locked.*

5. Type a value in the **Channels** box to change the current channel selection. You can separate individual channels with commas or use hyphens to indicate channel ranges.

   Click the **Lock Channels** button ( ) if you want to lock the channel selection.

   For example, type 1, 3, 6-8 to select channels 1, 3, 6, 7, and 8.

   Changing this setting has the same effect as holding Ctrl while clicking channels in the data window.

   > **Note:** *When using the Event tool, the* **Channels** *box is automatically locked.*

6. When using the Event tool, you can click the **Go to Previous Event** ( ) or **Go to Next Event** ( ) button at the bottom of the processing dialog to navigate events when multiple events are selected.

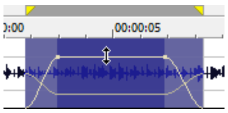
Changing the wet/dry mix and fade in/out

You can adjust the wet/dry mix and fade in/out by changing the envelopes in the data window or using the controls at the bottom of the processing dialog.

If you want to change the default fade values, you can use the Editing tab in the Preferences dialog. *For more information, see Editing tab on page 335.*

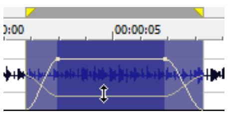
1. Click the **More** button in the processing dialog. Selection and gain controls are displayed at the bottom of the dialog.
2. Type a value in the **Wet Gain** box (or use the spinner) to set the level of the processed signal that will be mixed into the output.

   Changing this setting has the same effect as dragging the sustain portion of the wet gain envelope in the data window:

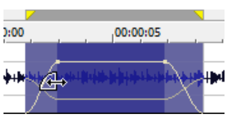
3. Type a value in the **Dry Gain** box (or use the spinner) to set the level of the unprocessed signal that will be mixed into the output.

   Changing this setting has the same effect as dragging the sustain portion of the dry gain envelope in the data window:

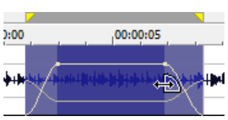
4. Type a value in the **Fade In** box (or use the spinner) to set the length of the fade in between the unprocessed and processed signal.

   Changing this setting has the same effect as dragging the attack portion of the envelope in the data window:

5. Click the **Fade Curves** button () and choose a curve type from the menu to set the speed of the fade in.
6. Type a value in the **Fade Out** box (or use the spinner) to set the length of the fade out between the processed and unprocessed signal.

   Changing this setting has the same effect as dragging the release portion of the envelope in the data window:

7. Click the **Fade Curves** button () and choose a curve type from the menu to set the speed of the fade out.

Tips:

- *To reset the fade in and out to the default values, right-click the processing dialog and choose* **Reset Fade Values** *from the shortcut menu.*
- *To save default fade in and out values, adjust the* **Fade In** *and* **Fade Out** *controls, and then right-click the processing dialog and choose* **Save Fade Values** *from the shortcut menu.*

Learning more about a specific effect

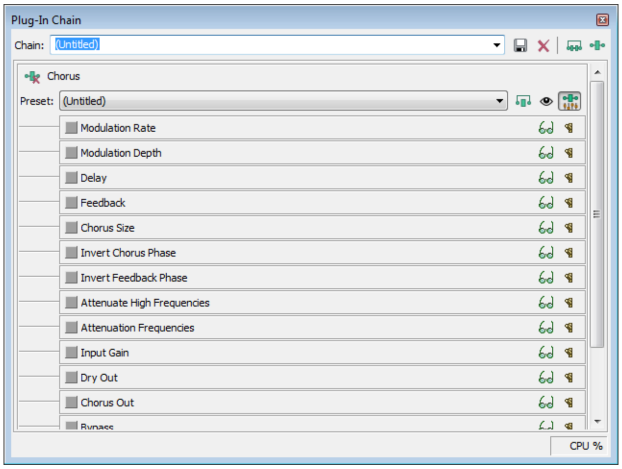
For more information about each effect, see Sound Forge effects on page 226.

Learning more about a specific process

For more information about each process, see Sound Forge processes on page 185.

Automating effect parameters

When you add an effect that supports automation to the Plug-In Chain, you can click the **Show Parameters** button ( ) to display the effect's automatable parameters. You can use these controls to show/hide and enable/bypass automation envelopes.



Plug-in parameters can be edited using the plug-in's controls or the automation envelope in the data window.

Note: *Choosing a new effect chain preset will clear the current effect automation settings.*

Adding a volume or panning envelope

Important: *Panning envelopes will have no effect on mono source data. Convert mono sound data to stereo before adding a panning envelope.*

1. Click within a data window to give it focus.
2. From the Insert menu, choose **Volume Envelope** or **Pan Envelope** (you can also press V to add a volume envelope, or press P to add a panning envelope).

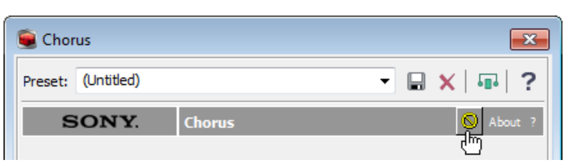
   An envelope is added to the data window, and the Volume or Pan plug-in is added to the Plug-In Chain. If the Plug-In Chain is not visible, it will be opened automatically.

Note: *If a data window already has a volume or panning envelope, pressing V or P will hide the envelope.*

3. Adjust volume or panning with the envelope in the data window.

Adjusting effect parameters with plug-in controls

1. Click to position the cursor in the data window and use the controls in the Plug-In Chain to adjust effect parameters at the cursor position. The envelope is updated as you adjust the plug-in's controls.
2. If you've enabled the **Bypass** parameter for a plug-in, you can click the **Bypass** button in the plug-in's banner to toggle the Bypass envelope at the cursor position.



3. You can also adjust the envelope to adjust parameters. *For more information about working with envelopes, see Adjusting envelopes on page 225.*

> **Note:** *When you automate an effect's frequency parameter — such as the modulation frequency parameter in the amplitude modulation effect — you may notice that the frequency changes are more apparent when moving through the lower frequencies. This is because frequency scales plug-ins use a logarithmic scale, but effect automation uses linear interpolation.*
>
> *To make the automated frequency changes sound more natural, change the fade curve types to change the interpolation rates between envelope points. For high-to-low frequency sweeps, use a fast fade curve; for low-to-high frequency sweeps, use a slow curve. For more information about changing fade curves, see Setting fade properties on page 225.*

Adjusting effect parameters with envelopes

An envelope is displayed in the data window for each effect parameter that you've chosen to automate. Envelope points represent plug-in parameter settings at a specific point in time.

You can add points, adjust their positions, and change the fade curves between points to modify effect parameters and the transitions between them. *For more information about working with envelopes, see Adjusting envelopes on page 225.*

> **Note:** *When you automate an effect's frequency parameter — such as the modulation frequency parameter in the amplitude modulation effect — you may notice that the frequency changes are more apparent when moving through the lower frequencies. This is because frequency scales plug-ins use a logarithmic scale, but effect automation uses linear interpolation.*
>
> *To make the automated frequency changes sound more natural, change the fade curve types to change the interpolation rates between envelope points. For high-to-low frequency sweeps, use a fast fade curve; for low-to-high frequency sweeps, use a slow curve. For more information about changing fade curves, see Setting fade properties on page 225.*

Previewing effect automation

Effects from the Plug-In Chain are previewed in real time when you play back a file.

To hear the results of your effect automation chain without applying it to the sound file, click the **Play Normal** button ( ) in the data window's playbar.

You can select the **Bypass** button ( ) in the Plug-In Chain to bypass all effects in the chain, or select the **Bypass** button for an effect to bypass individual effects.

Applying effect automation

Effects from the Plug-In Chain are applied when you save the file.

Showing or hiding effect automation envelopes

1. In the Plug-In Chain, click the **Show Parameters** button ( ) to display the effect's automatable parameters.
2. Select the **Show Envelope** button ( ) to display a parameter's envelope, or deselect the button to hide the envelope. Hiding an envelope simply removes the line from the data window while it retains the playback properties.

Enabling or bypassing effect automation envelopes

1. In the Plug-In Chain, click the **Show Parameters** button ( ) to display the effect's automatable parameters.
2. Select the **Enable Envelope** button ( ) to apply an automation envelope to your audio signal, or deselect the button to ignore the envelope.

   When the button is not selected, an effect automation envelope is ignored and the effect's initial state is used for the duration of the data window.

   Bypassed envelopes are drawn with a dashed line in the data window.

Adjusting envelopes

When the Envelope tool ( ) on the main workspace is selected, you can add, remove, select or adjust envelope points on effect automation envelopes.

The Edit tool ( ) allows you to add, remove, or adjust envelope points, but you cannot select envelope points with the Edit tool.

By default, a new envelope will contain a single envelope point. If you want to adjust the overall level of an envelope, drag the envelope up or down. A floating ToolTip will show you the envelope's current setting.

If an envelope has multiple points, you can drag each point, or you can drag envelope segments up or down.

Tips:

- *Hold Ctrl while dragging an envelope point or segment to adjust the value in fine increments without changing the envelope points' horizontal positions.*
- *Hold Ctrl+Alt while dragging an envelope point or segment to adjust the value in normal increments without changing the envelope points' horizontal positions.*
- *Hold Alt while dragging an envelope point to move the point's horizontal position without changing its value.*

  With the Envelope tool, you can drag horizontally to select multiple envelope points in the selected data window.

Adding envelope points

To create more complex envelopes, you will need to add points. To add an envelope point, double-click the envelope. You can then drag and position the point as necessary.

To delete a point, right-click it and choose **Delete** from the shortcut menu.

Flipping an envelope

You can flip an envelope to invert the envelope around its center. Volume, panning, and effect automation envelopes can be flipped.

Flipping all points

1. Right-click an envelope or a point. A shortcut menu is displayed.
2. Choose **Flip All Points** from the shortcut menu.

Flipping selected points

1. Create a time selection with the **Envelope** tool ( ) to select the points you want to flip.
2. Right-click an envelope in the time selection. A shortcut menu is displayed.
3. Choose **Flip Selected Points** from the shortcut menu.

Setting fade properties

You can adjust the fade curve for each envelope segment individually. To change the fade curve, right-click an envelope segment and choose a fade command (such as **Linear Fade** or **Fast Fade**, for example) from the shortcut menu.

Cutting, copying, and pasting envelope points

1. Select the **Envelope** tool ( ).
2. Click within a data window to select it.
3. Drag horizontally in a data window to select envelope points.
4. From the **Edit** menu, choose **Cut** or **Copy**.

5. Click to position the cursor where you want to paste envelope points.

Tip: *Click within a different data window if you want to paste envelope points across data windows.*

6. From the **Edit** menu, choose **Paste**.

Copying an envelope to another data window

1. Select the **Envelope** tool ( ).
2. Click within a data window to select it.
3. From the **Edit** menu, choose **Select All**.
4. From the **Edit** menu, choose **Copy**.
5. Click within a data window to select it.

Tip: *You can paste envelope points to a different envelope type by selecting the envelope where you want to paste.*

6. Click **Go to Start** ( ) if you want the envelope to appear exactly as it was in the original data window, or click to position the cursor where you want the envelope to start.
7. From the **Edit** menu, choose **Paste**.

Using the Preset Manager

After you have created custom presets for effects or effect chains, you can use the Sound Forge Preset Manager to back up, transfer, or delete custom presets from any of the effects, processes, tools and plug-ins installed in the software. The Preset Manager also functions as a standalone application, meaning that you can use the Preset Manager outside of Sound Forge software to manage ACID and Vegas presets as well.

Note: *If you purchased the boxed version of Sound Forge Pro software, the Preset Manager is included on the Sound Forge Pro application disc. If you purchased the downloadable version, you can get the Preset Manager on our Web site at http://www.sonycreativesoftware.com/download/utilities.*

To display the **Preset Manager**, choose **Preset Manager** from the **Tools** menu. In the Preset Manager, choose **Contents and Index** from the **Help** menu for instructions on how to manage your presets.

Sound Forge effects

The remainder of this chapter describes the functions located in the **Effects** menu.

Acoustic Mirror

The Acoustic Mirror effect is a powerful digital signal processing tool that allows you to add environmental coloration to your existing recordings. *For more information, see What are the Acoustic Mirror effects? on page 233.*

Amplitude Modulation

Use the Amplitude Modulation effect to apply a sinusoidal or square-shaped periodic gain to the input signal. The frequency of the gain waveform can be specified to create effects varying from a slow tremolo to unusual sound distortions.

For information about using the Amplitude Modulation effect, click the **Help** button ( ) in the Sony Amplitude Modulation dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Chorus

From the **Effects** menu, choose **Chorus** to add a pitch-modulated and delayed version of the input signal to the unprocessed input signal. The effect simulates the variances in pitch and timing that occur naturally when two or more people try to play or sing the same thing at the same time.

For information about using the Chorus effect, click the **Help** button (?) in the Sony Chorus dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Delay/Echo

From the **Effects** menu, choose **Delay/Echo**, and then choose a command from the submenu to apply an echo effect to a selection.

For information about using the Multi-Tap Delay and Simple Delay effects, click the **Help** button (?) in the plug-in dialogs or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Distortion

From the **Effects** menu, choose **Distortion** to tailor the gain at all input levels of a signal. You can create effects ranging from dramatic to subtle distortion, simple compression, expansion, and noise gates.

For information about using the Distortion effect, click the **Help** button (?) in the Sony Distortion dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Dynamics

From the **Effects** menu, choose **Dynamics**, and then choose a command from the submenu to modify the dynamic range of a selection.

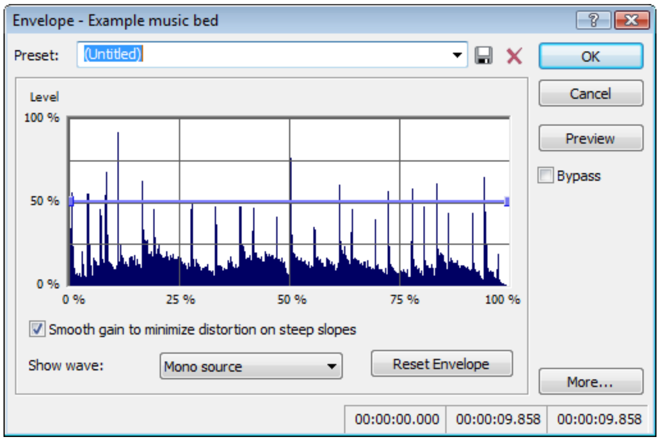
For more information about using the Graphic Dynamics and Multi-Band Dynamics effects, click the **Help** button (?) in the plug-in dialogs or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Envelope

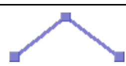
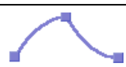
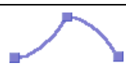
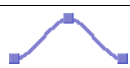
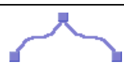
From the **Effects** menu, choose **Envelope** to apply an envelope to vary the amplitude of a waveform over time. Unlike the Graphic Fade command, which simply fades a waveform by a specific amount over time, the gain at each point is dynamically calculated to achieve the exact specified envelope.

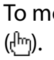
Applying an amplitude envelope

1. From the **Effects** menu, choose **Envelope**. The Envelope dialog appears.



2. Adjust the envelope to achieve the desired sound:
   - Drag the small envelope points up or down.
   - To create a new envelope point, double-click the envelope.
   - To delete an envelope point, right-click it and choose **Delete** from the shortcut menu.
   - To change the fade curve between two points, right-click an envelope segment and choose a command from the shortcut menu:

Fade Type	Fade In/Out Envelope
Linear Fade	
Fast Fade	
Slow Fade	
Smooth Fade	
Sharp Fade	
Hold	

   - To move all envelope points, press Ctrl+A and drag when the envelope has focus. The cursor will be displayed as a hand ().

Note: *You can create up to 16 envelope points. Click the* **Reset Envelope** *button to reset the graph to a simple ADSR (Attack, Decay, Sustain, Release) curve.*

3. Select the **Smooth gain to minimize distortion on steep slopes** check box to prevent the gain from changing too quickly, which might result in unwanted distortion. Also, when this option is on, the gain will always begin at 0%.
4. Click the OK button.

Displaying the waveform

Choose a command from the **Show wave** drop-down list if you want to display the waveform in the envelope graph.

If you're working with a multichannel file, you can choose to view individual channels or the mixed waveform.

> **Note:** *The waveform is unavailable when the selection is greater than 300,000 samples.*

Flange/Wah-Wah

From the **Effects** menu, choose **Flange/Wah-Wah** to apply flanging, phasing, and wah-wah effects to a sound.

For information about using the Flange/Wah-Wah effect, click the **Help** button ( ) in the Sony Flange/Wah-Wah dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Gapper/Snipper

From the **Effects** menu, choose **Gapper/Snipper** to cut chunks from the sound file or insert silence in the sound file periodically at a set frequency.

For information about using the Gapper/Snipper effect, click the **Help** button ( ) in the Sony Gapper/Snipper dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Noise Gate

From the **Effects** menu, choose **Noise Gate** to remove signals below a specified threshold. This effect is used to remove noise from silent breaks in a sound file.

For information about using the Noise Gate effect, click the **Help** button ( ) in the Sony Noise Gate dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

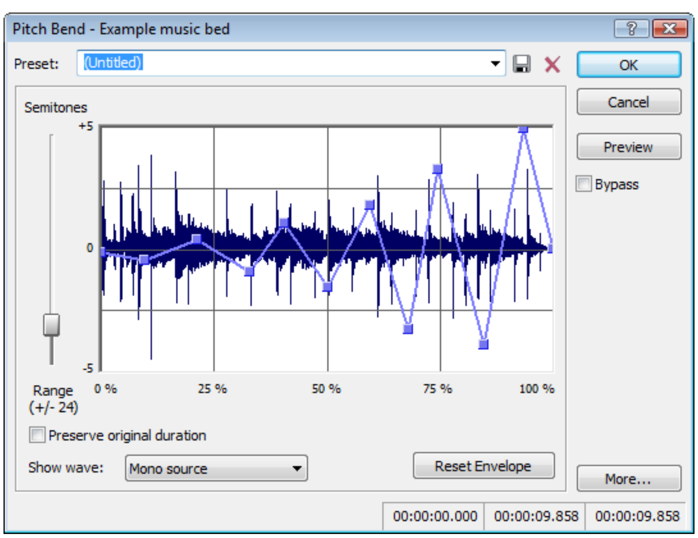
Pitch

From the **Effects** menu, choose **Pitch**, and then choose a command from the submenu to change the pitch of a selection.

Bend

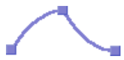
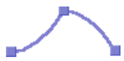
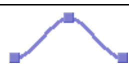
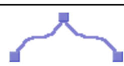
The **Bend** command allows you to draw an envelope that increases or decreases the pitch of a sound file over time.

1. From the **Effects** menu, choose **Pitch**, and then choose **Bend** from the submenu. The Pitch Bend dialog appears.



2. Drag the **Range** slider to determine the maximum and minimum pitch change in semitones (half-steps). Example: A range of twelve half-steps (one octave) allows an increase or decrease of the pitch by an octave.

3. Adjust the envelope to achieve the desired sound:
   - Drag the small envelope points up or down.
   - To create a new envelope point, double-click the envelope.
   - To delete an envelope point, right-click it and choose **Delete** from the shortcut menu.
   - To change the fade curve between two points, right-click an envelope segment and choose a command from the shortcut menu:

Fade Type	Fade In/Out Envelope
Linear Fade	
Fast Fade	
Slow Fade	
Smooth Fade	
Sharp Fade	
Hold	

   - To move all envelope points, press Ctrl+A and drag when the envelope has focus. The cursor will be displayed as a hand ( ).

Note: *You can create up to 16 envelope points. Click the* **Reset Envelope** *button to remove all but the outer two envelope points.*

4. Select the **Preserve original duration** check box if you do not want pitch bending to change the size of the sound file. This setting works best when performing small pitch corrections (up to +/- 2 semitones).
5. Click the **OK** button.

Displaying the waveform

Choose a command from the Show wave drop-down list if you want to display the waveform in the envelope graph.

If you're working with a multichannel file, you can choose to view individual channels or the mixed waveform.

Note: *The waveform is unavailable when the selection is greater than 300,000 samples.*

Shift

The Shift effect allows you to change the pitch of a sound with or without preserving the duration of the selection.

For information about using the Pitch Shift effect, click the **Help** button ( ) in the Sony Pitch Shift dialog or refer to Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Resonant Filter

From the **Effects** menu, choose **Resonant Filter** to restrict the range of a sound using low-pass, band-pass, or high-pass filtering, and then boost and add oscillation to the resonant frequency.

For information about using the Resonant Filter plug-in, click the **Help** button ( ) in the Sony Resonant Filter dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Reverb

From the **Effects** menu, choose **Reverb** to simulate various acoustic spaces. Reverb consists of early reflections, which are the first reflections that arrive back to your ear, and the reverb itself.

For information about using the Reverb plug-in, click the **Help** button (?) in the Sony Reverb dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Vibrato

From the **Effects** menu, choose **Vibrato** to apply periodic pitch modulation to a selection.

For information about using the Vibrato effect, click the **Help** button (?) in the Sony Vibrato dialog or refer to the Sound Forge online help (from the **Help** menu, choose **Contents and Index**).

Wave Hammer

Use Wave Hammer to boost the level of an audio signal for mastering or mixing. *For more information, see What is the Wave Hammer plug-in? on page 245*.

Chapter 15

Using Acoustic Mirror and Wave Hammer

This chapter is designed to familiarize you with the Sound Forge® Pro Acoustic Mirror™ and Wave Hammer™ effects.

The Acoustic Mirror effect is a powerful digital signal processing tool that allows you to add environmental coloration to your existing recordings.

The Wave Hammer effect is an audio mastering tool that features a classic compressor and volume maximizer.

What are the Acoustic Mirror effects?

The Acoustic Mirror effects represents an advance in reverb technology in that it incorporates the acoustical responses of a given environment or venue into your audio files. You may never play Carnegie Hall, but that does not mean that your recordings can't sound like it. Taking this concept even further, this effect allows you to simulate the signal response of vintage musical equipment. Imagine the money you'll save not having to buy those paired U-47s.

The acoustic signature

Acoustic Mirror effects use the environment's acoustic signature, or impulse response. These acoustic signatures are saved as impulse files and given the extension .wav or .sfi. An extensive library of high-quality impulse files are included. In addition, you can collect your own acoustic signatures and create custom impulse files.

Adding an acoustic signature to an audio file

1. Open and play the Saxriff.pca file.

> **Note:** *This file is located in the same folder as the application.*

2. From the **Effects** menu, choose **Acoustic Mirror**. The Acoustic Mirror dialog is displayed.

> **Note:** *You must have an active file in the Sound Forge workspace to start the Acoustic Mirror tool.*

3. Click the **Browse** button located next to the **Impulse** field and locate the **Acoustic Mirror Impulse Files** folder on the Sound Forge application disc.
4. Double-click the folder. Several impulse subfolders display.
5. Double-click the **Large venues** folder. Several impulse files display.
6. Double-click **Stadium, Camp Randall 50 yrd line.sfi**. This impulse file's acoustic signature is added to the Saxriff.pca file and you are returned to the Acoustic Mirror dialog.
7. Click **Preview**. The processed file plays and the sax riff is virtually placed in a football stadium-sized venue.
8. Select or clear the **Bypass** check box to toggle between the processed and unprocessed audio.

Adjusting the acoustic signature

After you add an acoustic signature to a file, you can use the controls of the Acoustic Mirror dialog to precisely configure the reverb effect. More importantly, you can preview configuration changes as quickly as you make them.

1. Open a file and display the Acoustic Mirror dialog.
2. Verify that the **Real-time** check box is selected.
3. From the **Impulse** drop-down list, choose the desired impulse file and click **Preview**. The processed audio file is played. Notice that all dialog controls are set to their default values.

4. Drag the **Dry Out** fader up. Notice the audible change in output as the balance between the **Wet Out** and **Dry Out** values changes.

Tip: *If you are experiencing difficulty previewing processing in real-time, decrease the* **Quality/speed** *value.*

5. Drag the **Response delay** slider to the right. Notice the audible change in the reverb's delay.

The Acoustic Mirror dialog contains four tabs: **General**, **Envelope**, **Summary**, and **Recover**. Each tab contains controls that allow you to precisely configure the effect as well as recover custom impulses. Notice that the Acoustic Mirror dialog contains the preset and preview controls found in all of the Sound Forge process and effect dialogs. *For more information, see Applying presets on page 181 and Previewing processed audio on page 183.*

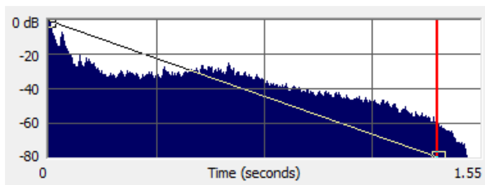
General tab controls

The following sections describe all controls located in the **General** tab.

Control	Description
Impulse	The **Impulse** drop-down list allows you to specify an impulse file from a list of those previously used. Clicking **Browse** displays the Open Impulse File dialog and allows you to locate an impulse file from your local system or network.
Response width	You can use the **Response width** slider to create some simple stereo expansion and stereo collapsing effects. This control's default setting of **50** represents normal stereo operation and is recommended to maintain the stereo field of the impulse response. A higher setting expands the stereo field, but may result in an unnatural sounding effect. Lowering this setting narrows the stereo field. A setting of **0** is essentially mono.
Response delay	The **Response delay** slider controls the time, in milliseconds, that elapses between the dry signal and the processed output. This control can be used to create interesting effects and add new dimensions to an acoustic signature. Configuring this control with a positive value results in the processed output following the dry output. A negative value results in the processed output preceding the dry signal, or a pre-delay.
Pan (left to right)	The **Pan** slider controls the balance between the left and right channels in stereo files. The default value is **0** and indicates a typical center position.
Dry Out	The **Dry Out** fader controls the amount of unprocessed signal mixed into the output.
Wet Out	The **Wet Out** fader controls the amount of processed signal mixed into the output.
Apply envelope and limit decay to (seconds)	When you select this check box, the length of the impulse is limited to the time specified in the adjacent box. Limiting the length of an impulse file shortens the decay of the reverberation and decreases the amount of processing required. In addition, selecting this check box results in the impulse fading according to the **Envelope Graph** configured on the dialog's **Envelope** tab. *For more information, see Envelope Graph on page 235.*
Low-shelf start frequency/ High-shelf start frequency	Acoustic Mirror high- and low-shelving filters to allow you to tailor the frequency response of the impulse. Notice that you can adjust the cutoff frequency and boost/ attenuation of each filter independently.
Convert mono to stereo	Selecting the **Convert mono to stereo** check box converts a mono signal to stereo output. If the impulse file is in stereo, selecting this check box imparts a pseudo-stereo effect on the mono input.
Quality/speed	The **Quality/speed** slider allows you to strike a balance between the quality and speed of the audio processing. Lowering this value immediately affects the frequency response of the impulse. The processed signal sounds dull and high frequencies sound unnatural. At very low values, the length of the impulse is shortened. When this control is set to a high value, the audio quality is excellent, but the processing takes longer. If you are experiencing difficulty previewing processing in real-time, decrease the **Quality/speed** value. However, you must return this value to **5** prior to actually processing the file to output the highest possible quality.

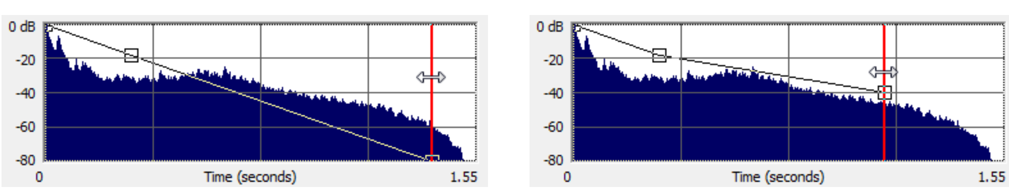
Envelope tab controls

The following sections describe all controls located on the **Envelope** tab.

Control	Description
Impulse	This control is identical to the **Impulse** drop-down list on the **General** tab. This allows you to view the envelope graphs for the specified impulse file. *For more information, see Impulse on page 234.*
Dry Out	This control is identical to the **Dry Out** fader on the **General** tab. *For more information, see Dry Out on page 234.*
Wet Out	This control is identical to the **Wet Out** fader on the **General** tab. *For more information, see Wet Out on page 234.*
Apply envelope and limit decay	This control is identical to the **Apply envelope and limit decay** check box on the **General** tab. *For more information, see Apply envelope and limit decay to (seconds) on page 234.*
Envelope Graph	Selecting the **Apply envelope and limit decay** check box turns on the **Envelope Graph**. You can use the envelope graph to decrease the length of the specified impulse file, which consequently decreases the reverberation decay time and processing time. You can also use the envelope graph to apply fades to the specified impulse file.  The horizontal axis of the graph represents the time of the impulse file and the vertical axis represents peak amplitude in dB. Specifying an impulse file from the **Impulse** drop-down list automatically displays its envelope in the graph. **Note:** *If the impulse file is greater than 6 seconds in length, it does not display in the envelope graph.*
Envelope points	Envelope points are used in the envelope graph to specify a fade curve. The fade amount can vary from 0% to 100%. You can create, delete, and arrange envelope points just as you can in all of the Sound Forge envelope graphs. *For more information, see Envelope graphs on page 49.* **Note:** *The fade value at any point in a curve does not use the same vertical logarithmic (dB) scale used for displaying the impulse file.*
Reset	Clicking this button resets the envelope points to 100%, indicating no fade.
Package Impulse into Preset	Clicking this button creates a link between the current preset and the selected impulse file, along with encoding the impulse information. You can use the Preset Manager to share presets and the accompanying impulse files between computers without losing information. *For more information, see Using the Preset Manager on page 226.*

Adjusting the impulse length

Drag the vertical **Envelope Endpoint** line to the desired location. The **Envelope Endpoint** is repositioned and the length of the impulse is adjusted.



Repositioning the envelope endpoint line

Summary tab controls

The **Summary** tab provides information about the impulse file. The following section describes all controls located on the **Summary** tab.

Control	Description
Impulse	This control is identical to the **Impulse** drop-down list on the **General** tab. *For more information, see Impulse on page 234.*
Dry Out	This control is identical to the **Dry Out** fader on the **General** tab. *For more information, see Dry Out on page 234.*
Wet Out	This control is identical to the **Wet Out** fader on the **General** tab. *For more information, see Wet Out on page 234.*
Quality/speed	This control is identical to the **Quality/speed** check box on the **General** tab. *For more information, see Quality/speed on page 234.*

Recover tab controls

The **Recover** tab is used in creating your own impulse files. *For more information, see Creating impulse files on page 238.* The following section describes all controls located on the **Recover** tab.

Control	Description
Recorded file	The **Recorded file** box allows you to select the file containing the test tone recorded in the field. You can enter the path directly into the box or click **Browse** to locate and select a file.
Test file used	The **Test file used** box allows you to select the file that was used as a test tone. You can enter the path directly into the box or click **Browse** to locate and select a file. **Note:** *You should use one of the test files included in the* **Acoustic Mirror Impulse Files\Test Tones** *folder on the Sound Forge application disc.*
Impulse output	The **Impulse output** box allows you to specify where the recovered impulse response file is saved. You can enter the path directly into the box or click **Browse** to locate and select a folder.
Remove very low frequencies	When you select this check box, very low frequencies (which are typically comprised of noise) are removed from the impulse response. This increases the impulse response's signal-to-noise ratio.
Recover Impulse	Clicking the **Recover Impulse** button starts the impulse recovery process. After the process is complete, an impulse file is created and saved in the folder specified in the **Impulse output file** box.

<table>
<tr>
<td>Impulse recovery mode</td>
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You can choose from three Impulse recovery mode options to determine the method used to recover the impulse: Use the start and end of the recorded file as timing spikes, Auto-detect timing spikes, or Do not use timing spikes. Each of these modes is described below.

Use the start and end of the recorded file as timing spikes This option specifies that the beginning and end of the recorded file are used as timing spikes. This option is recommended for the best results during impulse recovery.
You must trim the file as close to the timing spikes as possible for this method of recovery to work most effectively. The first sample of the file should contain the start of the first spike and the last sample of the file should contain the start of the second spike. Therefore, most of the second spike is deleted.
Auto-detect timing spikes This option specifies that the timing spikes exist near the start and end of the recorded file and that they should be auto-detected. Timing spikes are used to correct for clock or tape speed mismatches. If you have not trimmed the recorded file so that the timing spikes are at the very beginning and end, select this option for the best results.
With this option, you need only ensure that the first spike occurs within one second of the start of the file and that the second spike exists in the file. To improve detection accuracy, you can also boost the level of the start and end spikes in the recorded file.
Tip: If the spike's level is close to the noise floor, select the Use the beginning and end of the recorded file as timing spikes option.
Do not use timing spikes This option specifies that no timing spikes are used. This is the least desirable option as no timing information is used. To use this option, you must trim the recorded file so that the test tone starts and ends at the start and end of the file, with no blank audio before or after. This option should only be used if the timing spikes are lost in the recording or if you are certain that the play and record clocks are synchronized (such as when using an ADAT).

</td>
</tr>
</table>

Creating impulse files

You can obtain impulse responses from anything that accepts test tone input and supports recording the output. This includes physical spaces as well as electronic audio equipment. Creating custom impulse files requires planning, work, and additional audio equipment.

> **Note:** *Impulses derived from electronic devices that produce nonlinear effects such as overdrives, distortion pedals, pitch shifters, harmonic enhancers, chorus pedals, or flange pedals cannot be modeled using the Acoustic Mirror tool. While they produce interesting effects, the acoustic signature cannot be correctly replicated.*

What you need to create custom impulses

The equipment required to create custom impulses depends upon whether you want to create the impulse from a physical acoustic space or from a piece of equipment. Regardless of the method, you need a playback device that reproduces test tones and a recording device that has microphone or line-level inputs. Be aware that the quality of the impulse is directly affected by the quality of your playback and recording devices. The flatter your system's response, the more accurate the impulse response.

Recovering an impulse from an acoustic space

To recover an impulse from an acoustic space, you need the following equipment:

- A playback device and speakers
- A stereo pair of microphones to record the test tone
- A recording device for recording the signal captured by the microphones

Recovering an impulse from an electronic device

To recover an impulse from an electronic device, you need the following equipment:

- A playback device that connects to the device's inputs
- A recording device that connects to the device's outputs

Recording the impulse in an acoustic space

When you have assembled the required equipment, you are ready to begin recording the impulse. The following sections describe the typical impulse recording procedure.

Transferring the test tone

The first step in recording the impulse is to transfer the desired test tone to your playback device. The Sound Forge application disc contains two test tones: a 24-second test tone and a 48-second test tone. We typically recommend that you use the 24-second tone because longer tones result in greater signal-to-noise ratios. The 48-second tone should be used in particularly noisy environments or when the decay time of the acoustic space is greater than six seconds.

> **Tip:** *There are spikes at the beginning and end of each test tone. You should include the spikes in the recording to simplify the recovery of the impulse in the later stages of the process.*

Placing equipment

When recording the test tone in an acoustic space, you must determine where to place your playback system, speakers, microphones, and recording system to produce optimal results. Microphone placement is crucial to the quality of the impulse. The distance between the speakers and the microphone is the perceived distance of audio processed with the impulse you create. For example, if you record the test tone with the speakers positioned 100 feet from the microphones, all sounds processed with the resulting impulse sound as if they are originating 100 feet from the listener.

Setting levels

After the devices are positioned, you should begin playback of the test tone. The test tone should be played as loudly as possible (or practical) to produce the best signal-to-noise ratio. With the test tone playing at optimum volume, set the levels on the recording device. Recording devices levels should also be set as high as possible, but not permitted to clip or distort. Safe levels are determined by whether you are recording to an analog or digital medium.

Recording the test tone

Begin recording on the recording device and begin playback of the test tone. Remember to include the spikes at the beginning and end of the test tone. Record the test tone several times using the initial setup, then move the microphones and record the test tones several more times. Continue moving the microphones and recording until you have exhausted the space's acoustic possibilities. Recording impulses in this manner provides you with several distinct impulses for each space.

Recording the impulse through an electronic device

The recording process is similar if you are recording the output of an electronic device, but there are no speakers or microphones to be placed.

Using the appropriate cables, connect the playback system's outputs to the electronic device's inputs and the electronic device's outputs to the recording system's inputs. Once the devices are connected, play the test tone through the electronic device and record its output on the recording system.

Other impulses

Any number of methods can be used to create an impulse, including starter pistols, clap boards, or even a sharp hand clap. The drawback of these "impulse generators" is that they add their own coloration to the sound. For best results, we recommend using the test tones included on the Sound Forge application disc.

Recovering the impulse

After you have recorded your test tones, they must be processed and converted into impulse responses. The following sections describe the typical impulse recovery procedure.

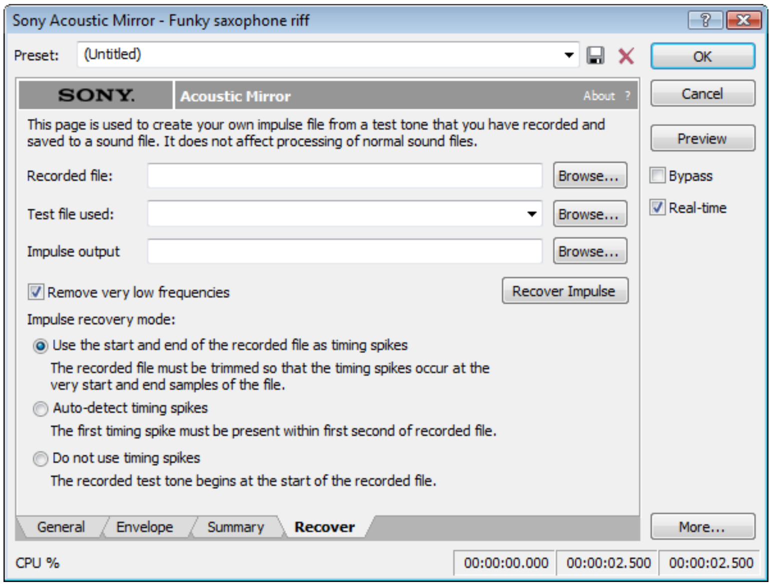
Trimming the test tone

1. Open your test tone file (the "room processed" output test tone) in the software.
2. Locate the first timing spike and delete all audio before it. Cut as close to the beginning of the timing spike as possible, but do not delete the spike itself.
3. Locate the second timing spike and delete all data from the start of the second spike to the end of the file. Again, cut as close to the start of the timing spike as possible.

   You should now have an audio file with a spike at the beginning, a test tone, and silence.
4. Save the test tone file.

Naming, configuring, and recovering the impulse

1. From the **Effects** menu, choose **Acoustic Mirror**. The Acoustic Mirror dialog is displayed.
2. Click the **Recover** tab.



3. Enter the name and path of your impulse file in the **Recorded File** box or click **Browse** and locate the file.
4. Enter the name of the original test tone file in the **Test file used** box.
5. Enter the desired name for the impulse response file to be created in the **Impulse output** box. If necessary, click **Browse** and specify the folder in which the impulse file will be saved.
6. Select the **Remove very low frequencies** check box.
7. If the recorded file was trimmed exactly to the start and end spikes using the procedure described previously, choose **Use start and end of the recorded file as timing spikes** from the **Impulse recovery mode** drop-down list. *For more information, see Trimming the test tone on page 239*.
8. Click the **Recover Impulse** button to begin recovering the impulse.

   After processing is complete, you can open the impulse file in the Sound Forge software and perform any necessary trimming or editing. *For more information, see Trimming the impulse file on page 240*.

Trimming the impulse file

After the impulse file is recovered, it may still require minor trimming. In general, you should try to make the impulse response as short as possible to increase processing speed when using the Acoustic Mirror tool. Impulse files greater than 131,071 samples (about 3 seconds) in length require substantial processing time. When possible, trim the impulse response to less than 65,535 samples (about 1.5 seconds). In addition, we recommend fading the tail of the impulse. Of course, this is not always an option when dealing with spaces that produce extended reverberations.

1. Open the recovered impulse file in Sound Forge software and play it.
2. Delete any silence or low-level noise that occurs before or after the actual audio data. Typically there are between 900 and 1100 samples of data at the beginning of the impulse that should be removed.
3. Save the trimmed impulse response file using the standard WAV format.

> **Tip:** *To prevent phase problems when mixing the dry and wet signals, you may also want to verify the phase of the impulse file. The file should begin by going positive (above the centerline). If the impulse file has a negative (below the centerline) phase, choose* **Invert/Flip** *from the* **Process** *menu.*

Adding summary information to your impulse file

If you plan on sharing impulses with other Sound Forge users, we recommend adding summary information and BMP images to your files.

1. Open the impulse file in the software.
2. From the **View** menu, choose **Metadata**, and then choose **Summary Information** from the submenu. The Summary window is displayed.
3. Enter the appropriate information in each box.

> **Note:** *If the data you want to edit is not displayed in the window, you can right-click the window, choose* **Insert** *from the shortcut menu, and then choose a metadata field from the submenu.*

Using the new impulse file

To use your new impulse file, open the Acoustic Mirror dialog and choose it from the **Impulse** drop-down list as you would any other impulse file. If you performed the previous procedures properly, the custom impulse file should realistically recreate the reverberation characteristics of the electronic device or acoustic space.

Using impulse files in creative ways

Now that you understand the use and creation of impulse files, you may want to begin using the Acoustic Mirror tool in more interesting ways than simply applying an impulse to an audio file. The following sections describe some creative and advanced uses for Acoustic Mirror technology that can contribute to the professionalism of your work.

Processing individual audio elements

Instead of applying an impulse file to an entire song, try applying an impulse to individual elements of the song. Applying an impulse to specific notes, chords, riffs, or phrases can quickly change the dynamics of a song. This technique is possible because the tail of processed audio is automatically mixed with the adjacent unprocessed audio.

Adding realistic stereo to mono recordings

You can give mono recordings realistic stereo characteristics by selecting the **Convert mono to stereo** check box in the **General** tab of the Acoustic Mirror dialog when applying the specified impulse file. The stereo image produced using this method is virtually indistinguishable from an actual stereo recording.

If you choose to use the Acoustic Mirror effect for stereo simulation, you may find the output too reverberant. If this is the case, decrease the **Apply envelope and limit decay** value. Frequently, setting this value to as little as 0.1 seconds provides stereo realism without adding a distracting amount of reverb.

Creating special effects

Processing an audio file using a non-impulse WAV file can produce any number of unexpected and interesting special effects. To demonstrate this concept, create several short (less than 12 seconds) audio files using the **FM Synthesis** tool and save them as individual WAV files. Now choose any of these files from the **Impulse** drop-down list and preview the results.

We have included several short files on the Sound Forge application disc to allow you to experiment with this technique. After some experimentation, you should begin to notice a few general rules regarding this use of the tool:

- Impulse files that cover the entire frequency spectrum prevent the output from sounding too filtered.
- Using a frequency sweep as an impulse creates a frequency-dependent delay effect.
- Panning within the impulse causes the stereo image of the output to flutter between channels.
- Using staccato sounds (such as drum hits) creates a variety of echo effects.

Recreating spaces for foley effects and dialog replacement

Frequently, dialogue recorded in the field is rendered unusable by ambient noise. If you are shooting in the field and realize that overdubbing will be necessary, you should create an impulse in each filming location. This allows you to overdub dialog during post-production that is indistinguishable from dialog recorded on location.

If you intend to use the Acoustic Mirror effect as a film/video post-production tool, there are some factors to keep in mind:

- Distance information is determined by the distance between the source and the microphone when creating the impulse. Record multiple impulses at various distances for each location to create realistic dialog effects when matching audio processing to approximate camera positioning.
- The frequency response of the human ear changes as the volume of a sound increases. As a result, impulses created from a significant distance may sound unusual at high volumes.
- Placing a microphone off center allows you to create directional information in the recovered impulse. For example, placing a microphone to the left of the speaker produces an impulse that approximates a source located on the left side of the screen.

Panning with head-related transfer functions

A head-related transfer function (HRTF) contains the frequency and phase response information required to make a sound seem to originate from a specific direction in a three-dimensional space. The **Acoustic Mirror Impulse Files\HRTF Impulses** folder on the Sound Forge application disc contains a collection of impulse files that contain directional cues.

To achieve optimal results using these impulse files, the original file should be mono and playback should be monitored using headphones. To begin, convert the mono file to stereo by replicating the mono signal in each channel. After the audio is converted to stereo, choose an impulse file from the **HRTF Impulses** folder. You will notice that the **HRTF Impulses** folder is further divided into **Left** and **Right** directories. Opening the desired folder displays the available impulse files, all of which are named based on their elevation (up or down) and azimuth (left or right) angles in degrees. The following table provides some examples:

File Name	Impulse positioning
0E000L	Straight ahead
0E090L	Far left
0E090R	Far right
90E000L	Directly above your head
0E180L	Directly behind you
-20E120L	Below, behind, and to your left

Note: *Refer to* **Readme.doc** *in the* **HRTF Impulses** *folder for more information.*

Troubleshooting the Acoustic Mirror effect

The following sections describe problems that may be encountered when working with the Acoustic Mirror tool.

Stuttering during real-time previewing

It is not uncommon to experience problems when previewing processing in real-time. The following sections contain several suggestions to remedy the situation.

Lower the Quality/speed setting

Lower the value of the **Quality/speed** control on the **General** page. When previewing lengthy impulse responses, a setting of **1** or **2** may be necessary; however, the quality suffers. This setting should always be returned to **5** prior to processing to maintain effect quality.

Increase the DirectX buffering size

1. Open the Acoustic Mirror dialog.
2. Right-click an empty area of the dialog outside of the four tabs and choose **Configuration** from the shortcut menu. The Real-Time Preview Configuration dialog is displayed.
3. Reconfigure the **Buffers to process per second** and **Total playback buffers** controls. Typically, lowering the **Buffers to process per second** value and increasing the **Total playback buffers** value reduces gapping during real-time previewing.

Close all memory-intensive applications

Real-time previewing may be limited by any additional applications operating on the desktop. To avoid this situation, close all memory-intensive applications prior to using this effect.

Add additional RAM to the system

We require at least 512 MB of RAM to operate Sound Forge software and its related tools.

Add a faster floating point arithmetic processor

Many high-speed processors are still lacking in speed when processing floating point arithmetic. We recommend using high-speed processors that provide exceptional floating point arithmetic for reliable real-time previewing.

Impulses do not recover properly

If you experience problems recovering custom impulse recordings, verify each of the following:

1. Verify that you have trimmed the recorded test tone based on the mode chosen from the **Impulse recovery mode** drop-down list. *For more information, see Trimming the test tone on page 239*.
2. Verify that the second spike is present in the recorded test tone if the **Auto-detect timing spikes** options is specified.
3. Verify that the file specified in the **Test file used** box is the exact test tone used to make to field recording and that neither its length or data has been changed.
4. If the impulse still does not recover properly in **Auto-detect timing spikes** mode, normalize the spikes in the recorded test tone file. This should aid the auto-detect algorithm in detecting the timing spikes and recovering the impulse.

Recovered impulse is too noisy

To maximize the impulse's signal-to-noise ratio, you should verify that the field recording's noise floor is not too high. When recording in noisy environments, increase the test tone's amplitude until the test tone is at least 25 dB louder than the noise floor. At least 40 dB of signal-to-noise is recommended for optimal impulses. If you cannot avoid noise when recording in the field, the Noise Reduction tool can salvage a session.

Speaker nonlinear distortion can also cause noisy impulses. The most common source of nonlinear distortion is loudspeaker harmonics. Most speakers display substantial harmonic distortion at low frequencies. For example, when you play a 60 Hz tone, the speaker vibrates at 60 Hz, but also outputs lower-level audio at multiples of 60 Hz (120, 180, etc.). The impulse recovery method greatly minimizes these low-frequency distortions; however, inexpensive tweeters often display substantial high-frequency distortion that can disrupt the recovery process. When possible, use high-quality components and do not overdrive the speakers.

Error message explanations

The following sections briefly describe Acoustic Mirror error messages that you may encounter.

The selected file is not a valid test file

The file specified in the **Test file used** box is not a test tone file included on the Sound Forge application disc.

The level of the first spike is low. Do you wish to use it as a timing spike?

This typically means that no actual timing spike was detected. Verify that the first spike is within one second of the start of the recorded file. If the recording is noisy and the spike is not very pronounced, you can aid detection by muting the audio immediately before and after the spike.

An error occurred reading the test tone file

Either the test tone file was not found or is not a valid test tone file. Always use a test tone file provided on the Sound Forge application disc.

The selected Recorded file is much smaller than the test tone size

This may indicate that the test tone or the recorded file specified in the **Recover** tab is not correct. Verify that the length of the recorded file roughly the same size as the test tone file.

The end spike was not found

Verify that the spike following the test tone is present in the recorded file when recovering impulses in **Auto-detect timing spikes** mode.

What is the Wave Hammer plug-in?

The Wave Hammer DirectX plug-in is an audio mastering tool consisting of a classic compressor and a volume maximizer.

The Wave Hammer tool can be used in any Microsoft DirectX-compatible host application (for example, Sound Forge and ACID® Pro software), and the quality and functionality of the Wave Hammer plug-in is the same in each host application; however, the method of previewing effects is different. Consult your host application's documentation to determine the available previewing methods.

Displaying the Wave Hammer plug-in

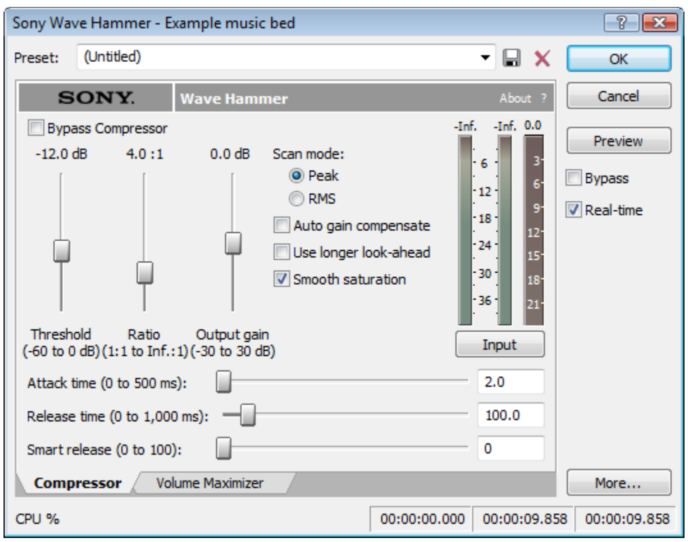
To display the Wave Hammer tool, choose **Wave Hammer** from the **Effects** menu.

The Wave Hammer dialog

The Wave Hammer controls are divided into two tabs: **Compressor** and **Volume Maximizer**.

Compressor tab

The controls on the **Compressor** tab are used to compress the audio signal. When applied properly, compression reduces the dynamic range of audio and allows you increase overall loudness. Compression has various uses. For example, applying heavy compression at a low threshold to electric guitar produces distortion. The controls are explained below.

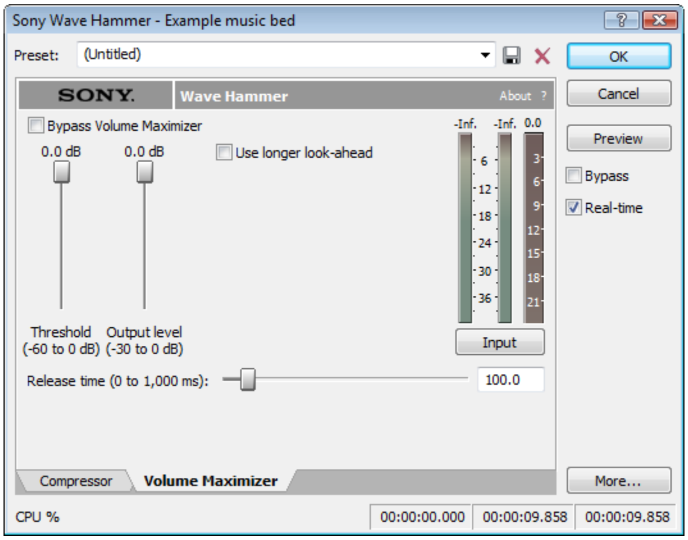


Control	Description
Threshold	The **Threshold** fader is used to adjust the audio level at which compression is applied. Audio with levels higher than the **Threshold** value are compressed, while audio at levels lower than this value pass through the compressor uninterrupted.
Ratio	The **Ratio** slider determines the amount of compression applied to audio signals surpassing the threshold. A ratio of **1:1** applies no compression to audio surpassing the threshold, while a ratio of **2:1** requires a 2 dB increase in actual volume to raise the processed volume 1 dB. A ratio of **Inf:1** is considered a limiter.
Output gain	The **Output gain** fader allows you to determine how much the audio signal is boosted following its compression.
Attack time	The **Attack time** slider allows you to determine how soon after rising above the threshold the audio signal is attenuated.
Release time	The **Release time** slider allows you to determine how soon after falling below the threshold the audio signal attenuation is interrupted.

Control	Description
Smart release	The **Smart Release** slider allows you to configure the compressor to automatically increase the release time for sustained notes and decrease the release time for sharp transients. Setting this value higher increases the internal variability of the specified **Release** value. Generally, louder overall audio levels can be achieved with lower **Release** values. However, low **Release** values can also lead to an increase in "pumping" artifacts. Configuring a **Smart Release** value increases the release time during sustained sounds, thereby preventing release changes from occurring too rapidly.
Scan mode	The **Scan mode** radio buttons allow you to specify whether **Peak** or **RMS** mode is used to determine the loudness of an audio file, which in turn determines the amount of compression that is applied. When compressing in **Peak** mode, the compressor applies compression where it detects audio signal peaks that surpass the threshold. However, when compressing using **RMS** mode, the compressor processes the audio using the detected average RMS value of the entire file. The Root Mean Square (RMS) of audio is a measure of its intensity over a period of time. Therefore, the RMS level of audio corresponds to the loudness perceived by a listener when measured over small intervals of time. As a result, rapid transient peaks may not be processed when compressing in **RMS** mode.
Auto gain compensate	When you select the **Auto gain compensate** check box, the compressor output is boosted by a constant amount derived from the **Threshold** and **Ratio** settings. This option prevents a loss in overall level when compressing audio. **Tip:** *When using the* **Auto gain compensate** *option, the* **Output gain** *fader should be used to fine tune the signal output level.*
Use longer look-ahead	When you select the **Use longer look-ahead** check box, the compressor scans farther ahead in the incoming audio to determine how much compression is needed. This results in compression being applied before the threshold-surpassing audio actually occurs, thereby allowing for a slower **Attack time** value. However, the pre-compression effect (fades that occur prior to attacks) of this option may be distracting.
Smooth saturation	Selecting the **Smooth saturation** check box lowers the amount of distortion caused when applying heavy compression. When this option is turned on, the compressed audio sounds warmer and not overly bright.
Input/Output meter	This meter allows you to monitor the level of the incoming and outgoing signals. When the **Input** button is displayed, the meters display the incoming signal level. Clicking **Input** toggles the button to an **Output** button and displays the outgoing signal level. Clicking **Output** returns you to the incoming signal display.
Attenuation meter	This meter allows you to monitor the audio signal attenuation derived from the current settings.

Volume Maximizer tab

The controls on the **Volume Maximizer** tab are used to limit the peak amplitude of an audio file or to boost the overall level without clipping the waveform and distorting the audio. These controls are explained below.



Control	Description
Threshold	The **Threshold** fader is used to adjust the audio level at which the volume maximizer activates. Audio with levels higher than the **Threshold** value are affected, while audio at levels lower than this value pass through the volume maximizer uninterrupted.
Output level	The **Output level** fader allows you to determine the level to which peaks above the **Threshold** setting are boosted or cut.
Release time	The **Release time** slider allows you to determine how soon after falling below the threshold the audio signal attenuation is interrupted.
Use longer look-ahead	When you select the **Use longer look-ahead** check box, the volume maximizer scans farther ahead in the incoming audio to determine the amount of limiting that is needed. This results in limiting being applied before the threshold-surpassing audio actually occurs. However, the pre-limiting effect (fades that occur prior to attacks) of this option may be distracting.
Input/Output meter	This meter allows you to monitor the level of the incoming and outgoing signals. When an **Input** button is displayed, the meters are displaying the incoming signal level. Clicking **Input** toggles the button to an **Output** button and displays the outgoing signal level. Clicking **Output** returns you to the incoming signal display.
Attenuation meter	This meter allows you to monitor the audio signal attenuation derived from the current settings.

Chapter 16

Using Scripting

You can use scripting to streamline repetitive tasks and implement customized features. When the Script Editor window displays, you can use it to create, edit, or run scripts.

Sound Forge® software can use scripts written using JScript, VBScript, or C# as well as scripts that have been compiled as DLLs.

Scripting references

Sample scripts

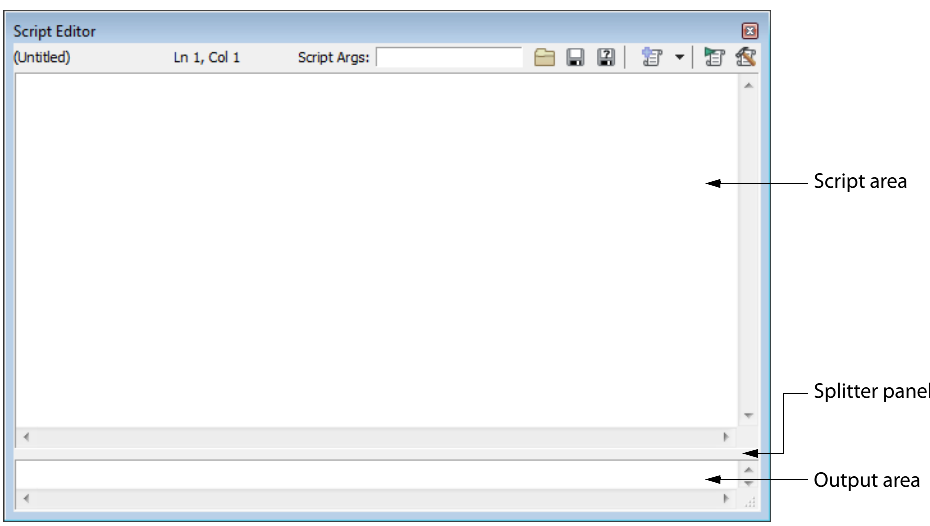
You can find the most recent scripting API (application programming interface) and sample scripts on our Web site at *http://www.sonycreativesoftware.com/download/devkits*.

Additional scripting information

For additional information about scripting, we encourage you to check out the Sound Forge scripting forum on our Web site at *http://www.sonycreativesoftware.com/forums/showtopics.asp?forumid=27*.

Using the Script Editor window

From the **View** menu, choose **Script Editor** to display the Script Editor window. You can use the Script Editor window to open, run, create, or edit scripts.



- **Script area** - Displays the current scripts written code.
- **Output area** - Displays text results for the current script.
- **Splitter panel** - Allows you to adjust the size of the output area window by dragging it up or down.

The Script Editor toolbar is displayed by default when you open the Script Editor window.

Open Opens the Open Script dialog.	**New Script Template** Opens a basic C#, JScript, or VBScript template needed to write a script.
Save Saves the current script.	**Run Script** Runs the current script.
Save As Saves the current files with a new name or format.	**Compile Script** Compiles and tests your script.

Opening and running a script

You can open and run a script that has already been developed.

> **Warning:** *Scripts can pose a security risk to your computer. A script has the power to delete files, read files, write files, execute programs, access the Internet, access files on your network, and so on. Always examine the contents of a script before running it. If you don't understand the script, do not run it unless it comes from a trusted source. In general, take the same precautions you would take for any program you download from the Internet or receive in an e-mail attachment.*

Running a script from the Script Editor window

1. Click in the data window where you want to apply the script to establish focus.
2. From the **View** menu, choose **Script Editor** to display the Script Editor window if it isn't already displayed.
3. Click the **Open** button ( ) in the Script Editor toolbar. The Open Script window is displayed.
4. Select the script file (.vb, .js, .cs, or .dll) that you want to run. The script data is displayed in the top portion of the Script Editor window.
5. If you need to pass an argument to the script, type it in the **Script Args** box. Arguments are specified as follows: **ArgName=ArgValue&ArgName2=ArgValue2. . .**

   For example, a script that uses an argument to indicate where files should be saved could use the following argument to save files to a ScriptOutput folder on your D:\ drive: **dir=d:\ScriptOutput**.

   If your script can create a log file to record the results of a scripting operation, you could append that argument as follows: **dir=d:\ScriptOutput&logFileName=myLog.txt**. In this example, the ampersand (&) separates the two arguments. The first argument sets the save folder, and the second argument sets the file name for the log file.

   For more information, see Script arguments on page 251.
6. Click the **Run Script** button ( ).

Running a script from the Scripting menu

1. Click in the data window where you want to apply the script to establish focus.
2. From the **Tools** menu, choose **Scripting**.
3. Choose a script from the submenu or choose **Run Script** from the submenu to browse to the script file (.vb, .js, .cs, or .dll) that you want to run.

Running a script from the command line

In addition to running scripts from the Script Editor and the Scripting menu, you can run scripts directly from the command line using the following commands.

SCRIPT

Starts Sound Forge and runs the specified script.

Example: `"C:\Program Files\Sony\Sound Forge Pro 11.0\Forge110.exe" -SCRIPT:"C:\Scripts\MyScript.cs"`

SCRIPTARGS

Starts Sound Forge and passes the specified arguments to a script.

Example: `"C:\Program Files\Sony\Sound Forge Pro 11.0\Forge110.exe" -SCRIPTARGS:"in\C:\Test\input.dls&out=C:\Test\output.dls&repeat=2" -SCRIPT:"C:\Scripts\MyScript.cs"`

EXIT

Exits Sound Forge after running the specified script.

Example: `"C:\Program Files\Sony\Sound Forge Pro 11.0\Forge110.exe" -SCRIPT:"C:\Scripts\MyScript.cs" -EXIT`

Script arguments

A script can accept arguments to dynamically change the behavior of a script. Arguments allow you to develop a single script that performs multiple functions controlled by the arguments sent to the script.

Each argument is a key/value pair. The *key* is a string that identifies the argument's *value*. Multiple arguments are delimited by an ampersand (&).

```
key1=value1&key2=value2&key3=value3…
```

When you supply arguments to a script, the following static functions can be used to extract and format the parameters:

```
public static string GETARG(string key, string str) { string val = Script.Args.ValueOf(key); if (val == null) val = str; return val; }
public static int GETARG(string key, int ii) { return Script.Args.AsInt(key,ii); }
public static Int64 GETARG(string key, Int64 cc) { return Script.Args.AsInt64(key,cc); }
public static bool GETARG(string key, bool ff) { return Script.Args.AsBool(key,ff); }
public static double GETARG(string key, double dd) { return Script.Args.AsDouble(key,dd); }
```

The first argument to the GETARG functions specifies a key name that is used to identify the argument to be extracted. The second argument in the GETARG function is a default value to be returned if the function cannot find the key name. The second argument also determines which overloaded function the script will use and how the value will be formatted.

For example, consider a script that accepts three input parameters. The syntax for the arguments is as follows:

```
in=C:\Test\input.dls&out=C:\Test\output.dls&repeat=2
```

The script to handle the parameters would look as follows:

```
using System;
using System.IO;
using System.Collections;
using System.Runtime.InteropServices;
using System.Windows.Forms;
using System.Drawing;

using SoundForge;
using SoundForge.BatchConverter;

public class EntryPoint {
public string Begin(IScriptableApp app) {

    string inFile  = GETARG("in", "");
    string outFile = GETARG("out", "" );
    int count = GETARG("repeat", 0 );
```

Adding scripts to the Scripting menu

When you start the program, Sound Forge software looks at the Script Menu folder in the Sound Forge program folder to determine which scripts appear in the Scripting submenu. This folder is C:\Program Files\Sony\Sound Forge Pro 11.0\Script Menu by default.

1. To change the contents of the submenu, add or delete scripts in the Script Menu folder.

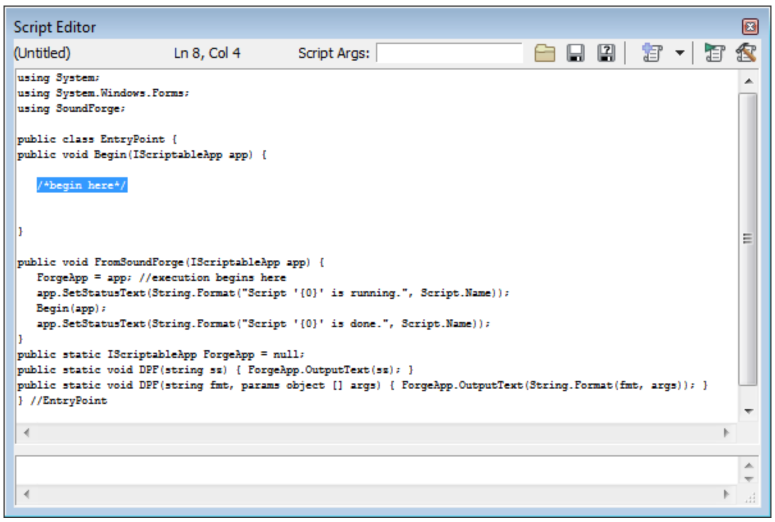
> **Tip:** *To prevent duplication of script files, you can use shortcuts in the Script Menu folder.*

2. From the **Tools** menu, choose **Scripting** and then choose **Rescan Script Menu Folder** to update the menu.

Creating a script

Sound Forge scripting uses the Microsoft .NET framework for scripting. You can write scripts in JScript, Visual Basic .NET, or C#.

1. From the **View** menu, choose **Script Editor** to display the Script Editor window if it isn't already displayed.
2. Click the **New Script Template** button ( ) and choose **C#**, **JScript**, or **VBScript**. A new script is displayed in the Script Editor window, with what is needed to write a script.



3. Replace the /***begin here***/ text with your script.
4. Click the **Compile Script** button ( ) to compile and test your script. If there are any errors, they will be displayed at the bottom of the window.
5. Click the **Save** button ( ) to choose the file name and location that you want to use to save your script.

Editing an existing script

Editing a script in the Script Editor window should not be very difficult as the scripts that are included with Sound Forge software are fully commented to help you find and edit the parameters you need.

1. From the **View** menu, choose **Script Editor** to display the Script Editor window if it isn't already displayed.
2. Click the **Open** button ( ) in the Script Editor window, choose the script you want to edit and then click **Open**. The script data is displayed in the top portion of the Script Editor window.

> **Note:** *You cannot edit scripts that have been compiled as DLLs.*

3. Edit the script as necessary. The comments in the script will help you find the parameters you need to edit.

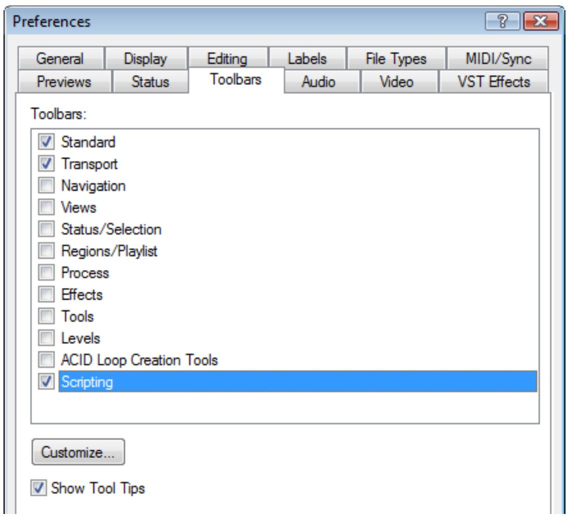
> **Note:** *Comments are indicated with double forward slashes: //.*

4. Click the **Compile Script** button ( ) to compile and test your edited script. If there are any errors, they will be displayed at the bottom of the window.
5. Click the **Save** button ( ) to replace the script you edited or click the **Save As** button ( ) to save the edited script with a different name or in a different location.

Using the Scripting toolbar

Adding or removing toolbar buttons

1. From the **View** menu, choose **Toolbars**. The Preferences dialog appears with a list of available toolbars.
2. Select the **Scripting** check box.



3. Click **Customize**. The Customize Toolbar dialog appears.
4. Use the controls in the Customize Toolbar dialog to add, remove, or rearrange the buttons on the selected toolbar. All scripts from the Script Menu folder are listed in the **Available tools** column.

If you want to	Then
Add a script to the toolbar	Select a script in the **Available tools** column and click the **Add** button. **Note:** *The script will appear before the currently selected button.*
Remove a script from the toolbar	Select a script in the **Current tools** column and click the **Remove** button.
Rearrange the buttons	Select a script in the **Current tools** column and click the **Move Up** or **Move Down** button.
Restore the toolbar to its default setting	Click the **Reset** button.

5. Click the **OK** button.

Creating custom button images

You can display custom button images for the scripts that you have added to the toolbar by adding .png files to your Script Menu folder.

1. Create a .png file with the icon that you want to use.

> **Note:** *Icons should be 16x16 pixels. Transparency is supported.*

2. Save the .png file in your Script Menu folder using the same name as the name of the script that you want it to represent (i.e. to assign a custom icon to the HelloWorld.js script, the icon should be saved as HelloWorld.js.png).

> **Note:** *The Script Menu folder can typically be found in the following location: C:\Program Files\Sony\Sound Forge Pro 11.0\Script Menu.*

3. Customize the toolbar as needed. The custom icons will display on the Scripting toolbar the next time you start the application.

Running a script

You can run scripts using a single click if you have customized the Scripting toolbar to include buttons for scripts that you have created.

1. Click in the data window where you want to apply the script to establish focus.
2. Click the button of the script that you would like to run on the Scripting toolbar.

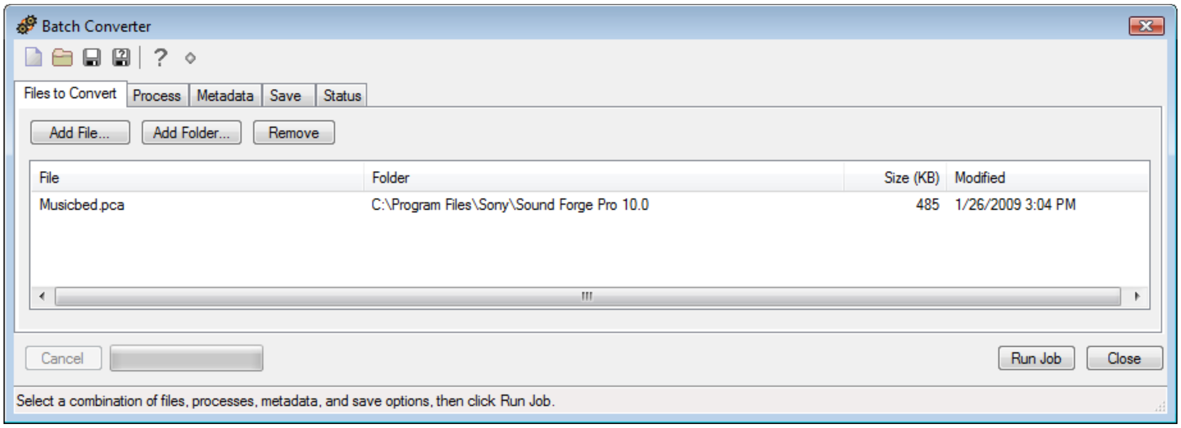
> **Tip:** *You may need to hover a button to display a ToolTip, which displays the name of the script associated with the button.*

Using the Batch Converter

You can use the Batch Converter to modify and manipulate audio files without having to process each file individually.

Converting files

1. From the Tools menu, choose **Batch Converter**. The Batch Converter window is displayed.



2. Open the batch job that you want to run. If you're not using a saved batch job, continue to step 3.
   a. Click the **Open Job** button (). The Open dialog is displayed.
   b. Browse to the folder where your batch job (.bj) file is saved.
   c. Select a batch job and click the **Open** button.
3. Select the Files to Convert tab and add the files that you want to process. All data windows that were open in the Sound Forge workspace when you started the Batch Converter are automatically included in the list.
   - Click the **Add File** button on the toolbar, browse to a file, and click the **Open** button to add individual files.
   - Click the **Add Folder** button on the toolbar, select a folder, and click the **OK** button to add all files within a folder. Only the contents of the folder you select are added; subfolders are not included.
   - Drag files from the Windows Explorer to the Files to Convert tab.
4. Select the Process tab and verify the processing settings. *For more information, see Creating or editing a batch job on page 256.*

   If you're simply converting to another file format, continue to step 5.

> **Tip:** *When you convert files to a compressed format such as MP3, peaks that are at or near 0 dB may be clipped by the compression process. Consider normalizing first to reduce the possibility of clipped peaks (normalizing to a peak level of -0.9 dB is a good starting point).*

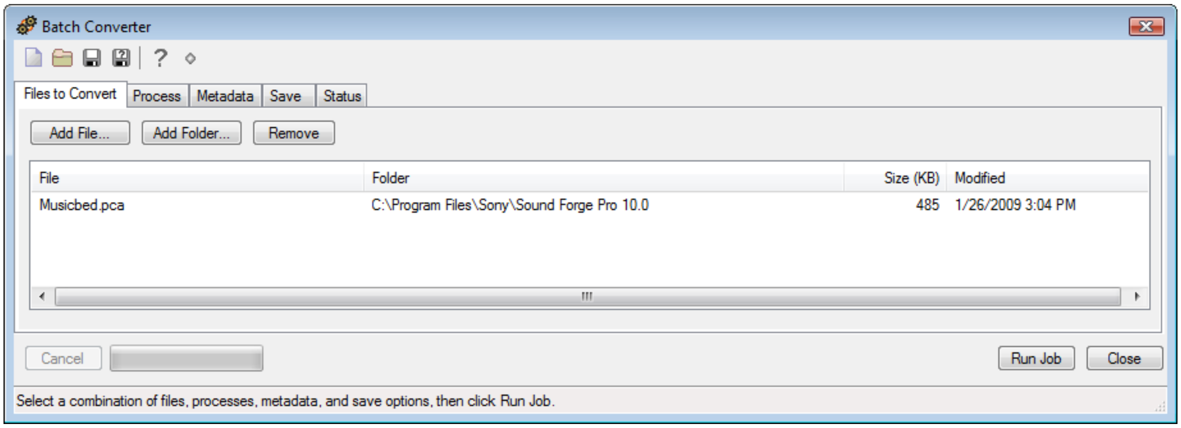
5. Select the Metadata tab and type values for any metadata (file information) that you want to save in the output files.
   - If the **Overwrite** check box is not selected and the destination file already includes information for a metadata item, the existing information is preserved (keywords, however, will be appended).
   - If the **Overwrite** check box is selected and the destination file already includes information for a metadata item, the existing information is overwritten with the information from the Metadata tab (existing information will be erased if the box is blank).
   - If a metadata type is not supported by the output format, it will be ignored.
6. Select the Save tab and verify the file output settings. *For more information, see Creating or editing a batch job on page 256.*

   If you want to convert to multiple formats at once, click the **Add Save Options** button to create a setting for each file type that you want to convert.
7. Click the **Run Job** button to start processing. After you click the button, the Batch Converter switches to the Status tab so you can monitor the progress of your batch job.

Creating or editing a batch job

A batch job contains the settings that will be used to convert files. If you routinely perform similar processing on multiple files, a batch job can combine multiple, time-consuming steps into a single process.

1. From the Tools menu, choose **Batch Converter**. The Batch Converter window is displayed.



2. Create a new batch job or open the batch job that you want to edit.
   - Click the **New Batch Job** button ( ) to create a new batch job.
   - Click the **Open** button ( ) and choose a batch job (.bj) if you want to edit an existing batch job.
3. Select the Process tab to choose the processing settings that you want to apply.

> **Tip:** *When you convert files to a compressed format such as MP3, peaks that are at or near 0 dB may be clipped by the compression process. Consider normalizing first to reduce the possibility of clipped peaks (normalizing to a peak level of -0.9 dB is a good starting point).*

   a. Choose a plug-in from the **Select** drop-down list, and click the **Add Effect** button to add it to the end of the list. The plug-in dialog is displayed.

   b. Use the plug-in dialog to adjust the effect's settings. For more information about an individual effect's settings, click the **Help** button (?) in the plug-in window.

   c. Repeat steps 3a and 3b as necessary to create your effects list.

   d. If you need to change an effect's preset, select the effect in the list and click the **Change Preset** button.

   e. If you need to change an effect's position in the list, select it and click the **Move Up** or **Move Down** button.

4. Select the Metadata tab and type values for any metadata (file information) that you want to save in the output files.
   - If the **Overwrite** check box is not selected and the destination file already includes information for a metadata item, the existing information is preserved (keywords, however, will be appended).
   - If the **Overwrite** check box is selected and the destination file already includes information for a metadata item, the existing information is overwritten with the information from the Metadata tab (existing information will be erased if the box is blank).
   - If a metadata type is not supported by the output format, it will be ignored.
5. Select the Save tab to choose file output settings for rendered files.

> **Note:** *If you don't specify save options, the settings on the Process and Metadata tab will be applied to your source files, but the modified files will not be saved.*

   a. Click the **Add Save Option** button to create a new setting or select an existing setting and click the **Change Save Options** button. The Output Options dialog is displayed.

   If you want to convert to multiple formats at once, click the **Add Save Options** button to create a setting for each file type that you want to convert.

b. In the File Format section, select a radio button to indicate the format that you want to use for processed files:

Button	Description
Same as source	Select this radio button if you want to save converted files using the same format as the original file.
Convert to	Select this radio button and choose a file type from the **Type** drop-down list if you want to convert your files to a new format. Choose a setting from the **Template** drop-down list to choose the parameters that will be used for rendering your file, or click **Custom** to choose a new template (custom templates are not available for .vox, .ivc, .au, or .dig files). **Tip:** *For any output format, choose* **Default Template** *to preserve the source file's format (sample rate, bit depth, and number of channels) in the output file.*

c. In the File Names section, select a radio button to indicate the format that you want to use for processed files:

Button	Description
Same as source	Select this radio button if you want to save converted files using the same name as the original file.
Append to name	Select this radio button and type text in the **Append to name** box if you want to add a descriptor to the file names of converted files. The text you enter will be added to the original file name during conversion. For example, if your source file is C:\Audio\DoorSlam.wav, the file could be saved as C:\Audio\DoorSlam-BatchConverted.wav during conversion.

d. In the File Folder section, select a radio button to indicate where you want to save processed files:

Button	Description
Same as source	Select this radio button if you want to save converted files in the same folders as the original files.
Save files to	Select this radio button and type a path in the edit box (or click **Browse**) in the **Append to name** box if you want to save all converted files in a specific folder. You can select the **Preserve source subfolders** check box if you want to use the same folder structure in your source and converted files. For example, if your source file is C:\Audio\DoorSlam.wav, you could specify D:\ as your output folder, and the file will be saved as D:\Audio\DoorSlam.wav during conversion.

e. Click the **OK** button.

6. Click the **Save** button () to save the updated batch job or click the **Save As** button () to save the edited batch job with a different name.

You're now ready to add files and run the batch job.

Chapter 17

Sampling

Used in conjunction with the Sampler Tool, Sound Forge® Pro software's powerful editing capabilities allow you to create, edit, and transfer samples between external and internal samplers. This chapter describes the procedures used to transfer (dump) samples between the computer and sampler with the Sampler Tool.

Samplers

Samplers are devices that produce on-demand playback of audio samples at varying pitches. For the purposes of this manual, we will concentrate on two basic varieties: external samplers and internal samplers.

External samplers

External samplers are typically capable of recording samples or transferring prerecorded samples into their memory. You can choose between two methods to transfer samples to external samplers:

- MIDI Sample Dump Standard (SDS)
- SCSI MIDI Device Interface (SMDI)

MIDI Sample Dump Standard (SDS)

The MIDI SDS is used to send and receive digital samples using normal MIDI hardware and cable connections. Due to the limited bandwidth of the MIDI protocol and the large amount of data required by digital samples, a MIDI SDS transfer can be time consuming. Furthermore, SDS is limited to mono samples, though certain samplers allow two mono samples to be joined as a stereo sample.

SCSI MIDI Device Interface (SMDI)

The SCSI MIDI Device Interface (SMDI) allows music hardware and software to communicate using SCSI hardware and cables. Because SCSI hardware has a greater bandwidth than MIDI, SMDI transfers are considerably faster than SDS transfers. In addition, SMDI supports mono and stereo sample transfers.

Internal samplers

Internal samplers are cards installed in your system that, unlike typical sound cards, actually allow sounds to be downloaded into memory and played at varying pitches to simulate a musical instrument.

Using an unsupported internal sampler

If you have an internal sampler not directly supported by the Sampler Tool, you have two options:

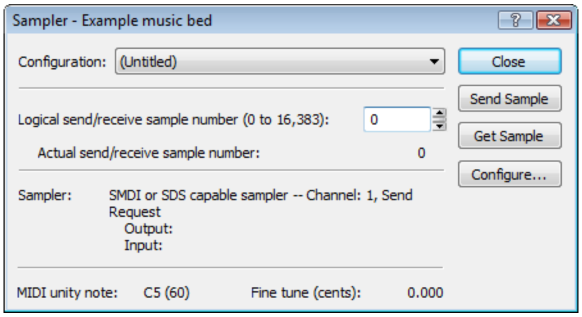
- Use the MIDI SDS transfer protocol.
- Use an open loop transfer.

Note: *If you have a Windows-compatible internal sampler, contact the manufacturer about supporting SDS in Windows drivers.*

Configuring the Sampler Tool

Configuring the Sampler Tool is fairly straightforward, especially if the desired configuration exists in the list of presets.

1. From the **Tools** menu, choose **Sampler.** The Sampler dialog appears.



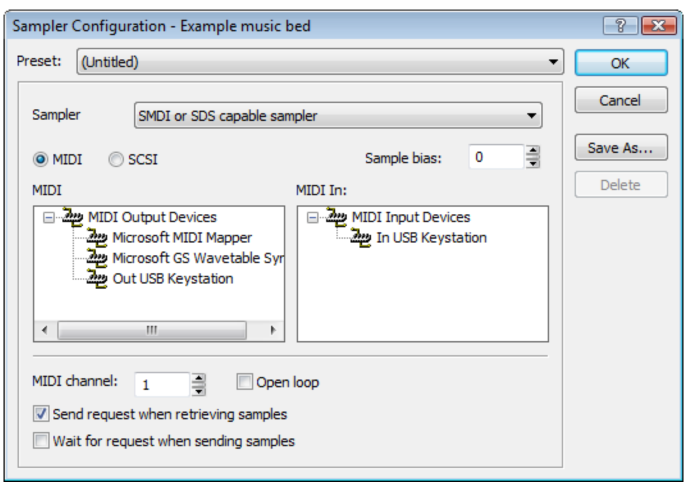
2. From the **Configuration** drop-down list, choose the desired configuration. If the desired configuration is not listed, you must create it in the Sampler Configuration dialog. *For more information, see Creating a sampler configuration on page 260*.
3. Enter a value in the **Logical send/receive sample number** box.

   This value determines the number that the sampler uses as its location reference when sending or receiving samples. This number can be biased for specific samplers with the **Sample bias** option in the Sampler Configuration dialog. *For more information, see Creating a sampler configuration on page 260*.
4. Begin the process of sending or receiving samples. *For more information, see Sending and receiving samples on page 262*.

Creating a sampler configuration

The Sampler Configuration dialog allows you to create new sample configurations that can be saved as presets and accessed from the Sampler dialog. Creating new custom configurations requires you to specify the sampler and sample transfer mode. However, the process of creating a custom sampler configuration differs based on which transfer mode is used.

1. From the **Tools** menu, choose **Sampler.** The Sampler dialog appears.
2. Click the **Configure** button. The Sampler Configuration dialog appears.



3. From the **Sampler model** drop-down list, choose the appropriate sampler. If the desired sampler is not included in the drop-down list, choose the generic **SMDI or SDS capable sampler** option. If the sampler supports the specified protocol, the Sampler Tool should interface with the sampler.

4. Specify input/output settings for the sampler:

If	Then
Your sampler uses MIDI/SDS transfer	Select the **MIDI** radio button and choose input and output ports in the **MIDI In** and **MIDI Out** boxes
Your sampler uses SCSI/SMDI transfer	Select the **SCSI** radio button and select your sampler in the **Sampler** box.

Note: *The* ***Sampler*** *box lists all devices connected to the selected SCSI host, including devices that are not samplers.*

5. If desired, enter a value in the **Sample bias** box. Sample bias is a user-specified value that is added to the logical sample number to determine the actual sample number used for sending or receiving.

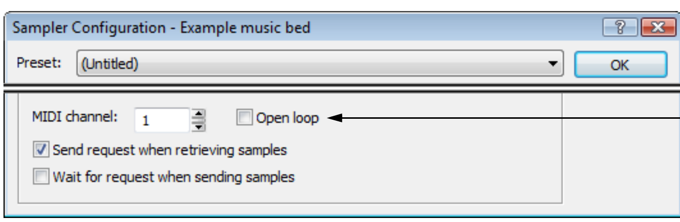
   Additionally, sample bias can be used to define unique biases for multiple projects. For example, when composing multiple pieces using different samples, it is possible to create unique sampler configurations for each project. Simply establish a unique sample bias to segregate the samples within the sampler.
6. Enter a value in the **MIDI channel** box to specify which MIDI channel (1-16) is used when transferring samples.
7. Select the **Open loop** check box if you want to send SDS sample data immediately upon clicking the **Send Sample** button. This is an unconditional transfer of sample data (no handshake).
8. Select the **Send request when retrieving samples** check box if you want the Sampler Tool to send a request for the sample to the sampler when you click **Get Sample**.

   Clearing the **Send request** check box requires that the sample transfer be initiated from the sampler, even after you click **Get Sample**. Typically, pressing the appropriate button on the sampler satisfies this request.
9. Select the **Wait for request when sending samples** check box if you want the Sampler tool to wait for the sampler to request the sample transfer before sending the sample, even after you click **Send Sample**. Typically, pressing the appropriate button on the sampler satisfies this request.

   Clearing the **Wait for request** check box configures the Sampler Tool to send the sample as soon as you click **Send Sample**.

Open loop versus closed loop

Open loop describes a unidirectional communication protocol. When the **Open loop** check box is selected, the source transmits all data to the destination without listening for instruction from the destination. The destination has no control over how the data is sent and cannot ask for information to be repeated. This lack of feedback makes open-loop transfers prone to error.



The **Open Loop** check box in the Sampler Configuration dialog

If the **Open loop** check box is cleared, the communication protocol is referred to as closed loop. A closed loop allows information to flow in both directions. Using closed-loop transfers, the source sends data in small packets and the destination, upon receiving the packet, either retains the data or discards the packet and requests the data to be resent. Using closed-loop protocol, the source does not send the next packet of data until the destination requests it. This makes closed-loop transfers more reliable than open-loop transfers.

In addition to being less reliable, open-loop transfers are slower than closed-loop transfers, especially when sending samples using the Sampler Tool. This is due to intentional delays placed between data packets to compensate for varying sampler speeds. Closed-loop transfers typically guarantee the most efficient timing between packets.

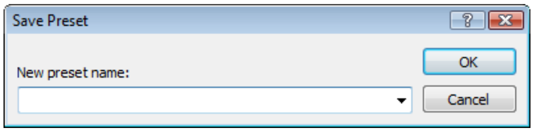
If possible, avoid using an open loop to receive samples from a sampler. The Sampler Tool cannot control the flow of data packets and there is a high probability that data will be missed.

Tip: *Open-loop transfers can be useful when you do not have enough cables to connect both the MIDI input and MIDI output ports.*

Saving sampler configurations

Once you complete a sampler configuration, you can save it as a preset and quickly access it in the future.

1. From the Sampler Configuration dialog, click **Save As**. The Save Preset dialog appears.



2. Enter a descriptive name in the **New preset name** box and click **OK**. The new configuration is saved and can now be chosen from the **Configuration** drop-down list in the Sampler dialog.

> **Note:** *To delete a preset, choose it from the* **Preset** *drop-down list and click* **Delete**.

Sending and receiving samples

Once you have accurately configured the sampler setup, you can send and receive samples using the **Send Sample** and **Get Sample** buttons in the Sampler dialog.

Sending a sample

1. From the **Tools** menu, choose **Sampler.** The Sampler dialog is displayed.
2. From the **Configuration** drop-down list, choose the sampler configuration.
3. Enter the sample number to be sent in the **Logical send/receive sample number** box. The Sampler Tool takes into account the configuration's sample bias and displays values for the **Actual send sample number** and **Actual receive sample number**. *For more information, see* *Creating a sampler configuration on page 260*.
4. Click **Send Sample**. Sample transfer starts. A meter in the status bar indicates the progress of the transfer. You can cancel a transfer at any time by clicking **Cancel** or pressing Esc.

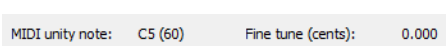
Receiving a sample

1. From the **Tools** menu, choose **Sampler.** The Sampler dialog appears.
2. From the **Configuration** drop-down list, choose the sampler configuration.
3. Enter the sample number to be received in the **Logical send/receive sample number** box. The Sampler Tool takes into account the configuration's sample bias and displays values for the **Actual send sample number** and **Actual receive sample number**. *For more information, see* *Creating a sampler configuration on page 260*.
4. Click **Receive Sample.** Sample transfer starts. A meter in the status bar indicates the progress of the transfer. You can cancel a transfer at any time by clicking **Cancel** or pressing Esc.

MIDI unity note and Fine tune

Once you specify a configuration in the Sampler dialog, the **Sampler** area near the bottom of the dialog displays all relevant sampler configuration information.

The bottom pane of the dialog contains two additional parameters: **MIDI unity note** and **Fine tune**.



Parameter	Description
MIDI unity note	The **MIDI unity note** value indicates the pitch to which the sample is tuned.
Fine tune	The **Fine tune** value indicates any minor tuning differences (measured in cents) in the sample.

Editing MIDI unity note and Fine tune

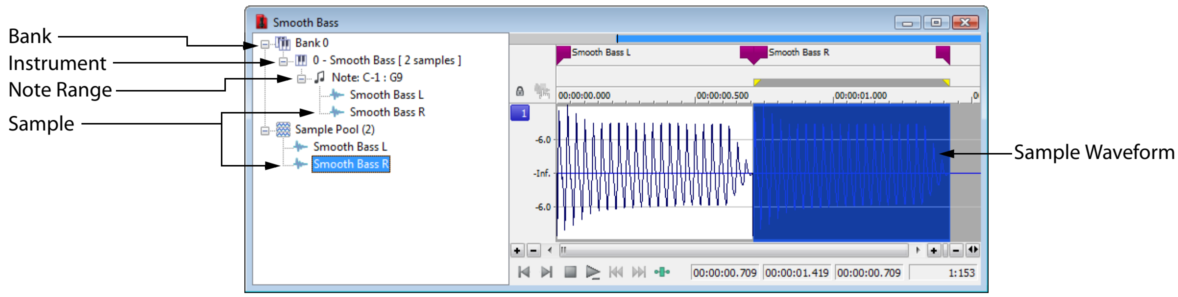
Both values can be edited and used with samplers that support tuning information in the Sampler Loops window. *For more information, see* *Sampler Loops window (Ctrl+Alt+M, 6) on page 57*.

Note: *The software does not use this information.*

Processing musical instrument files

Sound Forge can open and save DLS, GigaStudio/GigaSampler, and SoundFont 2.0 musical instrument files, allowing you to add effects and processing to existing samples.

When you open a musical instrument file, you'll notice some additions to the data window:



The left pane lists the banks, instruments, note ranges, and recorded samples in the instrument file. Click a bank or sample to select it in the waveform display.

Markers ( ) represent each sample in the waveform display.

Opening musical instrument files

You can open musical instrument files just like any other file type:

- Drag a musical instrument file to the workspace.
- From the **File** menu, choose **Open**, and then use the Open dialog to browse to the file you want to open.
- Double-click an instrument file in the Sound Forge Explorer window.

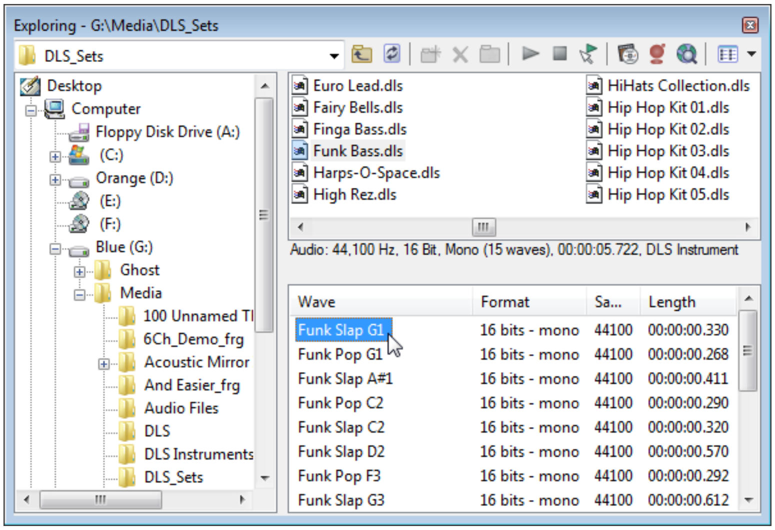
The following musical instrument file formats are supported:

- .dls: DLS level 1.0 and 2.0
- .sf2: SoundFont version 2.0
- .gig: GigaSampler/GigaStudio version 1/2/3

Important: *Compressed and encrypted GigaSampler/GigaStudio samples are not supported.*

If you want to open a sample's audio data, browse to an instrument file in the Explorer window. Click the down arrow next to the **Views** button ( ) and choose **Region View** from the menu.

Each wave is then displayed at the bottom of the Explorer window. You can double-click a wave to open it as a wave file:



Previewing samples

Playback for musical instrument files behaves slightly differently than playback in a normal data window.

- If no samples are selected, click **Play All** ( ) to play all samples in the data window. Click **Play** ( ) to play all samples from the cursor position to the end of the data window.
- If you have samples selected, click **Play All** ( ) to play all selected samples in the data window. Click **Play** ( ) to play all selected samples from the cursor position to the end of the data window.

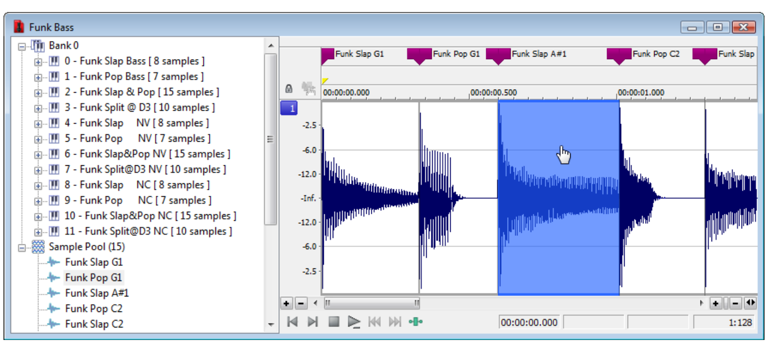
Selecting samples

Each sample in an instrument file is contained within an event in the data window. The data window selection determines which parts of the waveform will be processed.

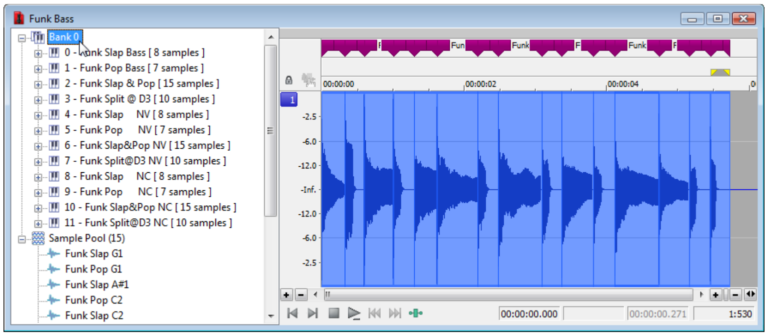
Event positions are locked. You cannot move events within the data window.

Effects and processes can be applied to individual events or multiple selected events.

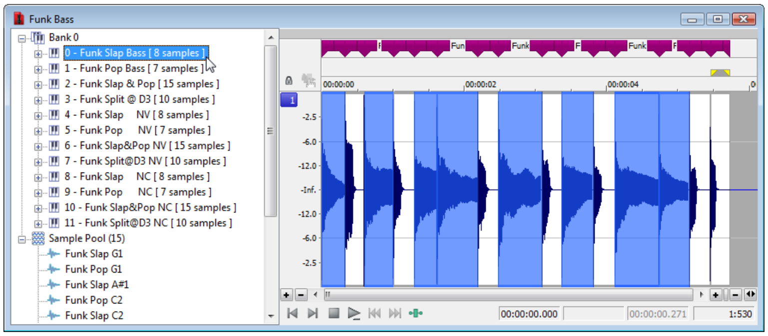
You can select an individual sample by clicking the event in the data window (hold Ctrl or Shift to select multiple samples):



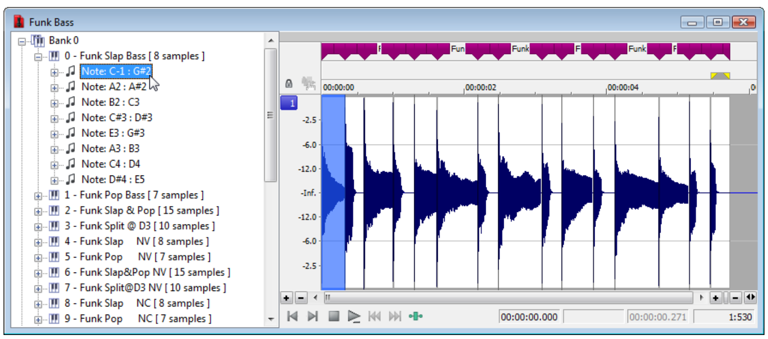
You can select all samples in a bank by clicking the bank in the left pane:



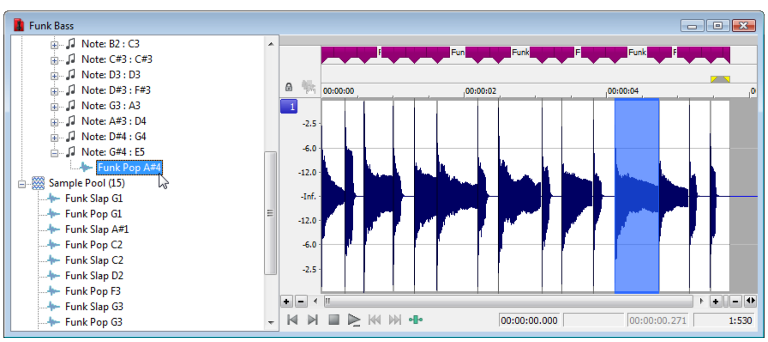
You can select all of an instrument's samples by clicking the instrument in the left pane:



You can select all samples within a note range by clicking the note range in the left pane:



You can select individual samples by expanding a note range or the Sample Pool list and clicking a sample in the left pane:



Processing selections

1. Select the samples you want to process. If no samples are selected, processing will be applied to all samples.
2. Choose a command from the **Process**, **Effects**, or **FX Favorites** menu.
3. Choose a preset from the **Preset** drop-down list or adjust the dialog controls as needed.

   For more information, see Applying presets on page 181.

> **Note:** *Plug-ins that can change the length of audio data (such as reverb or delay) will use* **Insert Tail Data** *mode. Tails will be added to the waveform, and audio to the right of the tail will be moved to accommodate the extra audio.*

4. Click the **Preview** button to hear the effects of your processing settings. Select the **Bypass** check box to hear the unprocessed signal.
5. Click the **OK** button to start processing.

   During processing, a progress meter is displayed at the bottom of the data window. You can cancel the operation at any time by clicking the **Cancel** button to the left of the progress meter, or you can press the Escape key.

> **Note:** *When applying an effect to a file via scripting, you can only specify the current time/channel-selection (DoMenu) or a new time/channel-selection (DoEffect). Event selection is not exposed to scripts.*
>
> *When processing musical instrument files, all events in the time/channel selection will be processed. If no selection exists or the entire file is selected (as when using the Batch Converter), all events will be processed.*

Editing samples

In addition to applying processing and effects, you can also edit the samples in an instrument file.

For example, if you need to remove a glitch or replace a sample in an instrument file, you can open an individual sample in a new editing window.

1. Select the sample you want to edit.
2. From the **Edit** menu, choose **Event**, and then choose **Edit** from the submenu.

   The selected sample is opened in a new window. If you selected multiple samples in step 1, each sample is opened in a separate window.

> **Tip:** *Press E (or right-click a sample and choose* **Edit** *from the shortcut menu) to open the edit window quickly.*

3. Perform edits as needed.

> **Notes:**
>
> - *If you want to replace sample data, you can clear the data in the edit window and paste data from another window.*
> - *Changes to bit depth, sample rate, or number of channels cannot be saved back to the original musical instrument file.*

4. From the **File** menu, choose **Save** to close the edit window and save your changes back to the musical instrument file.

   If you want to save the edited sample to a different format (other than a musical instrument file), you can use **Save As**.

Saving changes

After you're finished processing a musical instrument file, you can use the Save or Save As commands to save your changes to the original file or to a new file. *For more information, see Saving a file on page 72 and Using the Save As/Render As dialog on page 72.*

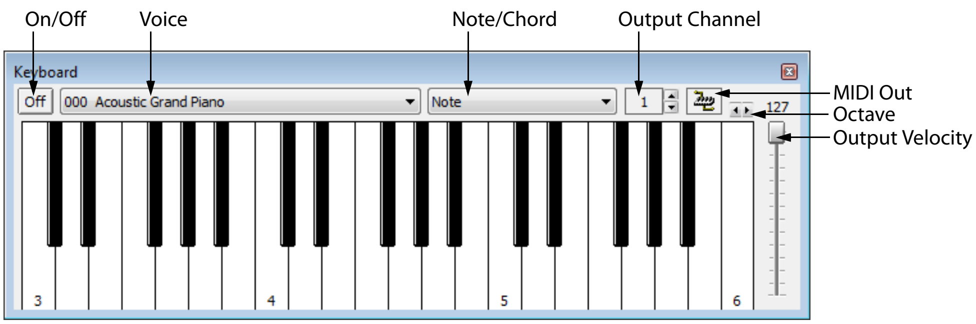
Musical instrument files must be saved to the original format.

Using the MIDI keyboard

With the MIDI keyboard, you can control internal/external synthesizers and samplers from the Sound Forge application. The MIDI keyboard can also be used to listen to the sounds on a synthesizer or in the synthesis section of the sound card.

Displaying the MIDI keyboard

To display the MIDI keyboard, choose **Keyboard** from the **View** menu. The keyboard can be resized, moved, or docked within the workspace.

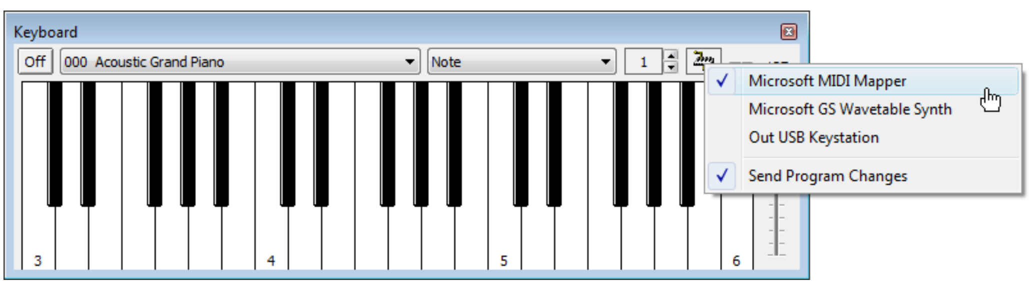


Turning on the MIDI keyboard

Clicking any key turns the keyboard on. If you do not hear any sound, verify that the output is connected to the MIDI Output device.

Configuring the MIDI keyboard output port and channel

1. Click the **MIDI Out** button ( ) and choose an output device from the menu.



2. Choose **Send Program Changes** from the menu if the keyboard will be used to choose instrument voices. A check mark appears adjacent to the command to indicate that this option is turned on.
3. Configure the MIDI input channel of the selected device to correspond to the keyboard's output channel.

> **Note:** *Most MIDI devices are configurable to accept MIDI commands on any channel.*

Troubleshooting the MIDI keyboard

If after configuration, the keyboard fails to produce sound, check the following items:

- Verify that the output velocity of the keyboard is set to a value greater than 100.
- Verify that the MIDI input channel in the sound module is set to the same channel as the keyboard.
- Verify that the device is configured to receive MIDI input.
- Verify the device output volume level.
- Verify external MIDI connections, if applicable.

Specifying instruments

1. Choose **Send Program Changes** from the **MIDI Out** button ( ) menu. A check mark appears adjacent to the command to indicate that this option is turned on. If this option is turned off, patches cannot be switched.
2. From the **Voice** drop-down list, choose the new voice and click any key.

Note: *Patch names are arranged as specified in the General MIDI Standard. For synthesizers not using the General MIDI convention, use the patch number instead of the instrument name.*

Generating chords

You can also generate chords instead of single notes by choosing a chord structure from the keyboard's **Note/Chord** drop-down list. Chords are generated using the specified note as the root of the chord or interval.

Setting up MIDI/SDS hardware

To use MIDI/SDS protocol with an external sampler that supports MIDI/SDS, you must install a MIDI card with MIDI input and output ports in the system.

1. Using a MIDI cable, connect the MIDI output port of the sampler to the MIDI input port of the MIDI card.
2. Connect the MIDI input port of the sampler to the MIDI output port of the card.

> **Note:** *This is the same configuration used to connect a MIDI keyboard to a computer for sequencing.*

Internal samplers do not require a MIDI card and MIDI cables; however, an open-loop protocol may be required when sending samples to an internal sampler. The sampler's documentation should specify the requirements for performing SDS transfers if the sampler supports this action.

Troubleshooting MIDI/SDS with open loop

Open-loop transfers, while not recommended for sending or receiving samples, can assist you in troubleshooting SDS hardware setup problems. If the Sampler Tool does not transfer data to (or from) the sampler, select the open-loop option and attempt single cable transfers. If open-loop transfers are successful, but closed-loop transfers are not, any of the following items may be the cause:

- The sampler does not support closed-loop transfers (handshaking).
- One or more of the MIDI cables or connections is faulty.
- The MIDI card is not receiving MIDI input (send) or sending MIDI output (receive). Interrupt conflicts are common for MIDI input.

Setting up SCSI/SMDI hardware

To use the SCSI/SMDI protocol with an external sampler that supports the SCSI/SMDI protocol under Windows 98SE, Windows Me, Windows 2000, or Windows XP, only a compatible SCSI adapter is needed. The computer and sampler must be powered-down prior to connecting or disconnecting SCSI cables to prevent damage to the computer and/or hardware.

Troubleshooting SCSI/SMDI

A brief description of some common problems encountered with SCSI and samplers follows.

Conflicting SCSI IDs

When connecting devices on a SCSI chain, each device must have a unique device identifier (ID). SCSI allows for up to eight unique ID values, numbered 0 to 7. Typically, device ID 7 is used for the internal SCSI controller card, leaving ID 0 through 6 for other devices.

> **Note:** *The ID of a bootable SCSI hard drive must be set to 0.*

The following table describes a typical SCSI configuration:

ID	Devices
0	Hard Drive
1	CD-ROM/DVD-ROM Drive
2-6	Samplers
7	SCSI Controller Card

Periodic transfer failures

Messages such as “The SCSI Device is not responding” or “A problem was encountered while transferring the sample” may indicate a problem with a SCSI bus.

1. From the **Tools** menu, choose **Sampler**. The Sampler dialog is displayed.
2. Click **Configure**. The Sampler Configuration dialog is displayed.
3. Repeat the selection of the SCSI host. This causes a series of SCSI commands to be executed that may settle the bus.

   If the problem persists, power down and restart all equipment.

Sampler is recognized but does not transfer reliably

The following items are possible causes of unreliable SCSI transfers.

Synchronous transfer mode

Select samplers (the Kurzweil K2000 among them) do not operate properly if there is a SCSI device set to synchronous transfer mode on the same SCSI chain. SCSI hard drives and CD-ROMs or DVD-ROMs often have the option of using a synchronous transfer mode. If there is a host versus device synchronous transfer option, select the host option. Refer to the SCSI device’s documentation for more information.

SCSI termination

If the SCSI chain is not properly terminated, unreliable SCSI transfers may be experienced. Refer to the SCSI card and SCSI device documentation for more information.

Long or faulty SCSI cables

SCSI cables that are very long or not properly shielded may not operate reliably. In addition, do not use cables that are not certified SCSI cables.

Adaptec 1540/1542CF does not recognize a sampler

If the Adaptec 1540/1542CF does not recognize the sampler, a change may be required in the configuration of the Adaptec controller. Some samplers do not operate when the **Reset SCSI Bus at Power-On** option of the Adaptec controller is turned on. This is the default operation for the 1540/1542CF and must be turned off to allow the system to work with the sampler.

> **Note:** *Turning off the* **Reset SCSI Bus at Power-On** *option may keep other devices on the SCSI chain from resetting correctly when using the system’s soft boot feature. Other systems may freeze temporarily. To guarantee that devices are reset when rebooting with this option turned off, use the system’s reset button or power-down and up to reset the system.*

SCSI/SMDI-compatible menu is not displayed under Windows 98SE and Windows Me

Verify proper SCSI termination and check for multiple devices on the SCSI chain using the same SCSI ID. If this fails to solve the problem, Adaptec SCSI card users may need to update the system’s mini-port drivers. Adaptec has a series of updated mini-port drivers available for Windows 98SE and Windows Me on the Web.

After you download the file, you must create a temporary directory or folder on the system and run the WIN95MPD.EXE program. Follow the directions in the readme.txt file to update the drivers for the Adaptec SCSI card.

Chapter 18

MIDI synchronization

The Musical Instrument Digital Interface (MIDI) is a set of commands that allow music software and hardware to communicate. MIDI is most often used for sending commands such as **Play Middle C Now**, but can also be used to send information such as **Current Time is: 00:00:01:23 SMPTE**, or even digital sound data. *For more information, see Sampling on page 259.*

The most common way to use MIDI is to have a master device (such as a MIDI sequencer) to generate MIDI commands to a slave device (such as a synthesizer, which plays a note when instructed). If both were in separate hardware devices, you would run a MIDI cable from the sequencer's MIDI out port to the synthesizer's MIDI in port.

Generating MIDI timecode

From the Options menu, choose **MIDI In/Out**, and then choose **Generate MIDI Timecode** from the submenu if you want to generate MIDI timecode (MTC) when you click **Play** (▶).

MIDI timecode (MTC) is a standard timecode that most applications and some hardware devices will use to synchronize themselves.

> **Note:** *You can specify a MIDI output port on the MIDI/Sync tab in the Preferences dialog.* *For more information, see MIDI/Sync tab on page 338.*

Triggering from MIDI timecode

From the Options menu, choose **MIDI In/Out**, and then choose **Trigger from MIDI Timecode** from the submenu if you want to trigger playback or recording by receiving timecode from another device.

When this option is selected, dialogs that specify MIDI triggers will also accept input from the MIDI input port, allowing easy entry of MIDI note and controller values. When this option is not selected, the MIDI triggers, Regions List triggers, and Playlist triggers specified will be ignored.

> **Note:** *You can specify a MIDI input port on the MIDI/Sync tab in the Preferences dialog.* *For more information, see MIDI/Sync tab on page 338.*

Triggering playback with MIDI timecode

You can use MIDI timecode to trigger Sound Forge playback from another device.

1. From the Options menu, choose **Preferences**, and click the MIDI/Sync tab.
2. On the MIDI/Sync tab, choose the trigger device from the **Input** drop-down list and click the **OK** button.
3. From the Options menu, choose **MIDI In/Out**, and then choose **Trigger from MIDI Timecode** from the submenu to enable MIDI input.
4. From the View menu, choose **Toolbars**. Select the **Regions/Playlist** check box and click the **OK** button. The timecode is displayed in the Regions/Playlist toolbar when you start your MIDI device.
5. Create a region that includes the sound data that you want to trigger. *For more information, see Inserting a region on page 121.*
6. Add your region to the Playlist. *For more information, see Adding regions to the Playlist on page 131.*
7. In the Playlist window, choose **SMPTE: Play at Time** from the **Trigger** drop-down list and enter the time at which you want to start playback in the **SMPTE time** box.

Triggering recording with MIDI timecode

You can use MIDI timecode to trigger Sound Forge recording to another device.

For more information, see Recording audio automatically on page 148.

Note: *If* **Trigger from MIDI Timecode** *is not selected when you open the Record dialog, Sound Forge Pro software will turn it on temporarily when you choose the* **Automatic: MIDI Timecode** *recording mode to allow recording and turn it off again when you close the Record dialog.*

Pre-queuing data for synchronization

From the Options menu, choose **MIDI In/Out**, and then choose **Pre-Queue for MIDI Timecode** from the submenu if you want to open the wave device and preload data from the next region to be played in the Playlist.

Pre-queuing helps ensure that audio will begin playing the moment the designated SMPTE time is detected by Sound Forge software when triggering from MIDI timecode (MTC).

Note: *This option is suspended when any other audio command is used such as* **Play**, **Stop**, *or* **Record**.

Using MIDI triggers

You can use MIDI triggers to control Sound Forge Pro functions using MIDI commands from external devices such as a MIDI keyboard or sequencer.

Configuring an internal or external MIDI controller

1. From the Options menu, choose **Preferences**, and then click the MIDI/Sync tab.
2. On the MIDI/Sync tab, choose the trigger device from the **Input** drop-down list and click the **OK** button.
3. From the Options menu, choose **MIDI In/Out**, and then choose **Trigger from MIDI Timecode** from the submenu to enable MIDI input.
4. Use the MIDI Triggers dialog to configure the triggers you want to use. *For more information, see Assigning Sound Forge Pro events to MIDI triggers on page 272.*

Assigning Sound Forge Pro events to MIDI triggers

1. From the Options menu, choose **MIDI In/Out**, and then choose **Trigger from MIDI Timecode** from the submenu to enable MIDI input.
2. From the Options menu, choose **MIDI Triggers** to display the MIDI Triggers dialog.
3. Select the Sound Forge Pro function that you want to trigger in the **Event** list. *For more information, see MIDI trigger events on page 273.*
4. Click the radio button that corresponds to the type of trigger you want to use for the selected event:

Item	Description
None	Click to assign no MIDI trigger to the selected event or to remove an existing trigger.
Note	Click if you want to trigger the selected event with a MIDI note. Specify the MIDI channel to which the trigger is assigned in the **Channel** box, and specify the musical note that will trigger the event in the **Note** box.
Controller	Click if you want to trigger the selected event using a MIDI controller. Specify the MIDI channel to which the controller is assigned in the **Channel** box. Use the **Controller** box to specify which controller will trigger the event and specify a value in the **Value** box.

Tip: *You can automatically enter the values in the* **Channel**, **Note**, **Controller**, *and* **Value** *boxes. Select the* **Enable MIDI Input Sync/Trigger** *check box and press a key or controller on your MIDI device.*

5. Click the OK button.

Note: *MIDI triggers are different from triggers in the Playlist and Regions List. When using triggers in the Playlist, Regions List, or MIDI Triggers dialog, be aware that they can interact with each other to create unexpected results. Sound Forge software first looks at the MIDI Triggers, then the Regions List, and then the Playlist when determining what to do when a MIDI command is detected.*

MIDI trigger events

Item	Description
Enable/Disable Triggers	Click to assign no MIDI trigger to the selected event or to remove an existing trigger.
Play All	Plays the entire sound file from beginning to end, regardless of cursor position, selection, or playlist.
Play	Plays the sound file in the current playback mode.
Pause	Pauses playback and leaves the cursor at its current position.
Stop	Stops playback and returns the cursor to its position prior to playback.
Record Start/Stop	Starts or stops recording when the Record dialog is open.
Start	Moves the cursor to the beginning of the active data window.
End	Moves the cursor to the end of the active data window.
Next Window	Makes the next data window on the Window menu active.
Previous Window	Makes the previous data window on the Window menu active.
Loop Mode On/Off	Turns looped playback mode on or off.
Play Normal	Starts playback in Normal mode: • If there is no selection, playback occurs from the cursor to end of file. • If there is a selection, playback occurs from the beginning of the selection to the end of the selection.
Play as Sample	Starts playback in Sample mode: • If the file contains loops, the loops will repeat as many times as specified on the Edit Sample dialog. Use this to listen to a sound file as it would sound when played by a sampler. • If the file does not contain any loops, the file will be played once from beginning to end.
Play as Cutlist	Starts playback of the current cutlist.
Loop Mode On	Turns looped playback mode on.
Loop Mode Off	Turns looped playback mode off.
Preview Cut/Cursor	Plays the data before and after the current selection. This command lets you preview the result of a **Cut** or **Clear** operation without altering the file.
Pre-Roll to Cursor	Plays the data before the cursor position for the default pre-roll length (specified on the Other tab in the Preferences dialog).
Forward Slow	Moves the cursor through the data window in the current zoom ratio. For example, if the zoom ratio is 1:1024, this will move the cursor forward 1024 samples.
Forward Medium	Moves the cursor through the data window 25 times faster than **Forward Slow**.
Forward Fast	Moves the cursor through the data window 75 times faster than **Forward Slow**.
Rewind Slow	Moves the cursor through the data window in the current zoom ratio. For example, if the zoom ratio is 1:1024, this will move the cursor back 1024 samples.
Rewind Medium	Moves the cursor through the data window 25 times faster than **Rewind Slow**.
Rewind Fast	Moves the cursor through the data window 75 times faster than **Rewind Slow**.
Insert Marker	Places a marker at the current cursor position.
Create Region	Creates a region quickly using the currently selected data.

Item	Description
Mark In	Sets the beginning of a selection at the current cursor position. The data between this point and the Mark Out position will be selected.
Mark Out	Sets the end of a selection at the current cursor position. The data between this point and the Mark In position will be selected.
Zoom Level In	Zooms in vertically.
Zoom Level Out	Zooms out vertically.
Zoom Time In	Zooms in horizontally.
Zoom Time Out	Zooms out horizontally.
Pan Down	Pans down vertically. This is useful when the entire amplitude range is not visible in the data window.
Pan Up	Pans up vertically. This is useful when the entire amplitude range is not visible in the data window.
Go To Next Marker	Moves the cursor to the next marker.
Go To Previous Marker	Moves the cursor to the previous marker.
Go To Next Frame	Moves the cursor to the next video frame.
Go To Previous Frame	Moves the cursor to the previous video frame.
Go To Next Sample	Moves the cursor to the next sample.
Go To Previous Sample	Moves the cursor to the previous sample.

Using the MIDI keyboard

From the View menu, choose **Keyboard** to open or close the MIDI Keyboard window. The keyboard allows you to send MIDI note on and MIDI note off commands to your sound card or sampler.

With the Sound Forge Pro MIDI keyboard, you can control internal or external synthesizers and samplers from Sound Forge Pro software. For example, you might want to test a sound in a sampler after sending (downloading) it from Sound Forge Pro software. You can also use the MIDI keyboard to listen to the different sounds on a synthesizer or in the synthesis section of your sound card.

Opening the MIDI keyboard

From the View menu, choose **Keyboard** to open or close the MIDI Keyboard window.

Resizing the MIDI keyboard

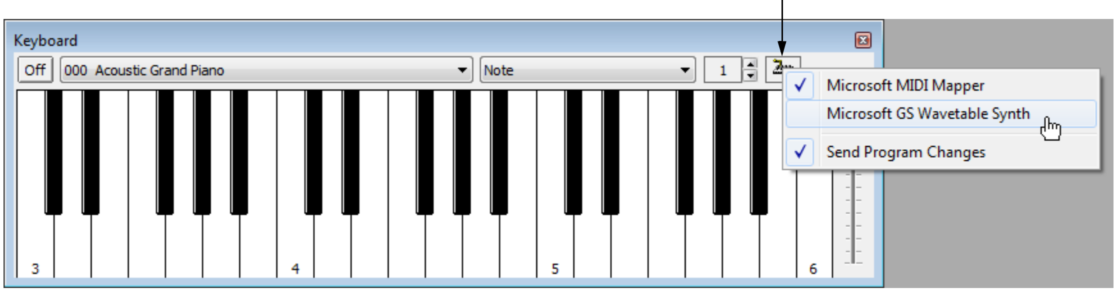
Like any other toolbar or shortcut menu window in Sound Forge Pro, you can resize the MIDI Keyboard window, move it, or dock it to any side of the Sound Forge Pro window.

When the cursor is over any edge of the window, the cursor is displayed as a . Drag to resize. Depending on how you resize the keyboard, the keys will be arranged horizontally or vertically.

Configuring the output device and channel

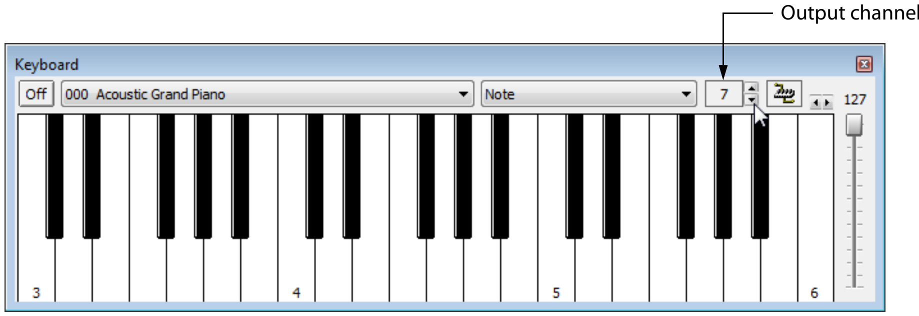
1. To select a MIDI device, click the **MIDI Out** button and choose a device from the menu:

MIDI Out button



2. Select the **Send Program Changes** command to select instruments in the device from the keyboard. Program changes are sent only when a key on the MIDI keyboard is clicked.

3. Click the up and down arrows to select the MIDI channel you want to use for the selected device. Most MIDI devices are configurable to accept MIDI commands on any channel.



Selecting instruments

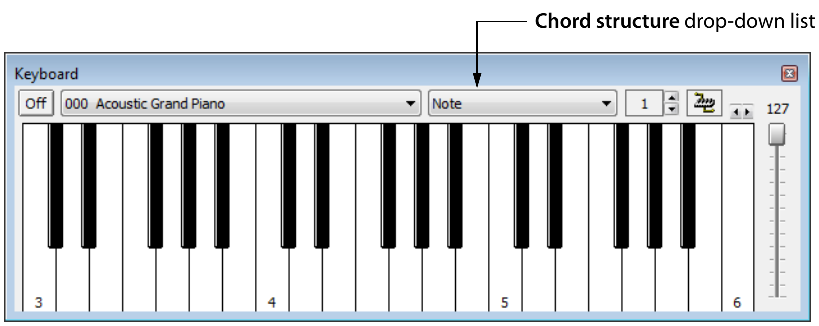
The Instrument drop-down list contains instrument patches listed in the General MIDI standard. Choose an instrument from the drop-down list and click a key on the keyboard to play the patch.

> **Notes:**
>
> - *The* **Send Program Changes** *command must be selected if you want to be able to select instruments. Program changes are sent only when a key on the MIDI Keyboard is clicked.*
> - *If your synthesizer does not use the General MIDI convention, choose a patch number instead of an instrument name.*

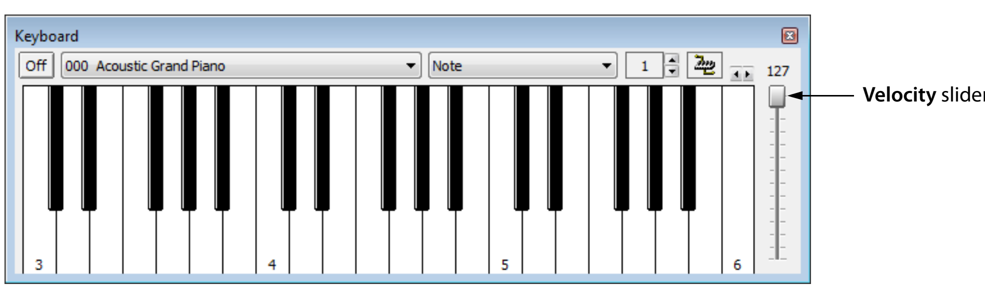
Playing the keyboard

Click the keys to play them:

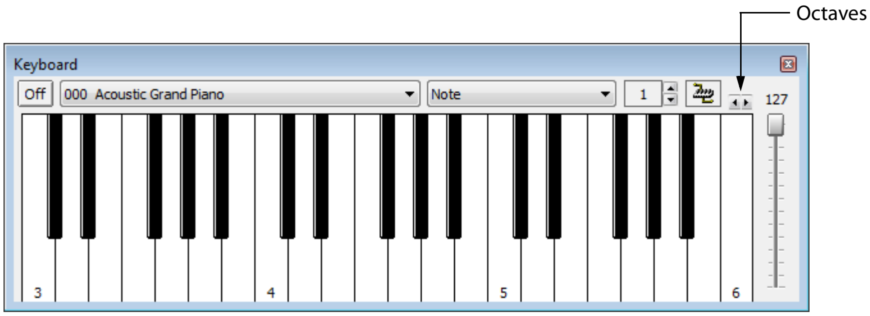
- Choose **Note** from the **Chord structure** drop-down list to play a single note, or choose a chord structure or interval to play a chord (the key you click will be the chord's root).



- Drag the **Velocity** slider to set the MIDI output velocity.



- Click the arrow buttons to change the range of the MIDI keyboard to display higher or lower octaves.



Troubleshooting the MIDI keyboard

If, after selecting the correct device, you still can't hear anything when you play on the keyboard, check the following:

- Set the MIDI output velocity to a high value (above 100).
- Set the MIDI input channel in your sound module to the same channel as the MIDI keyboard channel. Also, make sure the device is set to receive MIDI input.
- Check the device output volume level: is the mixer level for your sound card or the output volume of the sampler and speakers at a high enough setting?
- For external devices, check your MIDI cables.

Looping

Sound Forge® Pro software is an excellent tool for creating loops and provides the perfect compliment to the revolutionary ACID® line of loop-based music creation tools.

Creating loop regions in files is useful only when you intend to transfer the files to a hardware sampler that supports the loop regions.

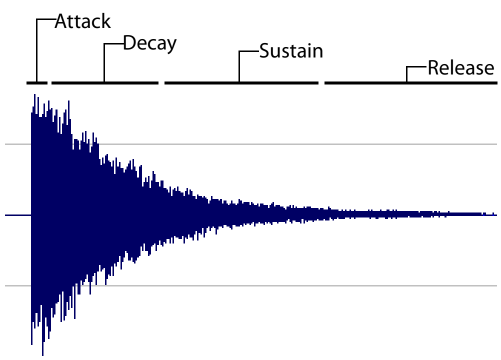
Loops

A loop is a sample or region in an audio file that is repeated during playback. Samples are finite and frequently very short in length. Therefore, they must be repeated (or looped) to create longer or sustaining sounds.

> **Note:** *Loops can also be used to repeat entire sections of music, although the Playlist is better suited to this purpose. For more information, see* *Using the Playlist on page 130*.

Sustaining and release loops

A sound envelope contains four elements: attack, decay, sustain, and release.

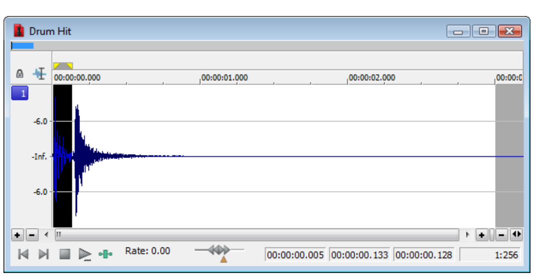


Typically, the sustain portion of the envelope is looped to lengthen the duration of a sound. This is referred to as the sustaining loop.

While sustaining loops are useful, it is frequently necessary to create a second loop, taken from later in the envelope. This allows you to reproduce longer, more complex sounds, such as a piano chord struck with the sustain pedal depressed. This second type of loop is referred to as the release loop.

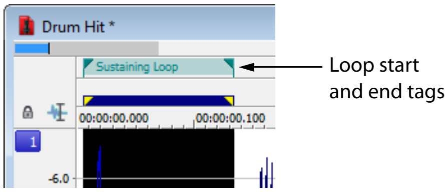
Creating a sustaining loop

1. Open the Drumhit.pca file and create a selection containing the snare hit at the beginning of the waveform.

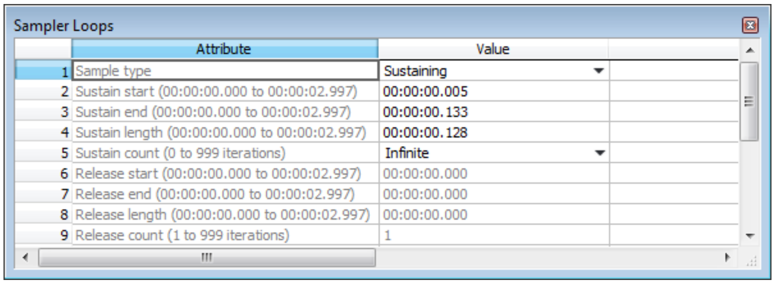


2. With the **Loop Playback** button ( ) selected in the transport bar, click the **Play Normal** button ( ) on the playbar to preview the loop.

3. From the **Insert** menu, choose **Sample Loop** (or press Alt+L). The data window displays the appropriate tags in the ruler to specify the loop's start and end points. The **Play as Sample** button ( ) appears on the playbar.



4. Right-click the sustaining loop and choose **Edit Sample Loop** from the shortcut menu (or press Alt+Shift+L). The Sampler Loops window appears.


Sampler Loops

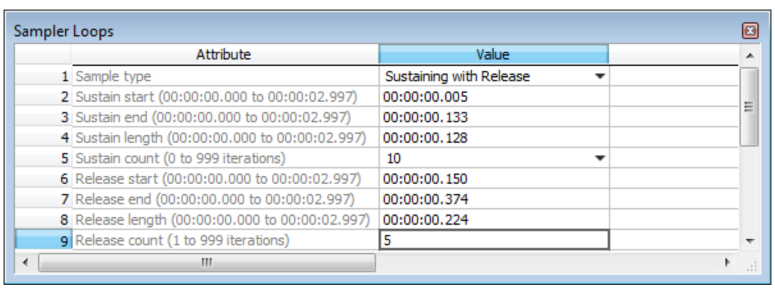
	Attribute	Value
1	Sample type	Sustaining
2	Sustain start (00:00:00.000 to 00:00:02.997)	00:00:00.005
3	Sustain end (00:00:00.000 to 00:00:02.997)	00:00:00.133
4	Sustain length (00:00:00.000 to 00:00:02.997)	00:00:00.128
5	Sustain count (0 to 999 iterations)	Infinite
6	Release start (00:00:00.000 to 00:00:02.997)	00:00:00.000
7	Release end (00:00:00.000 to 00:00:02.997)	00:00:00.000
8	Release length (00:00:00.000 to 00:00:02.997)	00:00:00.000
9	Release count (1 to 999 iterations)	1

5. In the **Sustain count** box, choose **Custom** and then type a value of 10. Press Enter.
6. Click the **Play as Sample** button ( ) on the playbar. The looped snare selection repeats ten times before the cymbal crash.

Creating a sustaining loop with a release loop

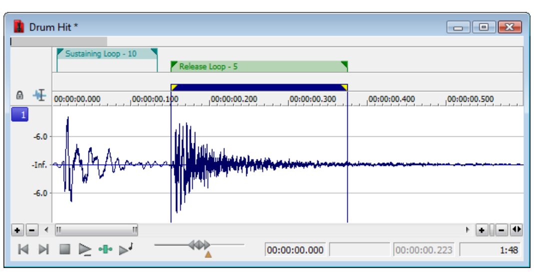
To add a release loop to the sustaining loop created in the previous procedure, you must insert another sample loop in the Sampler Loops window and rearrange the loop tags in the data window.

1. Right-click the sustaining loop and choose **Edit Sample Loop** from the shortcut menu (or press Alt+Shift+L). The Sampler Loops window appears.
2. In the **Sample type** box, choose **Sustaining with Release**. A release loop is created for the same length as the sustaining loop.
3. Edit the length of the new release loop to contain the cymbal crash.
4. Type a value of 5 in the **Release count** box. Both loops (sustaining and release) are configured.


Sampler Loops

	Attribute	Value
1	Sample type	Sustaining with Release
2	Sustain start (00:00:00.000 to 00:00:02.997)	00:00:00.005
3	Sustain end (00:00:00.000 to 00:00:02.997)	00:00:00.133
4	Sustain length (00:00:00.000 to 00:00:02.997)	00:00:00.128
5	Sustain count (0 to 999 iterations)	10
6	Release start (00:00:00.000 to 00:00:02.997)	00:00:00.150
7	Release end (00:00:00.000 to 00:00:02.997)	00:00:00.374
8	Release length (00:00:00.000 to 00:00:02.997)	00:00:00.224
9	Release count (1 to 999 iterations)	5

5. Click the **Play as Sample** button ( ) in the playbar. The entire file plays with the snare hit repeated ten times followed by the cymbal crash five times.



Looping techniques

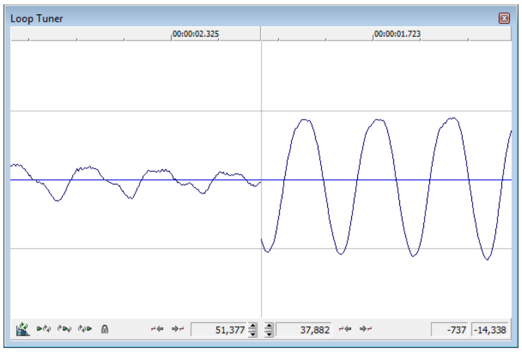
Depending upon the source material, creating a natural-sounding loop can be a difficult task. Many factors beyond your control may produce distracting pops and glitches, thereby calling unwanted attention to the loop. Although looping skill is largely the product of practice and experimentation, there are some guidelines to consider.

Match endpoint amplitudes

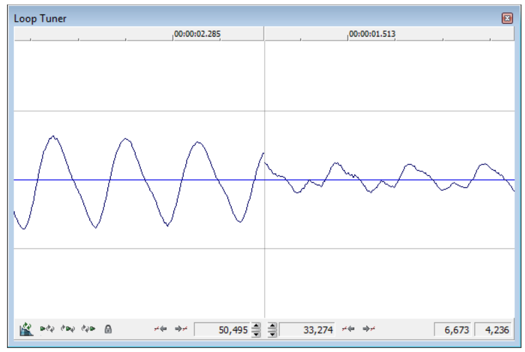
One of the easiest ways to minimize the occurrence of glitches when creating loops is to select loop endpoints that have an amplitude of zero. These points are known as zero-crossings.

Match endpoint waveform slope

Another technique for reducing loop glitches is to avoid matching loop endpoints where the waveform slope does not match. If the waveform slope changes drastically, a pop plays when the sample is looped.



Non-matching slope



Matching slope

Match endpoint sound levels

The overall amplitude (or loudness) approaching the loop's endpoints should be as similar as possible to prevent distracting glitches. Unfortunately, it is frequently difficult to avoid this problem, particularly with rapidly decaying source material.

Avoid very short loops

If the loop is shorter than ~50 ms (1/20 Hz), the pitch of the loop may not equal the sample pitch. Pitch-tuning a loop is accomplished by creating short loops with a length equal to 1/frequency. For example, a sample of pitch 440 Hz corresponds to A5 on the keyboard, meaning the loop can be pitch-tuned 2.27 ms. However, pitched loops do not sound like the original sample.

Editing loops

The loop you initially create in any situation is rarely perfect. Frequently, loops require some degree of editing before they are usable. Once you create a loop, you can quickly edit its beginning and end (and subsequently its length) by dragging the markers to a new location.

Editing a loop without the Loop Tuner

After you create a loop, you can quickly edit its beginning and end (and subsequently its length) by dragging the markers to a new location. However, this method frequently does not provide the control required to create seamless loops. In this case, you should edit the loop using the Loop Tuner.

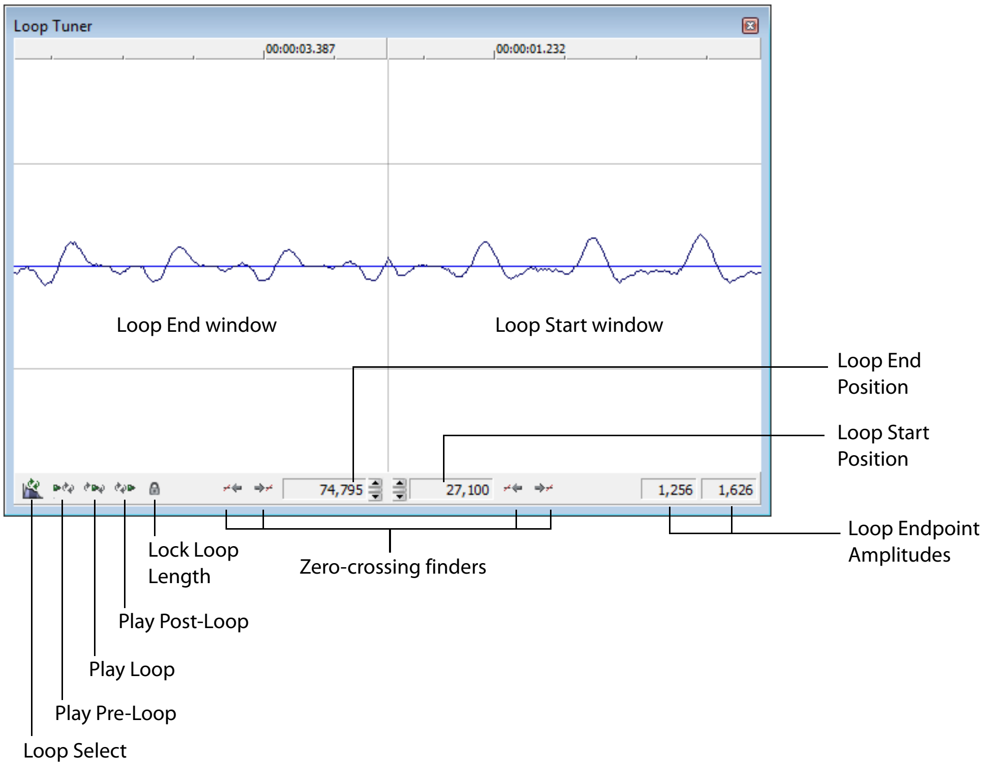
Editing a loop with the Loop Tuner

The Loop Tuner allows you to precisely edit loop points in order to prevent distracting audio glitches. This is accomplished by greatly magnifying the waveform and displaying the loop tags in relation to one another. You can also use the Loop Tuner to adjust the starting and ending points of a loop (or selection) to create smooth transitions.

The left side of the Loop Tuner window displays the end of the loop, while the right side displays the start of the loop. This arrangement allows you to fine-tune loops by viewing a graphical representation of the junction between the end and the start of a loop.

Displaying the Loop Tuner

1. Open the Loop.pca file. A sustaining loop appears in the data window.
2. From the **View** menu, choose **Loop Tuner**. The Loop Tuner appears and displays the waveform of the file's loop.



Playing loops using the Loop Tuner

The Loop Tuner contains three playback buttons: **Play Pre-Loop**, **Play Loop**, and **Play Post-Loop**.

	Play Pre-Loop Starts playback of a file from its beginning and stop playback at the loop start tag.
	Plays Loop Starts playback of the looped region. The number of times the loop plays is dependent upon the Loop count value in the Edit Sample dialog. Use this button to audition the loop for pops and glitches. *For more information, see* *Creating a sustaining loop on page 277*.
	Play Post-Loop Starts playback immediately following the loop end point and continue through the end of the file.

Tip: *You can use the* **Play as Sample** *button ( ) in the playbar to audition the entire sample with configured loops.*

Switching between the sustain and release loops

When working with a file that contains sustain and release loops, you can quickly toggle between the loops by clicking the **Loop Select** button ( ).

When working with a file containing two loops, this button indicates which loop is active.

- A selection in the middle of the **Loop Select** button icon( ) indicates that the sustaining loop is active.
- A selection at the end of the **Loop Select** button icon ( ) indicates that the release loop is active.

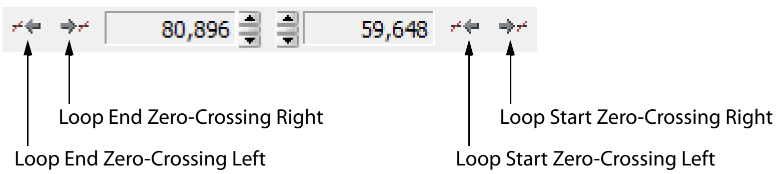
Viewing loop amplitude

The sample amplitude at the loop's start and end points appears in the lower-right corner of the Loop Tuner.

Although it is dependent on the specific waveform, a good rule of thumb is that the closer these two amplitude values are, the more natural the resulting loop sounds.

Finding zero-crossings

The Loop Tuner's zero-crossing finders are used to locate zero-crossings adjacent to the current loop tag location.



The Loop Tuner contains two zero-crossing finders for each of the loop points.

- The left button in each pair locates the zero-crossing to the left of the current location.
- The right button in each pair locates the zero-crossing to the right of the current location.

To use the finders, click the desired button. By experimenting with different locations and repositioning the start and end points, you can create seamless loops. You can also configure the zero-crossing finders to locate positive slope crossings, negative slope crossings, or all zero-crossings.

Configuring the zero-crossing finders

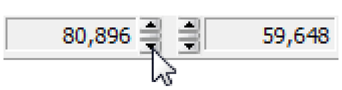
1. From the **Options** menu, choose **Preferences**. The Preferences dialog is displayed.
2. Click the **Editing** tab.
3. From the **Snap to zero-crossing slope** drop-down list, choose the desired slope and click **OK**.

Fine-tuning loop points

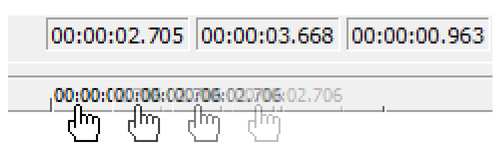
You can use the Loop Tuner to fine-tune loop points in three ways:

- To move loop points by small amounts, use the **Loop Start Position** and **Loop End Position** arrow spinners. Clicking the up or down arrow increments the loop point by one sample.
- To move loop points by larger amounts, use the mouse to drag the spinner up or down.
- To move loop points by very large amounts, use the mouse to drag the ruler at the top of the **Loop Start** or **Loop End** display.

Click the arrow spinners to move by one sample



Drag the ruler to move by large amounts



Locking loop length

The **Lock Loop Length** button () allows you to freely move the start and end points of a loop without altering its length. When the button is selected, any editing that moves a loop point affects both loop points, thereby keeping the loop length constant.

Clicking the **Lock Loop Length** button a second time turns this feature off and allows loop points to be edited independently with no regard for the loop's original length.

> **Tip:** *The* **Lock Loop Length** *button has the same function as the* **Lock Loop/Region Length** *command on the Options menu.*

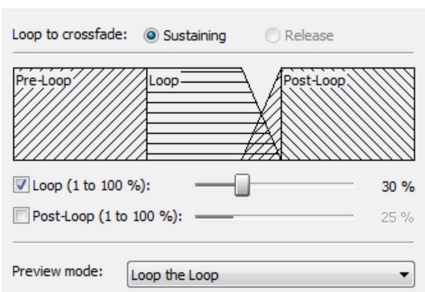
Crossfading loops

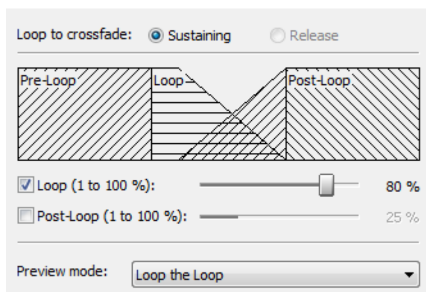
You can use the Crossfade Loop tool to loop audio from difficult source material. It allows you to crossfade the end of a loop with the beginning of the loop in order to create a smoother, more natural-sounding transition. In addition, you can configure the Crossfade Loop tool to crossfade the beginning of the audio loop with the beginning of the post-loop audio on the loop's final pass. This smooths the occasionally awkward transition from looped to non-looped audio.

> **Tip:** *Use the Loop Tuner before applying the* **Crossfade Loop** *command to match the loop ends as well as possible.*

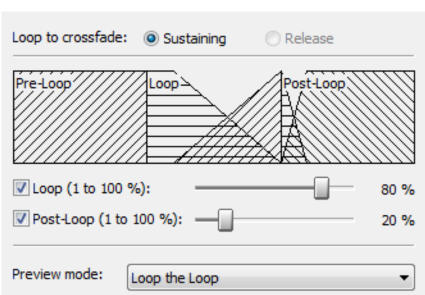
Using the Crossfade Loop tool

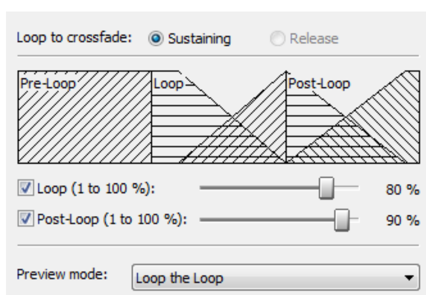
1. Open the Loop.pca file. A sustaining loop appears in the data window.
2. From the **Tools** menu, choose **Crossfade Loop**. The Crossfade Loop dialog is displayed.
3. Drag the **Loop** slider to configure the percentage of the loop to be crossfaded.





4. If desired, select the **Post-Loop** check box and drag the slider to configure the percentage of the loop to be crossfaded into the post-loop audio.





5. From the **Preview mode** drop-down list, specify how the **Preview** button operates: **Loop the Loop**, **Play Loop through Post-Loop**, or **Play as One Shot**.
6. Preview and tune the crossfade until you cannot detect the loop transitions.
7. Click **OK**.

Creating loops for ACID software

The Sound Forge application is an excellent tool for creating and editing loops to be imported into any of the ACID family of products. You can create three different types of files for ACID use:

- One-shot file
- Loop file
- ACID 3.0 or later beatmapped file

Creating an ACID one-shot file

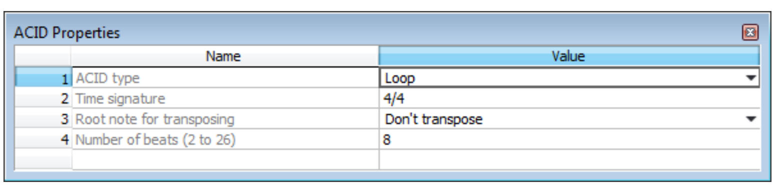
One-shots are files that do not stretch with tempo or change pitch to match the key of the ACID project. This behavior makes one-shots particularly suited for audio such as cymbal crashes, sound effects, and short vocal lines.

1. Open the Voiceover.pca file.
2. Create a selection containing the "Wow" and drag it to the workspace. A new data window is created containing the "Wow" audio data.
3. From the **View** menu, choose **Metadata**, and then choose **ACID Properties** from the submenu. The ACID Properties window appears.
4. Select **One-Shot** from the **ACID type** drop-down list.
5. From the **File** menu, choose **Save As** and save the file with a descriptive name.

Creating an ACID loop file

Loops are musical building blocks and are by far the most common type of file used in ACID software. Loops stretch with an ACID project's tempo and can be configured to change pitch.

1. Open the Voiceover.pca file.
2. Create a selection containing the "And easier" and drag it to the workspace. A new data window is created containing the "And easier" audio data.
3. From the **View** menu, choose **Metadata**, and then choose **ACID Properties** from the submenu. The ACID Properties window appears.
4. Select **Loop** from the **ACID type** drop-down list. The **Time signature**, **Root note for transposing**, and **Number of beats** boxes are added to the ACID Properties window.



5. Choose one of the following options for the **Root note for transposing** box:

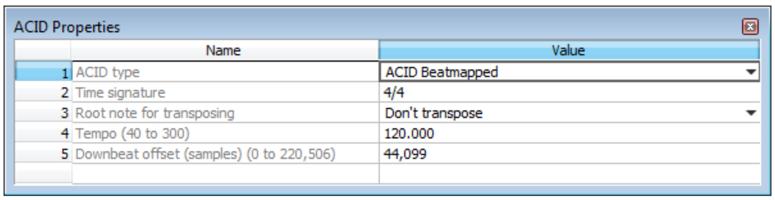
If	Then
The loop should be transposed when inserted in an ACID project	Choose it's root note from the **Root note for transposing** drop-down list.
The loop should not be transposed in an ACID project	Choose **Don't transpose** from the drop-down list.

6. Specify the length of the loop in beats in the **Number of beats** box. The default value is 4.
7. From the **File** menu, choose **Save As** and save the file with a descriptive name.

Creating an ACID beatmapped file

ACID beatmapped files can change tempo and pitch to match an ACID project. You must specify the file's original tempo and root note for transposing upon configuration. If you do not specify these values, no tempo or key changes occur. Beatmapped files are typically used in ACID software version 3.0 or later for extended vocal tracks or other long audio files that do not loop.

1. Open the Voiceover.pca file and select the entire waveform.
2. From the **View** menu, choose **Metadata**, and then choose **ACID Properties** from the submenu. The ACID Properties window appears.
3. Select **ACID Beatmapped** from the **ACID type** drop-down list. Complete the information for an ACID beatmapped file:
   - Select a value from the **Root note for transposing** drop-down list so that ACID software can transpose the file to match the project key. Select **Don't transpose** from this list to keep the key from being changed.
   - Specify the file's original tempo (40-300 bpm) in the **Tempo** box so that ACID software can stretch the file to match the project tempo. *For more information, see Setting loop tempo on page 287*.
   - Enter a value in the **Downbeat offset (samples)** box to indicate the location of the first downbeat.



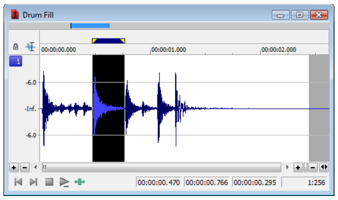
4. From the **File** menu, choose **Save As** and save the file with a descriptive name.

Editing loops for ACID software

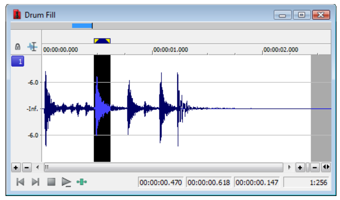
You can use a number of tools to prepare audio for use in ACID software.

Halving or doubling a loop

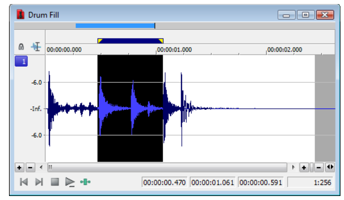
These commands allow you to quickly change the size of a selection.



Loop



Half loop



Double loop

Halving a loop

From the **Edit** menu, choose **Selection**, and choose **Halve** from the submenu.

> **Tip:** *You can also click the* **Halve Selection** *button ( ) on the Navigation toolbar or press the semicolon key (;).*

Doubling a loop

From the **Edit** menu, choose **Selection**, and choose **Double** from the submenu.

> **Tip:** *You can also click the* **Double Selection** *button ( ) on the Navigation toolbar or press the apostrophe key (').*

Shifting a selection left or right

The shift selection commands allow you to quickly create a new selection adjacent to the current selection while maintaining the size of the original.

Creating a new selection to the left of the current selection

From the **Edit** menu, choose **Selection**, and choose **Shift Left** from the submenu.

> **Tip:** *You can also click the* **Shift Selection Left** *button ( ) on the Navigation toolbar or press <.*

Creating a new selection to the right of the current selection

From the **Edit** menu, choose **Selection**, and choose **Shift Right** from the submenu.

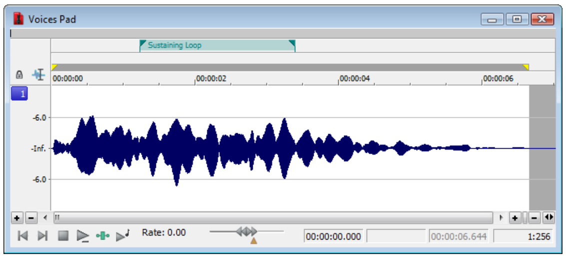
> **Tip:** *You can also click the* **Shift Selection Right** *button ( ) on the Navigation toolbar or press >.*

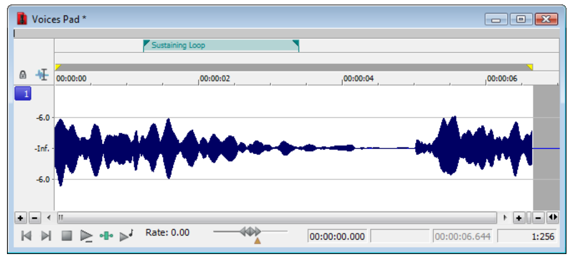
Rotating audio

You can move the beginning of a loop to the end, or the end of a loop to the beginning by rotating the audio. From the **Process** menu, choose **Rotate Audio**.

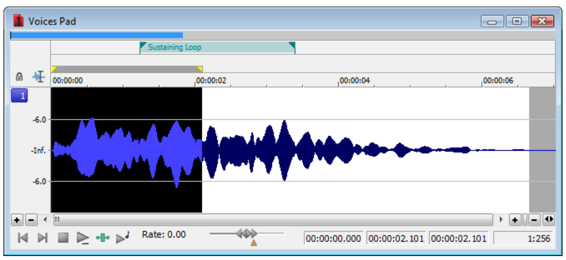
> **Notes:**
>
> - *You can also click the* **Rotate Audio** *button ( ) on the Process toolbar or press : (colon).*
> - *If the selected audio does not originate from the start or end of a loop,* **Rotate Audio** *has no effect.*
> - *Rotating the audio has different effects, depending on what is selected.*

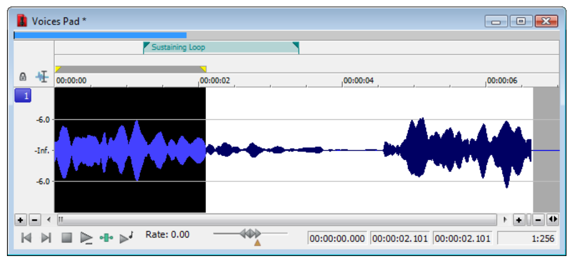
- If no audio is selected, **Rotate Audio** transfers the first 25 percent of the loop to the end of the loop.



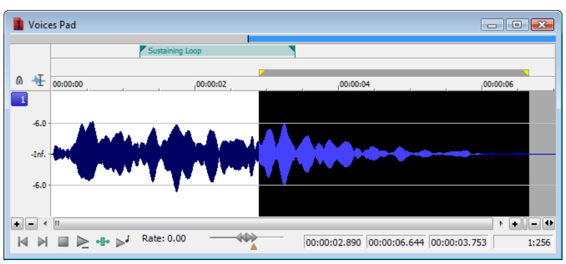


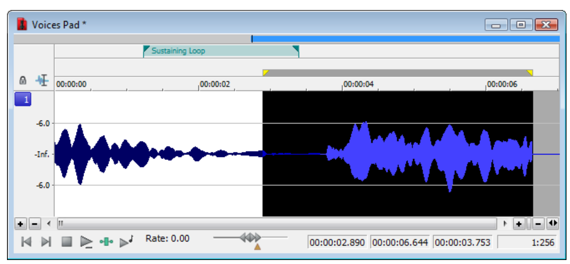
- If audio is selected from the start of a loop, **Rotate Audio** transfers the selection to the end of the loop.





- If audio is selected from the end of a loop, **Rotate Audio** transfers the selection to the start of the loop.





Setting loop tempo

You can calculate, and if necessary edit, the tempo of your loops. Loop tempo is especially important if the loop will be used for building a project in any ACID product. *For more information, see Creating loops for ACID software on page 283*.

Calculating loop tempo

1. Select the loop.
2. From the **Options** menu, choose **Status Format**, and then choose **Edit Tempo** from the submenu. The Edit Tempo dialog is displayed.
3. Specify the number of beats the loop represents in the **Selection length in beats** box.
4. Click the mouse pointer in the **Tempo in beats per minute** box. The loop tempo is calculated and displayed.

Saving loop points

To save loop information with the file, select the **Save metadata with file** check box in the Save As dialog. *For more information, see Using the Save As/Render As dialog on page 72*.

Working with Video

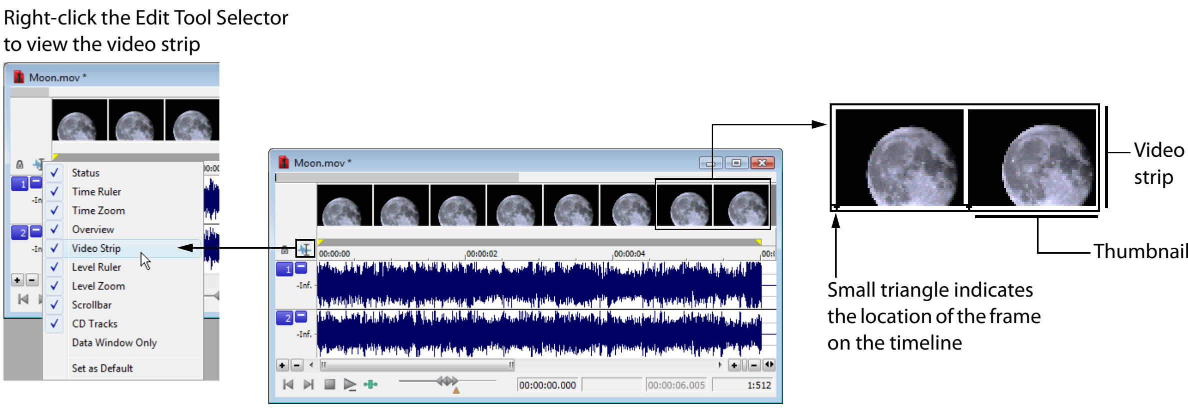
Sound Forge® Pro software supports opening and saving Microsoft® Audio and Video Interleave (AVI), Windows Media® Video (WMV), QuickTime® (MOV), and MPEG video files. You can edit a video file's audio track with single-frame accuracy.

Viewing video

You can view the video portion of a file in the data window's video strip and in the Video Preview window. You can also view video on an external monitor.

Using the video strip

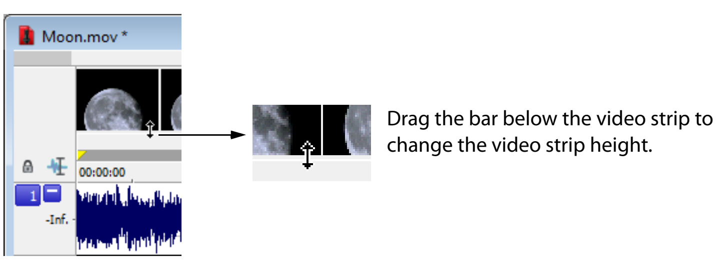
Though Sound Forge software does not perform video editing, the video strip display allows you to navigate video files.



By default, the video strip appears when you open a file containing video. If the video strip is not displayed, right-click the data window's Edit Tool Selector and choose **Video Strip** from the shortcut menu. A check mark appears adjacent to the command and the video strip is displayed. To hide the video strip, choose **Video Strip** from the shortcut menu again.

Changing video strip height

You can change the video strip height by dragging the thin bar at the bottom of the video strip. To change the default height for all video files you open, choose **Preferences** from the **Options** menu and set a **Default video strip height** on the **Display** tab.



Enabling frame animation

When playing a video file, you can specify whether frames are animated or displayed as still frames. To turn on frame animation, right-click the video strip and choose **Animate** from the shortcut menu. A check mark appears adjacent to the command to indicate this feature is turned on.

Using the cursor to select a frame

When frame animation is turned on, clicking anywhere within the audio portion of the data window displays the corresponding video frame in the video strip. To move the cursor by single frames, press Alt+Right Arrow or Alt+Left Arrow.

Viewing frame numbers

You can display frame numbers on each frame in your video strip, which can assist you in positioning your audio. As you zoom in more tightly, each frame in the strip represents one frame in the video.

Frame number

1. Open a video file and display the video strip.
2. From the **Options** menu, choose **Video**, and then choose **Number Frames**. A check mark appears next to this option on the menu when the feature is enabled, and a small box with a number appears at the bottom of each frame. The small black arrow marks the exact position of the frame.

> **Tip:** *You can also right-click the video strip and choose* **Number Frames** *from the shortcut menu.*

3. Using the data window's zoom ratio controls, zoom in/out on the waveform several times and observe the numbering of the video frames.

> **Tip:** *Select a frame number format by choosing* **Preferences** *from the* **Options** *menu and selecting an option from the* **Frame numbering on thumbnails** *drop-down list on the* **Video** *tab.*

Animating the video strip

During playback of a video file, the video strip can display animated or still frames. This can visually aid in editing and positioning your audio to match the video. From the **Options** menu, choose **Video**, and then choose **Animate Video Strip** (or right-click the video strip and choose **Animate** from the shortcut menu). A check mark appears next to this option on the menu when the feature is enabled. When the video strip is animated, the video strip always displays the frame that corresponds to the cursor position. Press Alt+Right Arrow or Alt+Left Arrow to move the cursor one frame.

When frame animation is turned off, the video strip always shows the frame that corresponds to the left edge of each image in the video strip.

> **Tip:** *If you experience slow or stuttering video preview, turn off animated video to reduce the load on your CPU.*

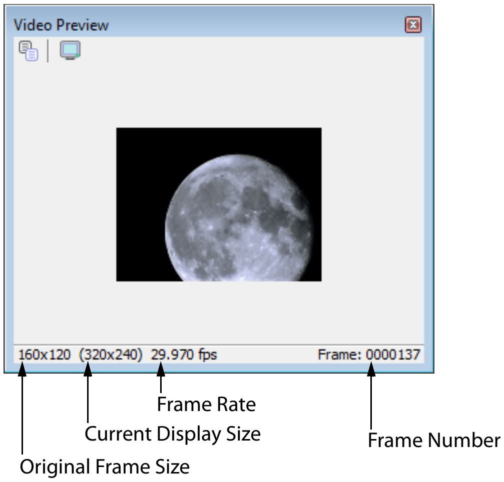
Copying the current video frame to the clipboard

You can use the clipboard to copy the video frame at the current cursor position. From the **Options** menu, choose **Video**, and then choose **Copy Frame** (or right-click the video strip and choose **Copy Frame** from the shortcut menu). The current frame is copied to the clipboard.

Previewing files with video

If you are working with a media file that contains video, you can use the Video Preview window for previewing. You must have the Video Preview window displayed to preview the audio stream. You can hide or display the Video Preview window by choosing **Video Preview** from the **View** menu. To begin previewing the current data window, click the **Play All** button () on the transport bar.

> **Tip:** *To display the* **Video Preview** *window, press Alt+4.*



	Copy Frame to Clipboard Copies the current frame to the clipboard.
	External Monitor Sets up previewing on an external monitor.

Changing the Video Preview settings

The Video Preview window can be configured in a number of ways to make it more useful. The Video Preview window can be used on a separate monitor (if your video hardware supports this feature), docked at the bottom of the workspace, or floated freely on the screen.

You can quickly access settings for the Video Preview window using the shortcut menu. Right-click the Video Preview window to adjust the following options:

Option	Description
Copy Frame	Copies the contents of the frame to Windows Clipboard.
Default Background	Sets the Video Preview window background color to the system default color.
Black Background	Sets the Video Preview window background color to black.
White Background	Sets the Video Preview window background color to white.
Integral Stretch	When selected, the Video Preview frame will only be stretched by integral amounts. Turning this setting on usually provides faster drawing.
Simulate Device Aspect Ration	Compensates for any spatial distortion due to non-square pixel aspect ratios when viewed on a computer monitor.
External Monitor	Sends the preview out to an external monitor. This only functions if your hardware supports this feature. If you have not configured your external monitor settings, clicking this button displays the Video tab of the Preferences dialog, allowing you to choose your external monitor device. *For more information, see Configuring your video settings on page 293.* **Important:** *Pulldown is automatically added when you preview 24p video on an external monitor.*
Passive Update	Reduces the overhead needed to update the Video Display window. The Video Display is updated when the processor is idle.
Show Toolbar	Toggles the display of the toolbar at the top of the window.
Show Status Bar	Toggles the information display at the bottom of the window.

Resizing the Video Preview window

Double-click the title bar of the Video Preview window to automatically resize the window to fit the current video file. Double-click the title bar again to resize the window to half its previous size. This smaller size window allows for faster video frame previewing.

Using an external monitor

You have the option of viewing video on an external monitor. To use this feature, you must have an OHCI-compliant IEEE-1394 DV interface and a device to convert the DV signal to video, such as a DV camcorder, deck, or media converter.

> **Note:** *Pulldown is automatically added when you preview 24p video on an external monitor.*

1. From the **Options** menu, choose **Preferences** and click the **Video** tab (or click the **External Monitor** button ( ) on the Video Preview window).
2. From the **External monitor device** drop-down list, select the appropriate device.
3. Click **Properties** and adjust the following settings as needed:

If	Then
Your source media does not conform to DV standards	Choose a setting from the **If project format is invalid for DV output, conform to the following** drop-down list. The video is automatically adjusted to display properly on your external monitor.
Your audio is not synchronized with your external monitor	You can configure an offset for your hardware. Drag the **Sync offset (frames)** slider to synchronize audio and video. This setting affects synchronization for previewing on an external monitor only; audio and video synchronization in the file is unaffected.

4. Click **Close** to close the External Monitor dialog.
5. Click **OK** to close the Preferences dialog.

Attaching video to an audio file

Once you have edited an audio file to your satisfaction, you can attach it to a video file and save it as a video file.

1. Open the audio file you want to use. *For more information, see Getting media files on page 61*.
2. From the **File** menu, choose **Attach Video**. The Open dialog is displayed.
3. Locate and select a video file you want to attach, and click the **Open** button to attach the video file.
4. From the **View** menu, choose **File Properties**. The File Properties window is displayed.
5. To change the field order setting for the video file, choose an option from the **Video field order** drop-down list. The options are explained below:
   - **None (progressive scan)** For video to be viewed on a computer monitor.
   - **Upper field first** For output that is jittery or shaky, or if specified by your hardware manual.
   - **Lower field first** For DV output.
6. To change the video's pixel aspect ratio, choose an option from the **Video pixel aspect ratio** drop-down list. The pixel aspect ratio should be based on the destination and use of the final media file.

> **Note:** *The file must be saved in a video file format to permanently attach the video.* *For more information, see Saving a video file on page 294*.

Detaching video from an audio file

You can detach the video stream from a media file.

1. Open the media file you want to use. *For more information, see Getting media files on page 61*.
2. From the **File** menu, choose **Detach Video**. The video stream is removed, and the video strip is hidden.

Setting video options

Video file properties

The video properties for a file affect how video is displayed and rendered when you save the file. In most situations, you can leave these settings at their default values. However, you can adjust the video properties of a file as needed.

1. From the **View** menu, choose **File Properties**. The File Properties window appears.
2. Choose a setting from the **Video field order** drop-down list. This setting affects how the video is displayed and rendered when you save the file.
   - **None (progressive scan)** Treats video as non-interlaced.
   - **Upper field first** Treats video as interlaced and reads the interlaced video as upper field first.
   - **Lower field first** Treats video as interlaced and reads the interlaced video as lower field first.

> **Note:** *The* **Video field order** *setting remains in effect only as long as the file is open. The setting is not retained when you save or close the file.*

3. Choose a setting from the **Video pixel aspect ratio** drop-down list to determine the ratio used to display and render the video. In most cases, this value is auto-detected for you.

Configuring your video settings

You can use the **Video** tab on the Preferences dialog to choose your video settings. From the **Options** menu, choose **Preferences**, and then click the **Video** tab.

The items on this tab are explained below.

Item	Description
Frame numbering on thumbnails	This drop-down list determines how the frame information is displayed on the video strip when you have frame number display enabled. To display frame numbers, choose the Frame number option. To display timecode, choose the Media timecode option.
Allow pulldown removal when opening 24p DV	If you want to automatically remove pulldown fields when opening 24 fps progressive-scan DV video files, select this check box. To open your 24p DV video files as 29.97 fps interlaced video (60i), clear this check box.
Deinterlace method	Choose a setting from the drop-down list to determine how the two fields that make up a video frame are separated when you render to a progressive format: • **Blend Fields** Maintains the data in the two fields by blending them together. This method can produce a smooth, motion-blurred image. • **Interpolate** Deletes one field and uses the remaining field to interpolate the deleted lines. This produces sharper images than **Blend Fields** but can introduce jagged motion or stair-stepping artifacts.
Resample source video when rendering to a higher frame rate	Select this check box if you want to interpolate video frames when you render to a frame rate that is greater than the source file's frame rate.
External monitor device	Allows you to identify an external video device with which Sound Forge software can communicate. This video device is used to display previews on an external monitor. **Important:** *Pulldown is automatically added when you preview 24p video on an external monitor.*

Saving a video file

1. From the **File** menu, choose **Save As**. The Save As dialog appears.
2. From the **Save as type** drop-down list, choose a video file format.
3. Name the file in the **File name** box.
4. Select or clear the following check boxes as needed:
   - **Stretch video to fill output frame size (do not letterbox)**: Selecting this check box stretches the source video frame if the destination frame size differs. When this check box is cleared, letterboxing or pillarboxing is used to keep the frame aspect correct.
   - **Fast video resizing**: Selecting this check box speeds the process of saving video. When this check box is cleared, the time required to save the file can increase dramatically. Clear this check box only when you have critical material where nothing but the highest quality video rendering will do.
5. From the **Template** drop-down list, select a template for rendering and compressing the file.

   You can click **Custom** to customize the settings in the Custom Settings dialog. For help on the different settings, click the **Help** button ( ) or press Shift+F1. Click **OK** to close the Custom Settings dialog and return to the Save As dialog.

> **Tip:** *You can save the custom settings to use again by entering a template name in the* **Template** *box and clicking the* **Save Template** *button ( ).*

6. Click **Save**.

Chapter 21

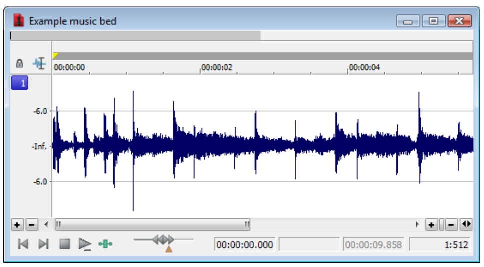
Using Spectrum Analysis

This chapter introduces you to the concept of frequency and describes the Sound Forge® Pro Spectrum Analysis. Spectrum Analysis allows you to examine audio frequencies and overtones using either spectrum graphs or sonograms.

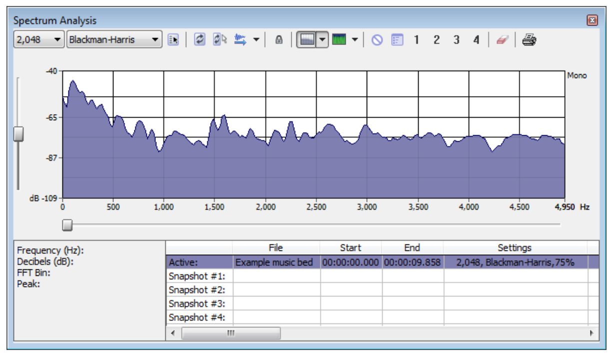
Working in the frequency domain

Unlike the waveform display, which represents audio in the time domain (amplitude vs. time), Spectrum Analysis allows you to examine sound by representing the sound in the frequency domain (amplitude vs. frequency).

Consider the following graphic, which depicts the same audio event as a waveform and as a spectrum graph.



Waveform display



Spectrum graph

Data displayed in the frequency domain (whether in the form of a spectrum graph or sonogram) shows the amplitudes and frequencies of sine waves that, if mixed together, would sound much like the original sound. Since it's relatively easy to remember how a sine wave sounds at different frequencies, it's possible to visualize how simple waveforms sound by looking at the spectrum of the sound.

Learning to "read" the frequency components of a sound in conjunction with their corresponding amplitudes makes it possible to determine the fundamental frequency of a sound, as well as its overtones. Similarly, you can identify unwanted noise, thereby allowing filtering to be applied where needed.

Fast Fourier Transform

A Fourier transform is computationally intensive, and for this reason it is common to use a technique called a Fast Fourier Transform (FFT) to perform spectral analysis. The FFT utilizes mathematical shortcuts to reduce the processing time at the expense of putting limitations on the analysis size.

The analysis size, also referred to as the FFT size, indicates the number of samples from the audio signal used in analysis and also determines the number of discrete frequency bands. When a large number of frequency bands are used, the bands have a smaller bandwidth and this provides for more accurate frequency readings.

However, because complex sounds have a rapidly changing spectrum, a large analysis size can blur the time-changing frequencies of a sound. For example, when performing FFT analysis of an audio file sampled at 44,100 Hz using an analysis size of 4096, almost 100 milliseconds (44,100/4096) of sound are analyzed. If the sound is not constant for those 100 milliseconds, it is impossible to focus on the instantaneous spectrum at smaller time intervals. This is the trade-off between time resolution and frequency resolution encountered when analyzing audio signals.

Spectrum Analysis allows you to perform precise FFT analysis and displays the resulting data in a spectrum graph or a sonogram display. The spectrum graph allows real-time monitoring of playback or input, while the sonogram displays a playback cursor for real-time preview. Both formats make it easy to navigate data and read audio frequency and position.

Using a spectrum graph

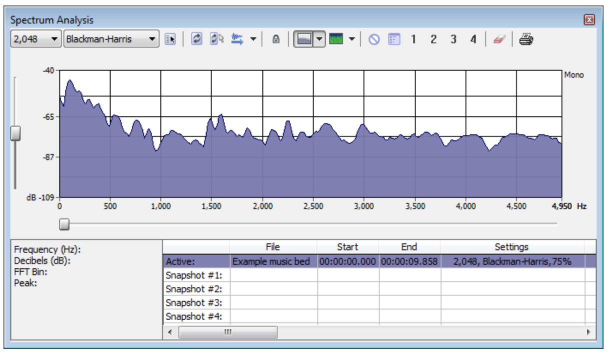
In the spectrum graph, the horizontal axis represents frequency in Hertz (Hz), while the vertical axis represents amplitude in decibels (dB).

Displaying a spectrum graph

1. Open an audio file.
2. Select the portion of the waveform you want to analyze. The sound or note you want to analyze should be in the center of the highlighted area.

> **Note:** *Analyzing long sections of audio can take a long time and decreases the time resolution, so your selection should be relatively short. Also, if the audio has a low amplitude level, you can boost it by using the Volume or Normalize functions. For more information, see Volume on page 204 and Normalize on page 196.*

3. From the **View** menu, choose **Spectrum Analysis**. The Spectrum Analysis window is displayed.



4. Use the toolbar at the top of the window to set your display options.

> **Tip:** *You can also click the* **Settings** *button ( ) in the Spectrum Analysis window to set additional options.*

The spectrum graph displays the amplitude (in dB) of each frequency component from 0 Hz (DC) to the Nyquist frequency. Frequency is displayed along the X (horizontal) axis, and the amplitude is displayed along the Y (vertical) axis.

> **Tip:** *You can continue to make selections in the sound file with the Spectrum Analysis window open (just move the cursor or make selections as you normally would). Click the* **Refresh** *button ( ) in the Spectrum Analysis toolbar to update the display. If no selection is made, analysis is performed on the samples immediately following the cursor position.*

Monitoring an input and output source

Click the **Real Time Monitoring** button ( ) to turn real-time spectrum analysis on or off. Click the down arrow next to the button and choose a command from the menu to specify whether you want to monitor your sound card's input or output:

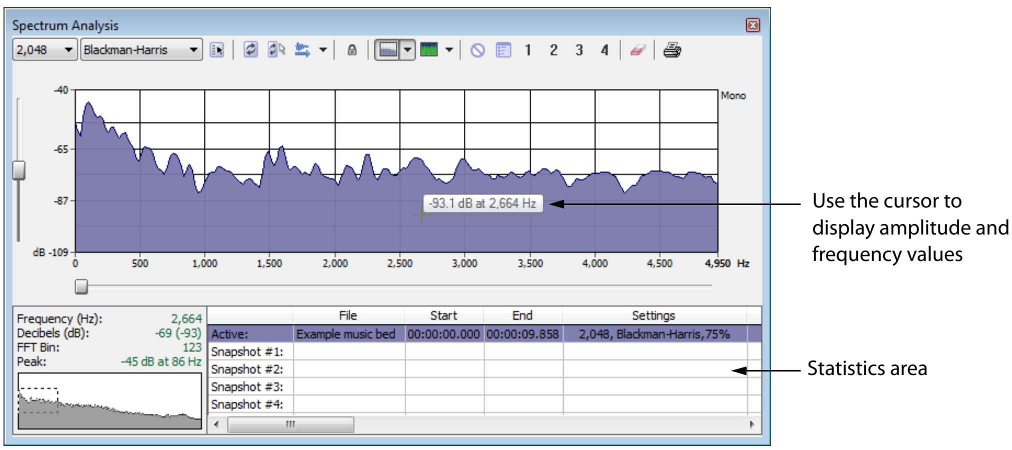
- When you choose **Monitor Input**, Sound Forge will monitor the recording devices selected on the **Record** page of the **Audio** tab in the Preferences dialog. *For more information, see* *Audio tab on page 342*.
- When you choose **Monitor Output**, Sound Forge will monitor the playback devices selected on the **Playback** page of **Audio** tab in the Preferences dialog. *For more information, see* *Audio tab on page 342*.

Notes:

- *When* **Monitor: Output** *is selected, the post-processing signal is monitored when you start playback from the Plug-In Chain.*
- *Real-time spectrum analysis can require significant processing power. If the spectrum graph's refresh rate seems sluggish, set the display mode to Line Graph, decrease the FFT size, or turn off snapshots.*

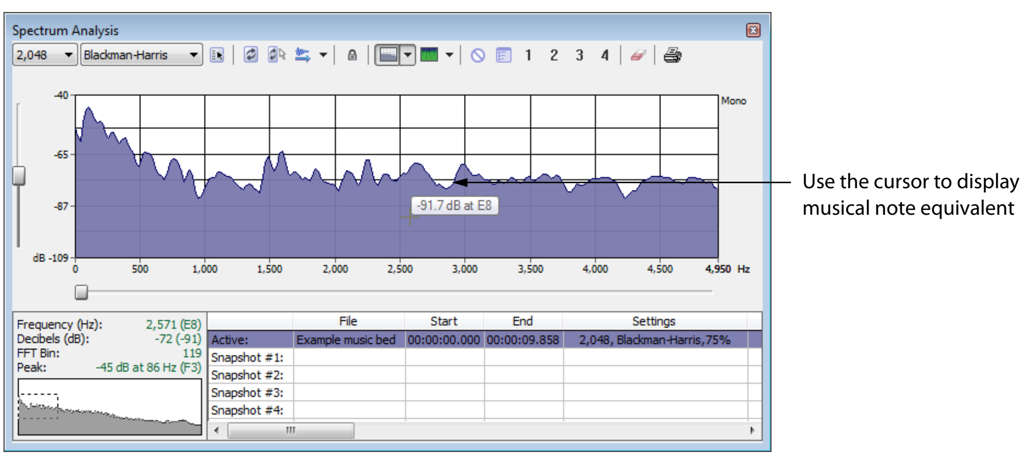
Viewing frequency and amplitude values, notes and statistics

As you move the cursor through the spectrum graph, the amplitude and frequency values at the current position are displayed in a ToolTip next to the cursor and in the Statistics area at the bottom of the window:



Right-click the graph and choose **Show Position** from the shortcut menu to toggle the display of ToolTips. The setting for each graph in a multichannel file is independent.

If you want to display the nearest musical note equivalent of the cursor position in a ToolTip, right-click the graph and choose **Show Notes** from the shortcut menu:



Right-click the Spectrum Analysis window and choose **Show Statistics** from the shortcut menu to toggle the display of the Statistics area at the bottom of the Spectrum Analysis window.

Navigating a spectrum graph

After a spectrum graph is displayed, Grab/Pan mode allows you to scroll vertically and horizontally. To enable Grab/Pan mode, right-click the Spectrum Analysis dialog and choose **Grab/Pan** from the shortcut menu. A check mark appears next to this option when Grab/Pan mode is enabled. The cursor appears as a hand ( ), and you can drag horizontally or vertically to scroll through the graph.

When you are zoomed into a selection of the spectrum graph, you can drag the horizontal and vertical sliders to scroll through the graph. The thumbnail image in the lower-left corner of the Spectrum Analysis window will show you which part of the graph is being displayed.

To turn off Grab/Pan mode, choose **Grab/Pan** from the shortcut menu again.

Changing the graph type

Click the down arrow next to the **Normal Display** button ( ) and choose **Line Graph**, **Filled Graph**, or **Bar Graph** from the menu to change the type of graph displayed in the Spectrum Analysis window. A check mark is displayed next to the selected graph type.

> **Note:** *Some video drivers have problems displaying* **Filled Graph** *and* **Bar Graph** *modes. If you encounter problems such as incorrect shading or very slow drawing, use the* **Line Graph** *option or change video drivers.*

If you're analyzing a multichannel file, you can click the down arrow next to the **Normal Display** button and choose **Single Graph** to see all channels in a single graph.

Right-click the graph and choose **Logarithmic** from the shortcut menu to toggle the x-axis between logarithmic and linear mode. In logarithmic mode, more of the graph is devoted to lower frequencies.

Changing the zoom level of the graph

Zooming can be accomplished in several ways:

1. Drag on the graph to draw a box around the area you want to magnify. You can toggle through mouse selection mode by right-clicking while holding the left mouse button:
   - The first type is a vertical zoom window. This will allow you to zoom to a frequency range.
   - The second type is horizontal zoom window. This will allow you to zoom to an amplitude range.
   - The third type is a combination of vertical and horizontal zoom. This will allow you to zoom to a frequency and amplitude range.
2. Right-click the graph and choose **Zoom Out Full** to view the entire amplitude and frequency range.
3. Right-click the graph and choose **Normalize dB** to set the Spectrum Graph amplitude range equal to the maximum and minimum values in the graph.

Synchronizing graphs in a multichannel file

When viewing a spectrum graph for a multichannel file, an individual graph displays for each channel. Click the **Sync** button ( ) to synchronize the displays so you can view the same region of the FFT in all channels.

Updating a spectrum graph

Select the **Auto Refresh** button ( ) if you want the Spectrum Analysis display to refresh automatically updated when you change your selection in the data window.

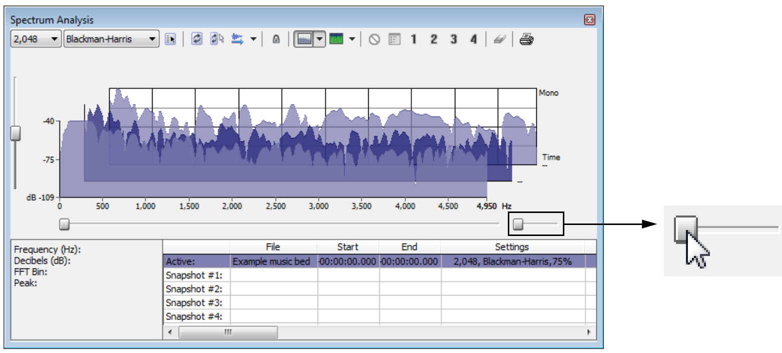
When the button is not selected, the display is not updated until you click the **Refresh** button ( ).

If you want the graph to refresh automatically during playback or input monitoring, select the **Real Time Monitoring** button ( ).

Viewing multiple spectrum graphs

Once you create a selection in the data window, you can display up to 64 individual spectrum graphs (each representing a specific point in time).

1. Open an audio file.
2. From the **View** menu, choose **Spectrum Analysis**. The Spectrum Analysis window appears.
3. Click the **Settings** button ( ). The Spectrum Settings dialog appears. *For more information, see Adjusting Spectrum Analysis settings on page 303.*
4. Type a number in the **Slices displayed** box. The **Forward** and **Backward** radio buttons activate.
5. Select either the **Forward** or **Backward** radio button.
   - Selecting the **Forward** radio button displays the first slice of the selection in the foreground of the spectrum graph.
   - Selecting the **Backward** radio button displays the last slice of the selection in the foreground of the spectrum graph.
6. Click **OK**.
7. Use the **Slice** slider to add or remove slice graphs in the Spectrum Analysis dialog.



Use the **Slice** slider to add or remove slice graphs.

Creating and comparing snapshots of the Spectrum Analysis window

You can store up to four snapshots to compare multiple spectrum graphs. You can take snapshots from a single data window or from different data windows.

> **Note:** *Snapshots are not available in sonogram display or when the* **Slices displayed** *setting in the Spectrum Settings dialog is greater than 1.*

Taking a snapshot

1. Navigate to the portion of the graph you want to capture.
2. Click the **Set Snapshot** button ( ), and then click a snapshot button (1) in the Spectrum Analysis toolbar.

   Available snapshots buttons are displayed in black, and buttons that are in use are displayed in blue and underlined.

Showing and hiding snapshots

1. Select a numbered button in the Spectrum Analysis toolbar to display a stored snapshot. All selected snapshots will be displayed in the Spectrum Analysis window at the same time.
2. Click a selected snapshot button to exclude it from the display.
3. Select the **Hide active plot** button ( ) to hide the current spectrum so you can concentrate on your snapshots.

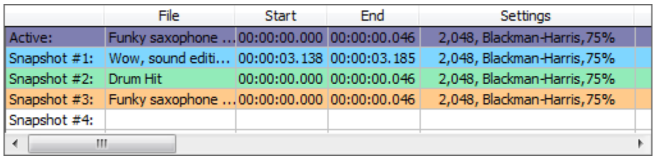
Erasing snapshots

You don't need to erase individual snapshots to update or replace them. Simply click the **Set Snapshot** button ( ), and then click a snapshot button (1) in the Spectrum Analysis toolbar to update its image.

If you want to erase all snapshots, click the **Clear all snapshots** button ( ).

Viewing snapshot statistics

Information about each snapshot is displayed at the bottom of the Spectrum Analysis window:



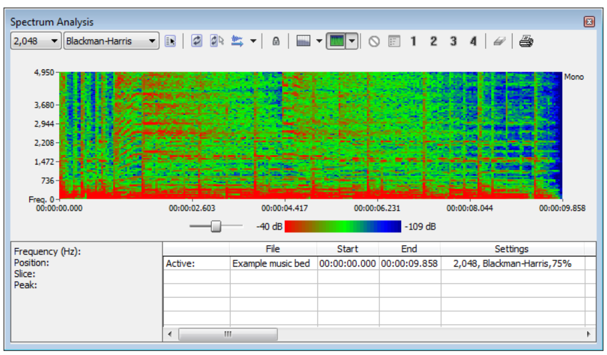
	File	Start	End	Settings
Active:	Funky saxophone ...	00:00:00.000	00:00:00.046	2,048, Blackman-Harris,75%
Snapshot #1:	Wow, sound editi...	00:00:03.138	00:00:03.185	2,048, Blackman-Harris,75%
Snapshot #2:	Drum Hit	00:00:00.000	00:00:00.046	2,048, Blackman-Harris,75%
Snapshot #3:	Funky saxophone ...	00:00:00.000	00:00:00.046	2,048, Blackman-Harris,75%
Snapshot #4:				

Printing the graph

Click the **Print** button ( ) to print the contents of the Spectrum Analysis window, including the graph and statistics data.

Using a sonogram

The sonogram is another way of displaying spectral data variations over time. In a sonogram, the X (horizontal) axis represents time, and the Y (vertical) axis represents frequency.



The amplitude of each frequency component in the sonogram is represented by the color intensity of each point in the graph. This method of displaying spectral information is useful for identifying distinctive spectral patterns created from sounds such as speech, musical instruments, and ambient noise.

Displaying a sonogram

1. Open an audio file and select the portion of the waveform you want to analyze. The sound or note you want to analyze should be in the center of the highlighted area.

Note: *Analyzing long sections of audio can take a long time and decreases the time resolution, so your selection should be relatively short. Also, if the audio has a low amplitude level, you can boost it by using the Volume or Normalize functions. For more information, see* *Volume on page 204 and Normalize on page 196.*

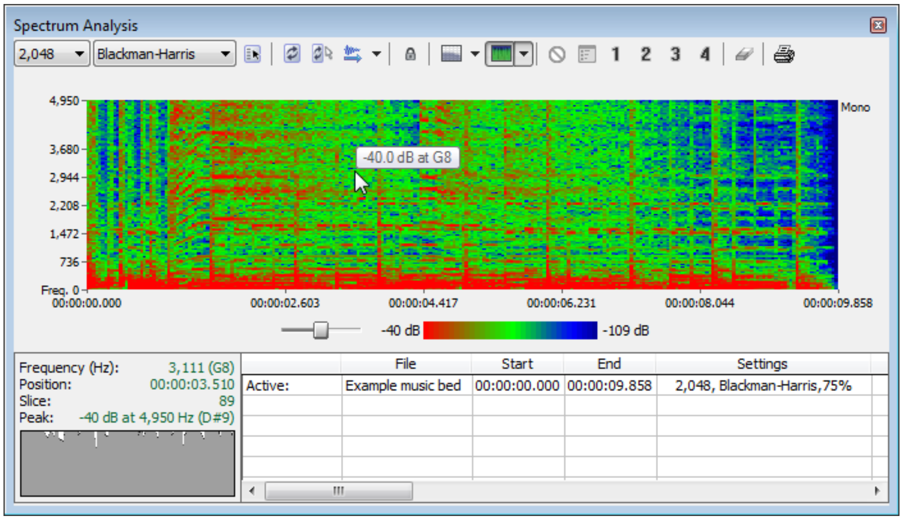
2. From the **View** menu, choose **Spectrum Analysis**. The Spectrum Analysis dialog is displayed.
3. Click the **Sonogram** button ( ) to display your data as a sonogram.
4. Use the toolbar at the top of the window to set your other display options.

Tip: *You can also click the* **Settings** *button ( ) in the Spectrum Analysis window to set additional options.*

If there is no selection in the waveform display window, the sonogram analyzes the sound data from the current cursor position to the end of the file.

Displaying frequency and amplitude values, notes and statistics

As you move the cursor through the sonogram, the amplitude and frequency values at the current position are displayed in a ToolTip next to the cursor and in the Statistics area at the bottom of the window:

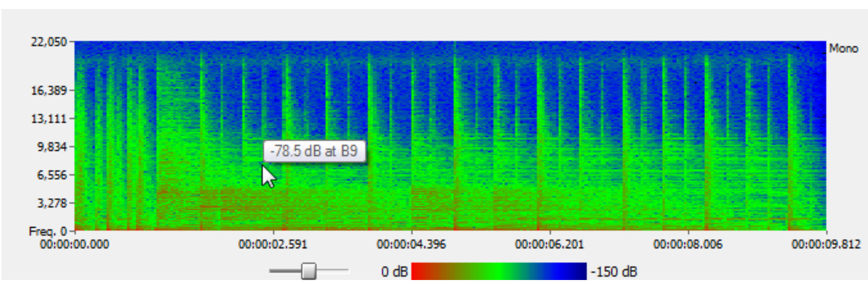


Use the cursor to display amplitude and frequency values

Statistics area

Right-click the sonogram and choose **Show Position** from the shortcut menu to toggle the display of ToolTips. The setting for each sonogram in a multichannel file is independent.

If you want to display the nearest musical note equivalent of the cursor position in a ToolTip, right-click the sonogram and choose **Show Notes** from the shortcut menu:



Use the cursor to display musical note equivalent

Right-click the Spectrum Analysis window and choose **Show Statistics** from the shortcut menu to toggle the display of the Statistics area at the bottom of the Spectrum Analysis window.

Updating a sonogram

A sonogram updates in the same method as a spectrum graph. *For more information, see Updating a spectrum graph on page 298*.

Monitoring an input and output source

Click the **Real Time Monitoring** button ( ) to turn real-time spectrum analysis on or off. Click the down arrow next to the button and choose a command from the menu to specify whether you want to monitor your sound card's input:

- When you choose **Monitor Input**, Sound Forge will monitor the recording devices selected on the **Record** page of the **Audio** tab in the Preferences dialog.
- When you choose **Monitor Output**, a cursor is displayed in the sonogram to indicate the play position (real-time output monitoring is not available in sonogram display mode).

Tuning a sonogram

It is frequently necessary to experiment with the control parameters in the Spectrum Settings dialog to produce the best possible sonogram. *For more information, see* *Adjusting Spectrum Analysis settings on page 303*.

Improving the graph’s contrast

To improve the contrast of the sonogram, decrease the frequency and amplitude ranges as much as possible.

Smoothing the graph’s display

If the graph appears too pixelated, raise the **Set sonogram resolution** value to **200**.

Improving the frequency resolution

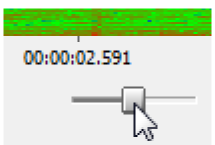
For greater frequency resolution, choose a higher value from the **FFT size** drop-down list.

Reducing the processing time

To reduce processing time, decrease the **Set sonogram resolution** value and/or choose a lower value from the **FFT size** drop-down list.

Adjusting color intensity

Adjust the sonogram’s color intensity using the **Color** slider located directly beneath the sonogram. Notice that the bottom pane of the dialog depicts the color scale in dB.



Adjust the color intensity

Tip: *This function may be fairly slow if you do not have a palletized driver or if Video for Windows is not installed.*

Synchronizing sonograms in a multichannel file

When viewing a sonogram for a multichannel file, an individual graph displays for each channel. Click the **Sync** button ( ) to synchronize the displays so you can view the same region of the FFT in all channels.

Returning to a spectrum graph

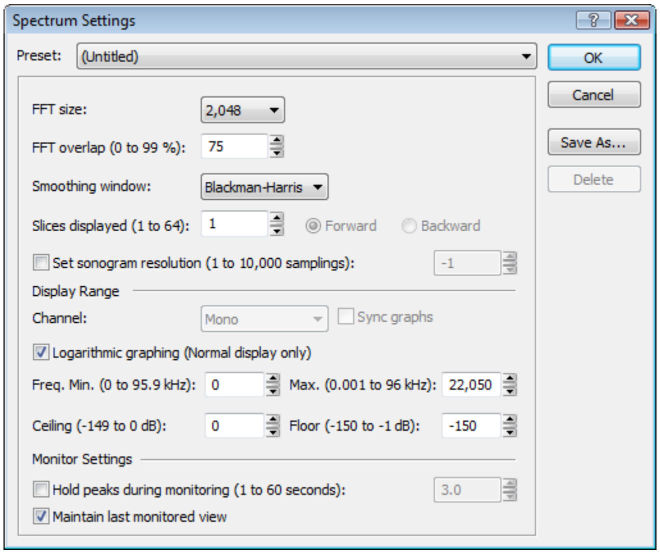
To return to the spectrum graph, click the **Normal Display** button ( ) in the toolbar.

Printing the sonogram

Click the **Print** button ( ) to print the contents of the Spectrum Analysis window, including the sonogram and statistics data.

Adjusting Spectrum Analysis settings

From the Spectrum Analysis toolbar, click the **Settings** button ( ) to display the Spectrum Settings dialog. The following table explains the role of each control in audio spectrum analysis.



Item	Description
FFT size	Choose a value from the **FFT size** drop-down list to set the size in samples of the analysis window and number of discrete frequencies analyzed. Higher numbers produce increased frequency resolution at the expense of lower time resolution and longer computational time.
FFT overlap	The value in the **FFT overlap** box specifies the percentage of overlap between FFT analysis windows. Overlapping allows for more accurate analysis. Lower settings decrease the number of distinct analysis functions performed, which decreases processing time. High settings allow for more analysis, but can result in slow processing.
Smoothing window	Choose a setting from the **Smoothing window** drop-down list to determine the window function applied to the input data before analysis. The window function affects the sharpness of peaks in an FFT graph and the leakage into neighboring frequencies. • Choose **Rectangle** to apply no window. This results in a very sharp peak, but high leakage. • Choose **Triangular** (also called a Bartlett or Parzen window) to apply a window that results in less leakage than the rectangle window. • **Hamming**, **Hanning**, and **Blackman** windows are commonly used in audio applications. • Choose **Blackman-Harris** to obtain the least sideband leakage of the six options. The major drawback of Blackman-Harris is rounded graph peaks.
Slices displayed	The **Slices displayed** value determines the number of FFT slices displayed. When displaying multiple slices in the spectrum graph, slices are displayed chronologically forward or backward based on whether you have the **Forward** or **Backward** radio button selected.
Set sonogram resolution	The **Set sonogram resolution** value determines the number of FFT samplings used in a sonogram. This keeps the processing time and graph resolution constant. Increasing this value increases the horizontal graph resolution, but requires more processing time. When this check box is cleared, the number of samplings is determined by the length of the selection and the **FFT overlap** setting. **Note:** *Increasing the samplings increases the horizontal graph resolution but requires more processing time.*
Channel	Choose a setting from the drop-down list to specify which graph you want to edit.

Item	Description
Sync graphs	If you are analyzing a multichannel file, select the **Sync graphs** check box to synchronize the displays so you can view the same region of the FFT in all channels.
Logarithmic graphing	Select the **Logarithmic graphing** check box to display the X-axis in logarithmic mode rather than linear mode. In logarithmic mode, more of the graph is devoted to lower frequencies. **Note:** *Logarithmic graphing affects the display only when* **Normal Display** *is selected.*
Freq. Min. (Frequency minimum)	Determines the lowest frequency displayed in the graphs.
Max. (Frequency maximum)	Determines the highest frequency displayed in the graphs.
Ceiling	Determines the highest amplitude displayed in the graphs.
Floor	Determines the lowest amplitude displayed in the graphs.
Hold peaks during monitoring	Select this check box to indicate the highest value of each frequency on the spectrum graph with a small horizontal line. The length of time (in seconds) that the peak is held is determined by the value entered in the edit box.
Maintain last monitored view	Select this check box if you want to maintain the state of the Spectrum Graph when you stop playback. Clearing this check box results in the graph resetting to the cursor position when playback stops.

Saving spectrum graph settings

After you configure the controls in the Spectrum Settings dialog, you can save the settings as a custom preset by clicking **Save As** and entering a name for the new preset. Click **OK** to save the new preset.

Chapter 22

Editing with SpectraLayers Pro

From the Tools menu, choose **Edit in SpectraLayers Pro** to open the current sound file in SpectraLayers Pro.

If SpectraLayers Pro 2.0 is not installed, the command is not available. For more information, please see *http://www.sonycreativesoftware.com/spectralayerspro*.

Editing a file in SpectraLayers Pro

1. Select a data window or drag the cursor in the data window or marker bar to make a time selection.
2. From the Tools menu, choose **Edit in SpectraLayers Pro**. SpectraLayers Pro starts, and the active data window is loaded as a new layer in a SpectraLayers project.
3. Edit your file as needed in SpectraLayers Pro. For information about using SpectraLayers Pro, please refer to the application help.
4. Save your file in SpectraLayers Pro. You'll be prompted to export your changes back to Sound Forge. Click **Yes**, and the Sound Forge data window is updated to reflect any changes.

Sending a file to a layer in SpectraLayers Pro

1. Select a data window or drag the cursor in the data window or marker bar to make a time selection.
2. From the Tools menu, choose **Send to SpectraLayers Pro**. If SpectraLayers Pro is running, the selection is added to the current project as a new layer. If SpectraLayers Pro is not running, the application starts, and the active data window is loaded as a layer in a new project.
3. Repeat step 2 as needed to build layers in your SpectraLayers Pro project.
4. Edit your file as needed in SpectraLayers Pro. For information about using SpectraLayers Pro, please refer to the application help.
5. When you're done editing, you can save your project in SpectraLayers Pro, render the mixed output, or use **Process** > **Send to Sound Forge Pro** to save your changes.

Chapter 23

Burning CDs

You can write audio to CD if your system is configured with a supported CD-R/RW drive and the necessary drivers.

Understanding track-at-once and disc-at-once burning

Sound Forge® Pro software provides two ways to burn audio to a CD: track-at-once and disc-at-once.

Track-at-once

Track-at-once burning records individual tracks to the disc and results in a partially recorded disc. However, the CD remains unplayable on most systems until you close the disc. The advantage of track-at-once burning is you can record tracks onto the disc as you finish them versus waiting until you have finished your whole album. Track-at-once writing burns the entire project as a single track.

Disc-at-once (single session or Red Book)

Disc-at-once burning is the most common burning method in the music industry. This writing mode is used when creating a master disc to be sent to a disc manufacturer for mass replication. Disc-at-once works just as it sounds. Multiple tracks of audio are written to the CD in one recording session.

Correcting the sample rate for CD burning

Sample rates deviating from 44,100 Hz cause CD track lengths to be miscalculated. When attempting to write a file to CD that deviates from the 44,100 Hz sample rate, you are prompted to change the sample rate. Selecting **Yes** automatically resamples audio to 44,100 Hz.

In addition, you can use the **Resample** tool to change the sample rate of a file prior to burning the CD. *For more information, see Resample on page 200*.

Writing mono tracks to a CD

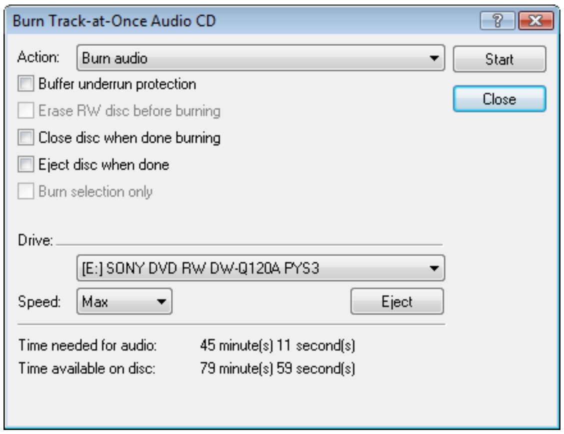
If you attempt to write mono audio tracks to a CD, you are prompted to create a stereo file by copying the mono data to both channels.

Burning track-at-once (TAO) CDs

You should always save your audio files prior to writing them to CD.

1. From the **Tools** menu, choose **Burn Track-at-Once Audio CD**. The Burn Track-at-Once Audio CD dialog is displayed. The bottom of the dialog displays the length of the current audio file and the amount of time remaining on the CD currently in the CD-R/RW.

> **Note:** *If there is no CD in the current drive, only the* **Drive** *and* **Speed** *drop-down menus and the* **Close** *button are available in this dialog. If you insert a disc or select a different drive after this dialog is displayed, it takes a moment to recognize the disc and make all options available.*



2. Choose a setting from the **Action** drop-down list:

Setting	Description
Burn audio	Begins recording audio to your CD when you click the **Start** button. You will need to close the disc before it can be played in an audio CD player.
Test, then burn audio	Performs a test to determine whether your files can be written to the CD recorder without encountering buffer underruns. Recording begins after the test if it is successful.
Test only	Performs a test to determine whether your files can be written to the CD without encountering buffer underruns. No audio is recorded to the CD.
Close disc	Closes your disc without adding any audio when you click the **Start** button. Closing a disc allows your files to be played on an audio CD player.
Erase RW disc	Erases your rewritable CD when you click the **Start** button. You should use this option if your rewritable CD already has data on it.

3. Select your burning options:

Option	Description
Buffer underrun protection	Select this check box if your CD recorder supports buffer underrun protection. Buffer underrun protection allows a CD recorder to stop and resume burning.
Erase RW disc before burning	If you're using a rewritable CD, select this check box to erase the CD before you begin burning if your rewritable CD already has data on it.
Close disc when done burning	Select this check box to close the CD after burning. Closing a disc allows your files to be played on an audio CD player. **Note:** *You can close the disc using a separate step later. For more information, see Closing a CD on page 309.*
Eject disc when done	Select this check box to eject the CD automatically when burning has completed.
Burn selection only	Select this check box to burn only the audio within the loop region.

4. From the **Drive** drop-down list, choose the CD-R/RW drive that you want to use to burn your CD.
5. From the **Speed** drop-down list, choose the speed at which you want to burn. **Max** will use your drive's fastest possible speed; decrease the setting if you have difficulty burning.

6. Click the **Start** button.

> **Important:** *Clicking* **Cancel** *after the CD writing process begins renders the CD unusable.*

After the audio is written to CD, the CD Operation dialog indicates whether the writing was successful.

7. Click **OK** to clear the message.

Closing a CD

Closing the CD allows you to listen to it in an audio CD player. However, you cannot add tracks to a CD once it is closed.

1. From the **Tools** menu, choose **Burn Track-at-Once Audio CD**. The Burn Track-at-Once Audio CD dialog is displayed.
2. From the **Action** drop-down list, choose **Close Disc**.
3. If desired, select the **Eject disc when done** check box to eject the CD automatically when the disc has been closed.
4. Click the **Start** button. The Sound Forge application begins closing the CD and displays a progress meter in the dialog.

   After the CD is closed, the CD Operation dialog indicates whether the closing was successful.
5. Click **OK** to clear the message.

Burning disc-at-once (DAO) CDs

From the **Tools** menu, choose **Burn Disc-at-Once Audio CD** to burn a disc-at-once CD using the current CD layout.

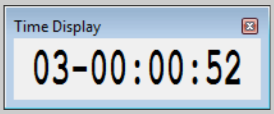
Use DAO CDs when you need to create a master disc for mass replication.

Tip: *When creating DAO CDs, right-click the ruler or Time Display window and choose* **Audio CD Time** *from the shortcut menu to help you arrange your project. In Audio CD Time format, the ruler will display hh:mm:ss:ff (hours:minutes:seconds:frames). Audio CD time uses a frame rate of 75 fps.*

Next, choose **Quantize to Frames** *from the* **Options** *menu to ensure that any edits you make will occur on frame boundaries.*

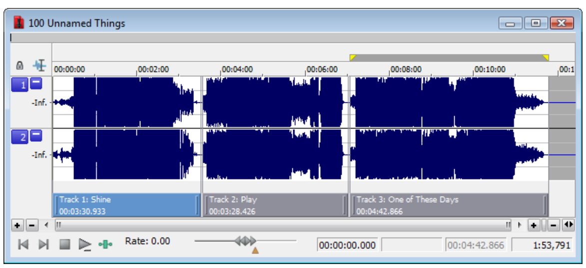
If your first track region begins before 00:00:02:00, a timeline offset is automatically added so the first track begins at exactly two seconds. This offset is added for burning only and is not reflected in the data window.

If you want to display track numbers in the Time Display window, right-click the Time Display window and choose **CD Track Position** *from the shortcut menu. In this mode, the Time Display will show track numbers and the running time for each track. Negative values indicate the pause time before a track:*



Creating and editing tracks for disc-at-once CDs

A disc-at-once CD requires that you define a track list before burning.



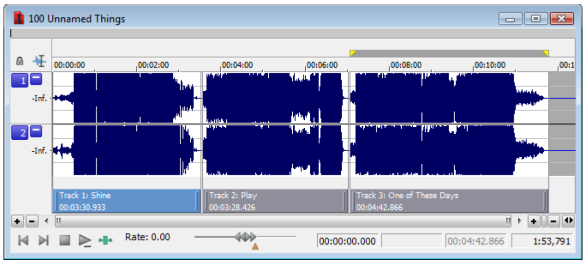
From the **Insert** menu, choose **CD Track** to add a CD track using the current selection as the track length.

Sound Forge also provides several other methods of adding tracks.

Adding CD tracks and index markers to a sound file

Creating CD tracks

1. Select the time range that you want to use to create a track. A track must be at least four seconds long.
2. From the **Insert** menu, choose **CD Track** (or press N). A CD track is added to the CD layout bar in the data window.

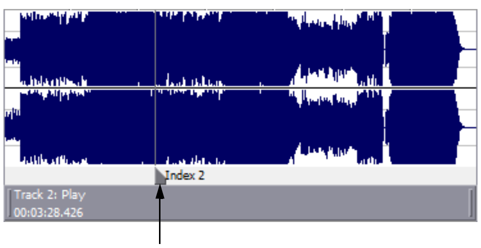


> **Tip:** *You can drag CD tracks to rearrange them, drag either end of a track to change its length, or use the Track List window to edit the track's position or name.*

Creating CD index markers

You can use index markers to subdivide a track. For example, in a track that contains an orchestral composition, index markers could allow navigation to each movement. Each track on a Red Book audio CD can contain up to 99 index markers.

1. Click to position the cursor where you want to add an index marker.
2. From the **Insert** menu, choose **CD Index** (or press Shift+N). A CD index marker is added to the CD layout bar in the data window.



CD index marker

Creating CD tracks from events in a data window

If you're working with a data window that contains multiple events, you can create tracks automatically using the events.

1. Create events on the timeline to lay out your CD. *For more information, see Using the Event Tool on page 171*.
2. From the **Edit** menu, choose **Track List**, and then choose **Create CD Tracks from Events** from the submenu. Each event is marked with a CD track.

> **Note:** *Tracks are not created for events less than four seconds long.*

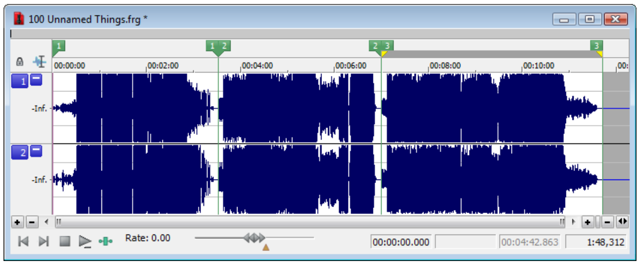
You can drag CD tracks to rearrange them, drag either end of a track to change its length, or use the Track List window to edit the track's position or name.

> **Tip:** *When space exists between events, Sound Forge Pro creates separate tracks for each event. If you want to create a track that spans multiple events, you can abut or overlap the events to create a single track.*

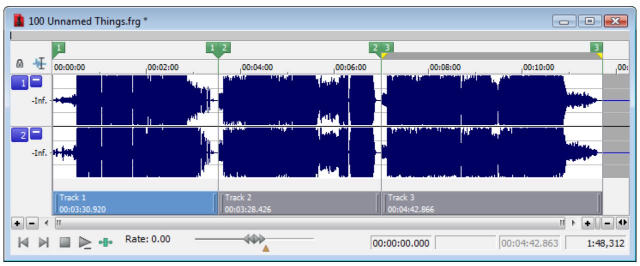
Creating CD tracks from regions in a file

If you have a live recording that uses regions to indicate the sections of the recording, you can use this feature to create tracks without having to scan through the audio and create tracks manually.

1. Add regions as necessary to indicate the tracks in your recording.



2. From the **Edit** menu, choose **Track List**, and then choose **Create Tracks from Regions**. Each region is marked with a CD track.



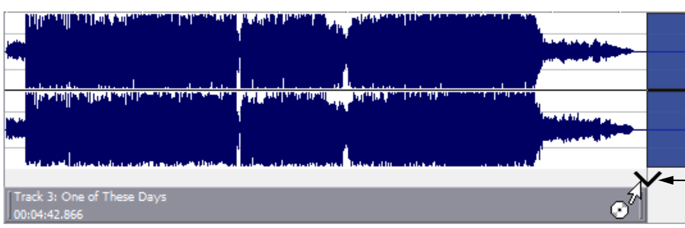
You can drag CD tracks to rearrange them, drag either end of a track to change its length, or use the Track List window to edit the track's position or name.

Adding files to a data window and creating tracks

> **Tip:** *When you create a track using a media file that includes title and artist metadata, this information will be added to the Track List window as CD Text.*

Dragging files to the CD layout bar

Drag a file to the CD layout bar at the bottom of a data window. An indicator is displayed to show you where the track will be added when you release the mouse. When you drop the file, a track is created, and pause time is added before the new track.

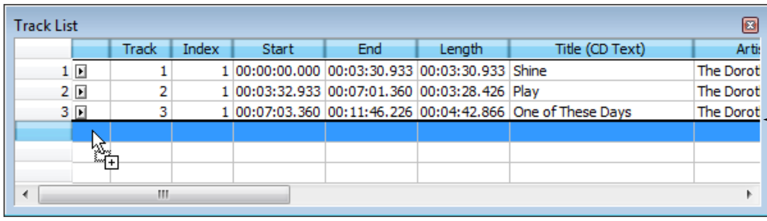


Track placement indicator

> **Note:** *If you want to add a CD track without pause time, drag the file to the waveform display above the CD layout bar. If* **Options** > **Drag-and-Drop Editing** > **CD Track** *isn't selected, click the right mouse button while dragging until the cursor is displayed with a CD icon () to show you where the track will be added.*

Dragging files to the Track List window

Drag a file to the Track List window. An indicator is displayed to show you where the track will be added when you release the mouse. When you drop the file, a track is created, and pause time is added before the new track.



Track placement indicator

Dragging files from the Explorer window

From the **Options** menu, choose **Drag-and-Drop Editing**, and then choose **CD Track** from the submenu if you want to create disc-at-once tracks by dragging files from the Explorer window or Windows Explorer to a data window. Choosing this command has the same effect as toggle-clicking the right mouse button while dragging until the cursor is displayed with a CD icon () to show you where the track will be added.

When you drop the file, the audio in the data window will be replaced, and no pause time will be added.

> **Tip:** *You can use the* **Default time between CD tracks** *control on the CD Settings tab of the Preferences dialog to adjust the default pause time. For more information, see* *CD Settings tab on page 341*.

Opening multiple files as CD tracks

In the Open dialog, hold Ctrl or Shift to select the files you want to open as CD tracks and then select the **Open as CD tracks** and **Append to current data window** check boxes.

The files will be added to the end of the current data window and a CD track will be created for each file.

> **Tip:** *You can use the* **Default time between CD tracks** *control on the CD Settings tab of the Preferences dialog to adjust the default pause time. For more information, see* *CD Settings tab on page 341*.

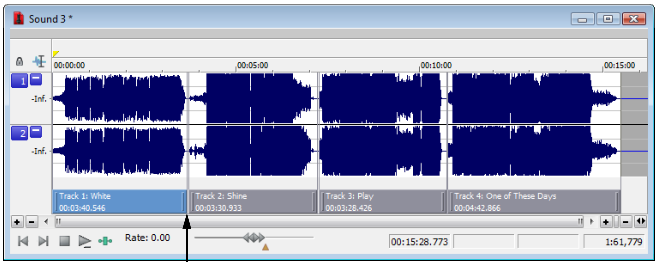
Creating new data windows using audio files on your computer

If no audio files are open in the Sound Forge workspace, a new data window will be created when you drop files on the Track List window.

1. Close all open data windows.
2. In the Windows Explorer or Explorer window, select the files that you want to include in your CD. You can hold Shift or Ctrl to select multiple files.

3. Drag the files to the Track List.

   When you drop the files, a track is created for each file, and pause time is added before each track.



Pause time

Tips:

- *When you create tracks using media files that include title and artist metadata, this information will be added to the Track List window as CD Text.*
- *You can use the* **Default time between CD tracks** *control on the* **CD Settings** *tab of the Preferences dialog to adjust the default pause time.*

Using the Track List window to create tracks

The Track List window is essentially a text representation of the markers on the CD layout bar.

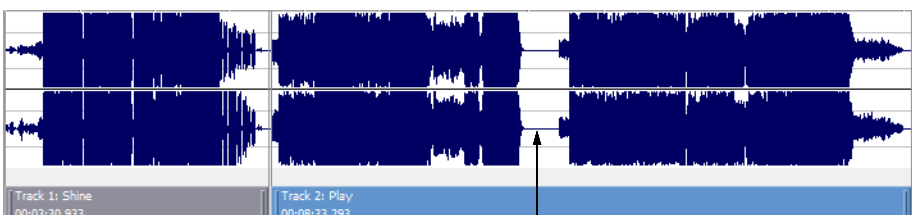
You can use the Track List window to view track and index markers, edit track position and length, edit track names, adjust pause time, toggle protection and emphasis flags, and edit ISRC data.

For more information, see Using the Track List window on page 318.

Adding bonus tracks at the end of CDs

You can hide a track at the end of a CD by adding silence at the end of the last track and adding the new track after the silence. Drag the end of the final track marker to the new end of the CD.

Because both events exist within a single track, they will be treated as one track by an audio CD player.

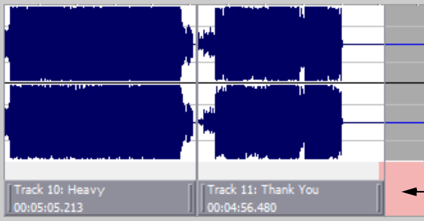


Silence between final track and bonus track

Moving tracks on the CD layout bar

The CD layout bar displays information about the tracks you've created for your disc-at-once CD project. Each CD track shows the track's number, active take name, and length.

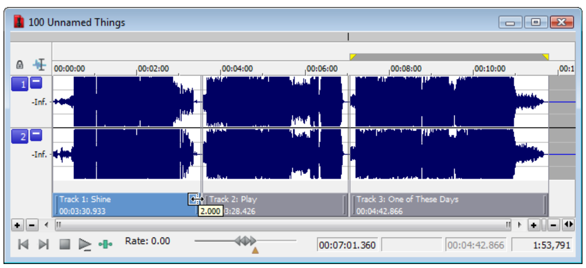
Note: *A red indicator is drawn at the end of the CD layout bar to represent the end of the disc (if the disc length is known). You can use the* **Automatically detect CD length** *and* **Default CD length** *controls on the* **CD Settings** *tab of the Preferences dialog to set CD length.*



You can use the CD layout bar to perform many of the track-editing functions from the Track List window.

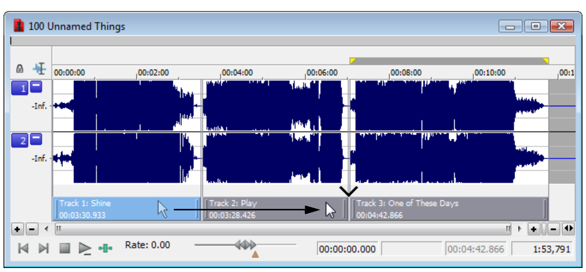
Changing track starting and ending points

Drag either end of the track to adjust the track's starting or ending position. The pause time between tracks is displayed in a ToolTip:



Reordering tracks

Drag a CD track to move the track, its associated media, and the pause time before the track. An indicator is displayed to show you where the event will be moved:



If you drag a track in a project where a single event spans more than one track, Sound Forge will split the event as necessary.

Tips:

- *Hold Shift while clicking CD tracks to select a range of tracks.*
- *From the Options menu, choose* **Lock to Selection**, *and then choose* **Audio and CD Tracks** *if you want to drag CD tracks when moving audio in the data window.*
- *Regions, markers, and envelope points are moved with a CD track. To turn this feature off, turn off the* **Lock to Selection** > **Markers/Regions** *and* **Envelope Points** *commands on the Options menu.*

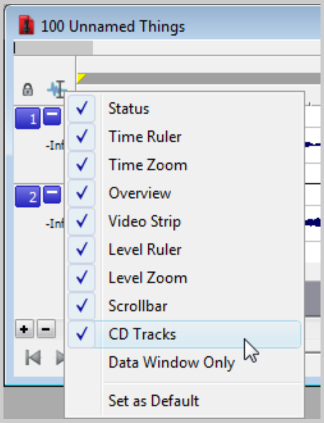
Deleting tracks

Right-click a track and choose **Delete** from the shortcut menu.

Hiding the CD layout bar

From the **Options** menu, choose **Data Window**, and then choose **CD Tracks** from the submenu. The CD layout bar is hidden. Choose **CD Tracks** from the submenu again to display the CD layout bar.

Tip: *Right-click the* **Next Edit Tool** *button to display or hide components for the selected data window.*

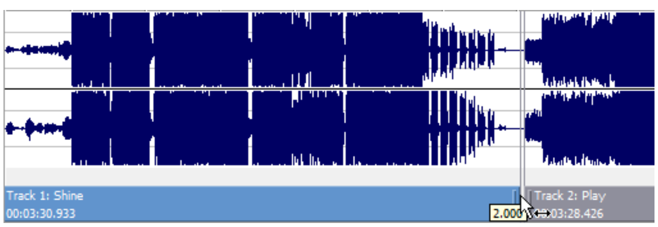


Editing pause time for a disc-at-once CD

Pause time is the space between CD tracks. This space may contain silence—as in a standard commercially produced CD—or can contain audio, as in a live performance captured on CD.

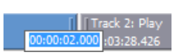
The Red Book standard calls for two seconds of pause time, but you can edit the default pause time on the **CD Settings** tab of the Preferences dialog.

When you hover in the pause time between two tracks, Sound Forge displays the pause time.



You can edit the pause time in several ways:

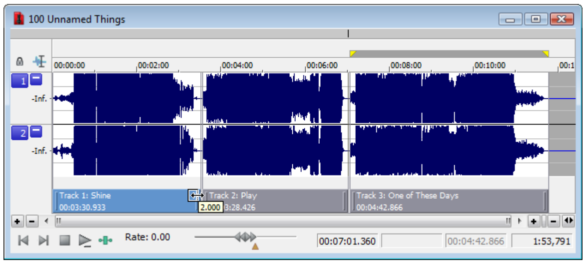
- Hover your mouse between two tracks in the CD layout bar. The pause time is displayed in a ToolTip. Double-click the ToolTip, and the display changes to an edit box where you can type a new value.

- Right-click between two tracks and choose a command from the shortcut menu:

Command	Description
Select Pause Time	Creates a time selection equal to the pause time between tracks.
Set to Default Pause Time	Moves all tracks upstream so the default pause time exists between the tracks where you clicked.
Edit Pause Time	Changes the pause time display to an edit box where you can type a new value.

- Drag either end of the track to adjust the track's starting or ending position. The pause time between tracks is displayed in a ToolTip:



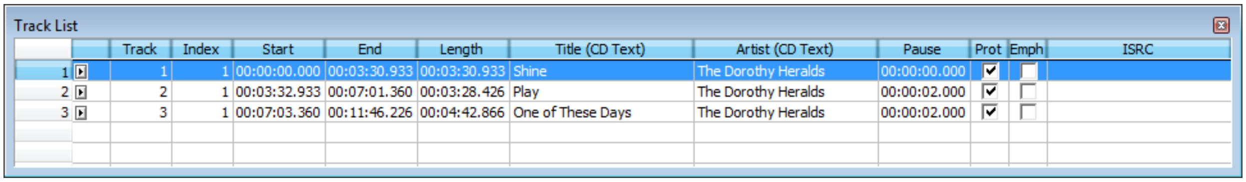
- The **Pause** column in the Track List window allows you to edit the pause time between tracks. Type a new value in the box, and tracks will move accordingly in the timeline.

Track List

	Track	Index	Start	End	Length	Title (CD Text)	Artist (CD Text)	Pause	Prot	Emph	ISRC
1	1	1	00:00:00.000	00:03:30.933	00:03:30.933	Shine	The Dorothy Heralds	00:00:00.000	☑	☐	
2	2	1	00:03:32.933	00:07:01.360	00:03:28.426	Play	The Dorothy Heralds	00:00:02.000	☑	☐	
3	3	1	00:07:03.360	00:11:46.226	00:04:42.866	One of These Days	The Dorothy Heralds	00:00:02.000	☑	☐	

Using the Track List window

From the **View** menu, choose **Metadata**, and then choose **Track List** from the submenu to toggle the display of the Track List window.

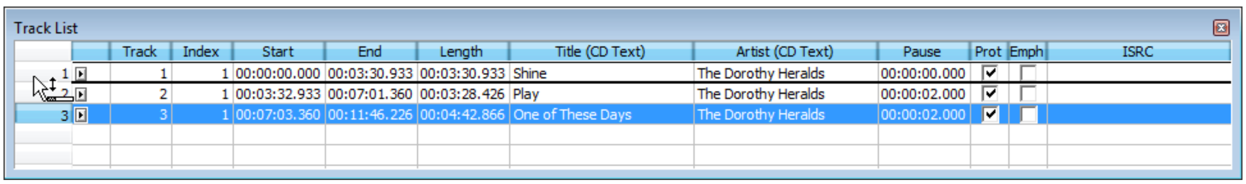


The Track List window is essentially a text representation of the events on the CD layout bar for a disc-at-once CD. You can use the Track List window to view track and index markers, edit track position and length, edit track names, adjust pause time, toggle protection and emphasis flags, and edit ISRC data.

You can also copy, save, and print the contents of the Track List window if you need to provide a track list to a CD duplicator.

Reordering tracks

1. Click in the numbered column to select a row.
2. Drag the row to a new position.



The track is moved to the position where you drop it, and the timeline is updated.

Editing track position and length

Double-click the **Start**, **End**, and **Length** boxes and type a new value in the box to edit a CD track's starting or ending point or length.

Typing a new value in the boxes has the same effect as moving or resizing the CD track on the CD layout bar:

- Editing the **Start** or **End** value moves the track forward or backward in time while preserving its length.
- Editing only the **Length** value changes the track's ending time while preserving its start time.

Editing track title and artist information

Double-click the **Title (CD Text)** and **Artist (CD Text)** boxes to edit their contents.

Notes:

- *In order to burn valid CD Text, you must specify a title for the disc and for each track on the disc (artist information is optional). If the* ***Title*** *box in the CD Information window or Track List window is left blank, a warning will be displayed before burning so you can choose to write the disc without CD Text or cancel burning and add title information as needed.*
- *You can write a maximum of 5000 characters as CD Text.*

If you select the **Write CD Text** check box on the Burn Disc-at-Once CD dialog, this data will be written to your disc. In order to display CD Text, your CD player must support CD Text.

Adjusting pause time

Double-click the **Pause** box to edit the pause time between tracks. Type a new value in the box, and tracks will move accordingly in the timeline.

Toggling protection and emphasis flags

Select the **Prot** check box to add a flag to the Q subcode to prevent digital copying of your CD.

In order to use copy protection, the CD player must support the copy-protection flag.

Select the **Emph** check box to add a pre-emphasis flag to the Q subcode.

Pre-emphasis is a basic noise-reduction process that is implemented by a CD player. Emphasis involves boosting high frequencies during CD writing and cutting those frequencies during playback. The emphasis process reduces high-frequency noise without disrupting the natural frequency of the source material.

Selecting the **Emph** check box does not impart the pre-emphasis boost on a track; it can only set the flag. In order for pre-emphasis to occur, the CD recorder and player must support the flag. Check your CD drive documentation to determine whether your drive supports pre-emphasis flags.

Editing ISRC codes

The Track List window allows you to specify an ISRC (International Standard Recording Code) that will be used to identify the tracks on your disc.

For more information about ISRC codes, see *http://www.ifpi.org/content/section_resources/isrc.html*.

1. Double-click the **ISRC** box in a track row.
2. Type the appropriate code for the track.
3. Press Enter.

Industry Standard Recording Codes (ISRC) were designed to identify CD tracks. The ISRC code is a 12-character alphanumeric sequence in the following format:

Field	A	B	C	D	E
Sample ISRC	SE	T38	86	302	12

Field	Description
A	**Country** Represents the recording's country of origin.
B	**First Owner** Assigned ID for the producer of the project. Each country has a board that assigns these codes.
C	**Year of Recording** Represents the year the recording was made.
D	**Recording** Represents the recording's serial number made by the same producer in that year: This value will use three digits (300-999) when the CD has 10 or more tracks. This value will use four digits (0001-2999) when the CD has 9 or fewer tracks.
E	**Recording Item (1 or 2 digits)** Identifies tracks on a CD (each track can have a different ISRC code).

Creating track lists (PQ lists or cue sheets)

You can right-click the Track List window and choose **Copy Track List to Clipboard**, **Export Track List**, or **Print Track List** to share your track list information with another application or with a CD-replication house.

For more information, see Creating track lists for disc-at-once CDs on page 319.

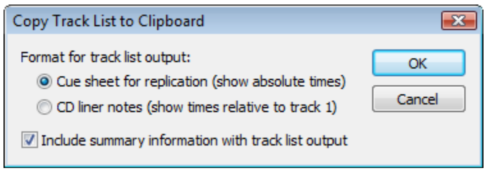
Creating track lists for disc-at-once CDs

If you need to share your track list information with another application or with a CD replication house, you can copy the track list information to the clipboard, save it to a text file, or print a hard copy.

Copying track lists to the clipboard

1. From the **Edit** menu, choose **Track List**, and then choose **Copy Track List to Clipboard** from the submenu. The Copy Track List to Clipboard dialog is displayed.

Tip: *Right-click the Track List window and choose* **Copy Track List to Clipboard** *from the shortcut menu.*

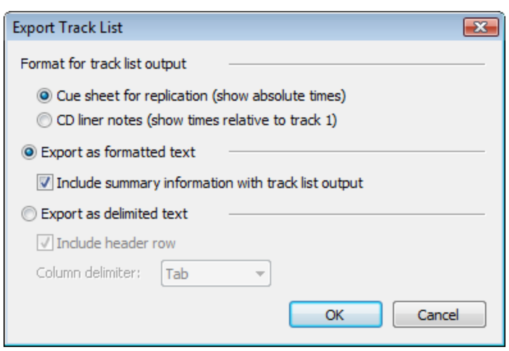


2. Choose the format you want to apply to your track list information:
   - **Cue sheet for replication** The track list is formatted so track times are listed relative to the beginning of the CD. This is the preferred format for CD replication houses.
   - **CD liner notes** The track list is formatted so track times are listed relative to the first track on the disc.
3. Select the **Include summary information with track list output** check box if you want to include UPC/MCN, title, engineer, and comment information with the track list.
4. Click **OK** to send the track list information to the clipboard. You can then paste the information into a text editor or e-mail message.

Exporting track lists as text files

1. From the **Edit** menu, choose **Track List**, and then choose **Export Track List** from the submenu. The Export Track List dialog is displayed.

Tip: *Right-click the Track List window and choose* **Export Track List** *from the shortcut menu.*

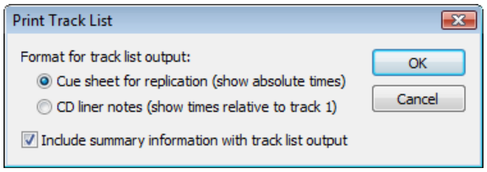


2. Choose the format you want to apply to your track list information:
   - **Cue sheet for replication** The track list is formatted so track times are listed relative to the beginning of the CD. This is the preferred format for CD replication houses.
   - **CD liner notes** The track list is formatted so track times are listed relative to the first track on the disc.
3. Choose the format you want to apply to your track list information:
   - **Export as formatted text** Track information is formatted in a table. Select the **Include summary information with track list output** check box if you want to include UPC/MCN, title, engineer, and comment information with the track list.
   - **Export as delimited text** You can choose the character that will separate columns of text and choose whether to include a header row to identify the columns.
4. Click **OK** to save your file.

Printing track lists

1. From the **Edit** menu, choose **Track List**, and then choose **Print Track List** from the submenu. The Print Track List dialog is displayed.

> **Tip:** *Right-click the Track List window and choose* **Print Track List** *from the shortcut menu.*



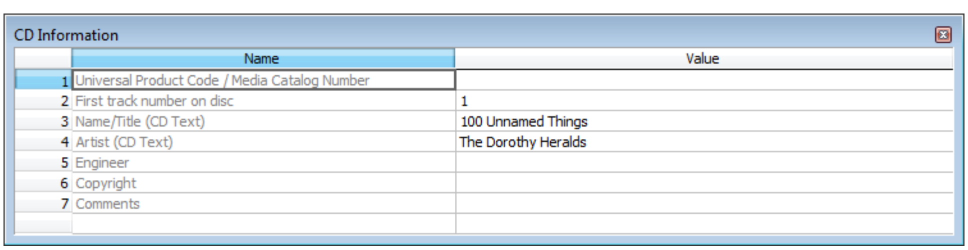
2. Choose the format you want to apply to your track list information:
   - **Cue sheet for replication** The track list is formatted so track times are listed relative to the beginning of the CD. This is the preferred format for CD replication houses.
   - **CD liner notes** The track list is formatted so track times are listed relative to the first track on the disc.
3. Select the **Include summary information with track list output** check box if you want to include UPC/MCN, title, engineer, and comment information with the track list.
4. Click OK to send the track list information to your printer.

Adding CD Text to disc-at-once CDs

When you add CD Text to your disc, title and artist information will be displayed when your CD is played on a CD player that supports CD Text.

> **Notes:**
>
> - *When you create a track using a media file that includes title and artist metadata, this information will be added to the Track List window as CD Text.*
> - *In order to burn valid CD Text, you must specify a title for the disc and for each track on the disc (artist information is optional). If the* **Title** *box in the CD Information window or Track List window is left blank, a warning will be displayed before burning so you can choose to write the disc without CD Text or cancel burning and add title information as needed.*
> - *You can write a maximum of 5000 characters as CD Text.*

1. Add title and artist information for your CD:
   a. From the **View** menu, choose **Metadata**, and then choose **CD Information** from the submenu. The CD Information window appears.
   b. In the **Name/Title (CD Text)** box, type the name of the CD.
   c. In the **Artist (CD Text)** box, type the name of the artist who performed the material on the disc.

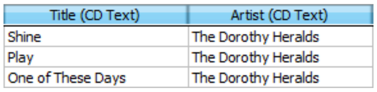


2. Add title and artist information for each track on your CD:

   a. From the **View** menu, choose **Metadata**, and then choose **Track List** to display the Track List window.

   b. Double-click the **Title (CD Text)** box and type the name of the track.

   c. Double-click the **Artist (CD Text)** box and type the name of the artist who performed the track.



Title (CD Text)	Artist (CD Text)
Shine	The Dorothy Heralds
Play	The Dorothy Heralds
One of These Days	The Dorothy Heralds

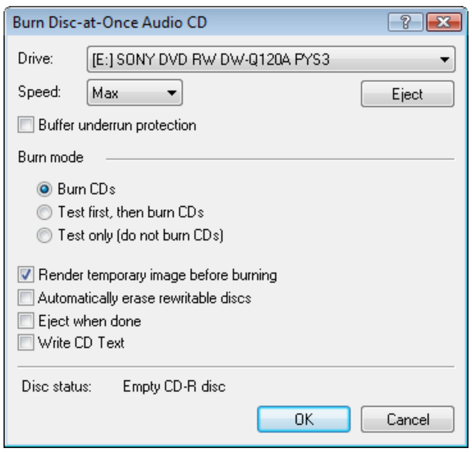
3. Burn your disc. Select the **Write CD Text** check box in the Burn Disc-at-Once Audio CD dialog to write CD Text while burning.

   When the check box is cleared, CD Text entries from the CD Information window and Track List window will be ignored.

Burning a disc-at-once CD

After you've created tracks in a disc-at-once CD project, you're ready to burn your disc. If you want to burn multiple copies, the application will prompt you to burn another copy after each disc is completed.

1. From the **Tools** menu, choose **Burn Disc-at-Once Audio CD**. The Burn Disc-at-Once Audio CD dialog is displayed.



2. From the **Drive** drop-down list, choose the CD drive that you want to use to burn your CD.

3. From the **Speed** drop-down list, choose the speed at which you want to burn. **Max** will use your drive's fastest possible speed. Decrease the setting to prevent the possibility of buffer underruns.

4. Select the **Buffer underrun protection** check box if your CD recorder supports buffer underrun protection. Buffer underrun protection allows a CD recorder to stop and resume burning.

> **Warning:** *Buffer underrun protection can create a disc that can be played in CD players but may contain a bit error where burning stopped and restarted. Consider clearing this check box when creating a premaster disc.*

5. Choose a radio button in the **Burn mode** box:

Item	Description
Burn CDs	Begins recording audio to your CD immediately.
Test first, then burn CDs	Performs a test to determine whether your files can be written to the CD recorder without encountering buffer underruns. No audio is recorded to the CD during the test, and recording begins after the test if it is successful.
Test only (do not burn CDs)	Performs a test to determine whether your files can be written to the CD recorder without encountering buffer underruns. No audio is recorded to the CD.

6. Select the **Render temporary image** before burning check box if you want to render your CD project to a temporary file before recording. Prerendering can prevent buffer underruns if you have a complex project that cannot be rendered and burned in real time.

Note: *The rendered temporary file will remain until you modify your project or exit the application. If an image file exists when you open the Burn Disc-at-Once Audio CD dialog, the check box is displayed as* **Use existing rendered temporary image**.

7. Select the **Automatically erase rewritable discs** check box if you're burning to rewritable media and want to erase the disc before burning.
8. Select the **Eject when done** check box if you want to eject the CD automatically when burning has completed.
9. Select the **Write CD Text** check box if you want to write CD Text data to your CD. In order to display CD Text, your CD player must support CD Text. The following information will be written:

Item	Source
Disc Title	**Name/Title (CD Text)** box in the CD Information window.
Disc Artist	**Artist (CD Text)** box in the CD Information window.
Track Title	**Title (CD Text)** column in the Track List window.
Track Artist	**Artist (CD Text)** column in the Track List window.

Notes:

- *In order to burn valid CD Text, you must specify a title for the disc and for each track on the disc (artist information is optional). If the* **Title** *box in the CD Information window or Track List window is left blank, a warning will be displayed before burning so you can choose to write the disc without CD Text or cancel burning and add title information as needed.*
- *You can write a maximum of 5000 characters as CD Text.*

10. Click **OK** to start burning.

Chapter 24

Optimizing Sound Forge Pro Software

This chapter contains information about configuring your system to maximize Sound Forge® Pro performance.

Defragmenting your hard drive

With time and usage, hard drives become fragmented, leading to discontiguous files and slow access. This is particularly true for older hard drives. Because Sound Forge Pro software is hard drive intensive, faster disk access equates better performance. Therefore, the initial step in improving system performance is hard drive defragmentation.

The Windows Disk Defragmenter should be run prior to using Sound Forge Pro software.

Increasing playback buffer size

The playback buffer size determines the amount of RAM used for playing from the hard drive. A buffer size of 0.10 seconds is recommended, but increased buffering may be necessary if you detect gaps during playback.

Increasing the total buffer size requires additional memory. Combined with a large preload size, this may result in a delay when starting and stopping playback.

1. From the Options menu, choose **Preferences**. The Preferences dialog appears.
2. Click the Audio tab.
3. Use the **Playback buffering slider** to configure an appropriate buffer size value and click **OK**.

   If you're using an ASIO device, click the **Advanced** button to display the Advanced Audio Configuration dialog, and then click the **Configure** button to edit your device settings.

Meters

If you experience gapping during playback and the channel meters are displayed, turn them off by choosing **Channel Meters** from the View menu.

If you experience gapping during recording and you have the **Monitor** check box selected in the Record dialog, clear the check box to turn off the record meters. *For more information, see Recording on page 143*.

Passive updating for video and time displays

If you experience gapping during playback or your computer just seems to be bogging down during playback, try turning on **Passive Update**. This will force the video and time displays to update only if there is time to do so. In most cases you won't be able to tell if it is missing some updates.

- From the Options menu, choose **Time Display**, and choose **Passive Update** from the submenu.
- From the Options menu, choose **Video**, and choose **Passive Update** from the submenu.

Audio and video synchronization

If your video has been opened from a slow device, such as a CD-ROM or network drive, Sound Forge Pro may have trouble accurately playing back the audio and video in sync. You should always copy your video files to a fast hard drive.

Following are tips that can help when trying to synchronize the audio and video:

- After assembling or editing the audio you wish to use with your video, place markers during video playback to correspond to any major synchronization points. You can locate a particular frame by dragging the cursor along the audio if the Video Preview window is open or the **Animate Video Strip** option is enabled. After primary locations have been identified, drag your audio to these markers to mix, paste, and crossfade audio.

- Features such as Insert Silence, Delete/Clear, and Time Stretch are commonly used to correct synchronization. Another useful trick is to create a region representing the offset between a video frame and audio event. Then you can enable **Lock Loop/ Region Length** and drag the offset region to a preceding silent section. Use the region as a template for adjusting the audio stream length—either copying and pasting to insert time or deleting to remove time. *For more information, see Locking loop and region lengths on page 127*.

Chapter 25

Customizing Sound Forge Pro Software

You can customize Sound Forge® Pro software to suit your project needs and working preferences. Many of the settings depend on your equipment or studio setup. Sound Forge Pro software can be set to work with the components that you use in your studio.

Saving and recalling window layouts

A window layout stores the sizes and positions of all windows and floating docks in the Sound Forge Pro workspace.

You can store any number of window layouts on your computer, and up to ten window layouts are available in the View menu (and via keyboard shortcuts) so you can quickly recall frequently used layouts. For example, you could have a layout dedicated to disc-at-once creation and another for ADR work.

Note: *Window layouts are saved in* **C:\Users\<user name>\AppData\Roaming\Sony\Sound Forge Pro\11.0**. *You can transfer layouts between computers by copying the .ForgeWindowLayout files.*

The AppData folder is not visible unless the **Show hidden files and folders** *radio button is selected on the View tab of the Windows Folder Options control panel.*

Loading default window layouts

Sound Forge software comes with three built-in window layouts: a default layout, a layout optimized for Red Book CD authoring work, and a layout optimized for 5.1 video work. To load these layouts, choose **Window Layouts** from the **View** menu, and then choose **Default**, **Stereo Recording**, **Red Book Authoring**, or **5.1 Video** from the submenu. You can also use the following keyboard shortcuts to load these layouts.

Layout	Keyboard shortcut
Default	Alt+Shift+D, D
Stereo Recording	Alt+Shift+D, R
Red Book Authoring	Alt+Shift+D, B
5.1 Video	Alt+Shift+D, V

Saving a window layout

Tip: *Press Ctrl+Alt+D, release the keys, and then press a number on your keyboard (not the numeric keypad) to save the layout in that space.*

1. Arrange the windows and docked windows as desired.
2. From the **View** menu, choose **Window Layouts**, and then choose **Save Layout As** from the submenu. The Save Layout As dialog is displayed.

   Note: *Choose* **Save Layout** *from the submenu if you want to update the current window layout. A bullet (•) is displayed next to the current layout.*

3. In the **Name** box, type the name you want to use to identify the layout. This name will be used in the Organize Layouts dialog.
4. Choose a setting from the **Shortcut** drop-down list to set the shortcut that will be used to load the layout.

   For example, if you choose 4, you could press Alt+Shift+D, release the keys, and then press 4 on your keyboard to load the layout.

5. The **Folder** box displays the path to folder where the layout will be saved.

   Window layouts are saved in the following folders by default:

   - In Windows XP: C:\Documents and Settings\<user name>\Application Data\Sony\Sound Forge Pro\11.0
   - In Windows Vista or Windows 7: C:\Users\<user name>\AppData\Roaming\Sony\Sound Forge Pro\11.0

   You can click the **Browse** button to choose a different folder.
6. Click **OK** to save the new layout.

Loading a saved layout

From the **View** menu, choose **Window Layouts**, and then choose the window layout you want to use from the submenu.

> **Tip:** *To load a layout quickly, press Alt+D, release the keys, and then press a number on your keyboard (not the numeric keypad) to recall the layout saved in that space. If no layout is saved in that space, nothing will happen when you press the shortcut keys.*

If you've modified the current window layout, choose **Window Layouts**, and then choose **Reload Selected Layout** from the submenu to reset the window layout to the last-saved version.

Adding a layout to the View > Window Layouts submenu

1. From the **View** menu, choose **Window Layouts**, and then choose **Organize Layouts** from the submenu. The Organize Layouts dialog is displayed.
2. Select a layout in the **Available layouts in current folder** box.

   This box lists the .ForgeWindowLayout files in the folder displayed in the **Current layout folder** box. If the layout you want to use is saved in a different folder, you can click the **Browse** button to choose a new folder.
3. Select a layout in the Current layouts in menu box.
4. Click the **Assign** (or **Replace**) button to add the layout to the **View > Window Layouts** submenu.

   You can click the **Move Up** or **Move Down** buttons to change the order of the layouts in the menu.
5. Click the **Activate** button to apply the selected layout to the Sound Forge Pro workspace.
6. Click **OK** to close the Organize Layouts dialog and apply your changes.

Removing a layout from the View > Window Layouts submenu

1. From the **View** menu, choose **Window Layouts**, and then choose **Organize Layouts** from the submenu. The Organize Layouts dialog is displayed.
2. Select a layout in the **Current layouts in menu** box.
3. Click the **Clear** button to remove the selected layout from the **View** > **Window Layouts** submenu.

   If you want to replace the selected layout, select a layout in the **Available layouts in current folder** box and click the **Replace** button.
4. Click **OK** to close the Organize Layouts dialog and apply your changes.

> **Note:** *Removing a layout from the* **View** > **Window Layouts** *submenu does not remove the .ForgeWindowLayout file from your computer.*

Deleting a layout from your computer

1. From the **View** menu, choose **Window Layouts**, and then choose **Organize Layouts** from the submenu. The Organize Layouts dialog is displayed.
2. Select a layout in the **Current layouts in menu** box.

3. Click the **Delete Layout** button to remove the selected layout from your computer.

> **Note:** *You cannot delete a layout that is included in the* **Current layouts in menu** *list. First, select the layout in the* **Current layouts in menu** *list and click the* **Clear** *button. Next, select the layout in the* **Available layouts in current folder** *list and click the* **Delete Layout** *button.*

4. Click **OK** to close the Organize Layouts dialog and apply your changes.

Customizing the Time Display window

From the **Options** menu, choose **Time Display** and choose an option from the submenu to adjust the Time Display window settings.

To display the Time Display window, choose **Time Display** from the **View** menu.

> **Tip:** *Right-click the Time Display window and choose an option from the shortcut menu.*

Option	Description
Position	When selected, the Time Display window shows the position of the edit or play cursor. The format of the display will depend on the Status Format setting. *For more information, see* *Selecting status formats on page 87*.
Sync/Trigger Status	When selected, the Time Display window shows incoming MIDI timecode and trigger signals. **Note:** *The current MIDI Input port is set on the MIDI/Sync tab of the Preferences dialog.*
Playlist Position	When selected, the Time Display window shows the position of the cursor (relative to the beginning of the playlist) during playback from the Playlist window. The format of the display will depend on the Status Format setting. *For more information, see* *Selecting status formats on page 87*.
CD Track Position	When selected, the Time Display window shows disc-at-once CD track numbers. In this mode, the Time Display will show track numbers and the running time for each track. Negative values indicate the pause time before a track: 

Option	Description
Record Status	When selected, the Time Display window shows status for recording. The contents of the Time Display window vary depending on the **Method** setting in the Record Options window. **Manual** • Shows **Armed** when recording is armed • Shows pre-roll time if you're recording with pre-roll. • Shows the time recorded while recording. • Shows post-roll time if you're recording with post-roll. **Automatic: Threshold** • Shows the input level (in dB) when recording is armed. • Displays the release time remaining during recording. **Automatic: MIDI Timecode** • Shows **Waiting...** when recording is armed and no MIDI timecode is being received. • Shows **Armed** when recording is armed and MIDI timecode has not started. • Shows **Armed** and counts down to the recording start time when recording is armed and MIDI timecode is being received. • Shows **Recording** and displays the time recorded during recording when no record stop time has been set. • Shows **Recording** and counts down to the recording stop time during recording when a record stop time has been set. **Automatic: Time** Counts down to the recording start time when recording is armed and counts down to the recording stop time during recording. **Tip:** *You can customize the colors used to display the armed, pre- and post-roll, and recording colors on the Display tab of the Preferences dialog. For more information, see Display tab on page 333.*
Passive Update	When selected, the Time Display window will be updated only when the processor is idle. Use this when using a slow computer to prevent the audio from glitching. The smallest increments may not be exact, but the major time increments will be accurate. Selecting this option can prevent gapping during playback on slower computers.

Setting preferences

Preferences affect how Sound Forge software functions. Any changes that you make to the preferences remain set until you change them again or reset Sound Forge software to use the default presets.

You can access the Preferences dialog by choosing **Preferences** from the **Options** menu. This dialog contains tabbed pages. The following sections explain the settings on each tab.

General tab

The **General** tab allows you to set miscellaneous Sound Forge options.

Option	Description
Open default Workspace on startup	If this check box is selected, files that were open when you last exited the program will be opened automatically.
Use Net Notify to stay informed about Sony products	When this check box is selected, information from Sony will be displayed periodically at startup. Clear the check box to bypass the Net Notify dialog.
Show logo splash-screen on startup	When this check box is selected, the Sound Forge splash screen will be displayed briefly upon startup.
Show a textured background on the Workspace	When this check box is selected, a stucco texture will be used for the application background.
Keep media files locked	Select this check box if you want to lock media files after you've opened them. Clear the check box if you want to unlock media files when you switch to another application.
Confirm on close	Select this check box if you want the application to present a confirmation message box before exiting.
Always open dropped files in new window	When this check box is selected, files that are dropped onto the Sound Forge workspace are automatically opened in a new data window.
Always proxy compressed formats	Select this check box if you want to create an uncompressed proxy (.sfap0) file when you open a compressed file format. Selecting this check box can improve performance on slower computers or for formats that cannot be decompressed quickly for real-time playback.
Ignore fact chunk when opening compressed Wave files	When this check box is selected, the software will ignore fact chunks in compressed WAV files. Compressed WAV files use fact chunks to specify how many actual samples are represented in the file. If a compressed file is improperly authored, this may cause some of the compressed data to not be loaded. If you suspect that not all sound data is being loaded from a compressed file, try checking this option and reopening the file. **Tip:** *If you change the setting of this check box, delete any proxy (.sfap0) files associated with compressed WAV files.*
Remember last-used sample rate for .vox and .ivc files	Select this check box if you want the software to remember the last-used sample rate when you open a .vox file. When the check box is cleared, you will be prompted to choose a sample rate each time you open a .vox file.
Remember last-used settings for .raw files	Select this check box if you want the software to remember the last-used settings when you open .raw files. When the check box is cleared, you will be prompted to choose a settings each time you open a .raw file.
Hide new temporary files	Select this check box if you want to turn on the **Hidden** file attribute when creating new peak (.sfk) and proxy (.sfap0) files. In the Windows Control Panel, double-click **Folder** Options and select the **View** tab. Select the **Show hidden files and folders** radio button if you want to be able to see hidden files.
Delete new temporary files on close	Select this check box if you want to delete the peak (.sfk) and proxy (.sfap0) files associated with a media file when you close a data window.
Remember last-used Save As folder	Select this check box if you want to use the last folder where you saved a file when using **Save As** or **Render As**. The first time you save a file in Sound Forge, the Documents folder is used. When the check box is cleared, files are saved to their current folder.

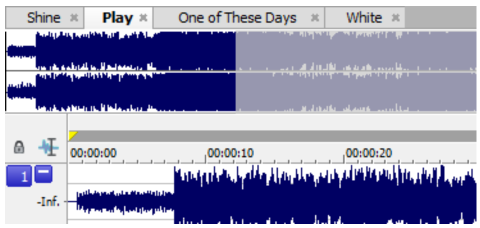
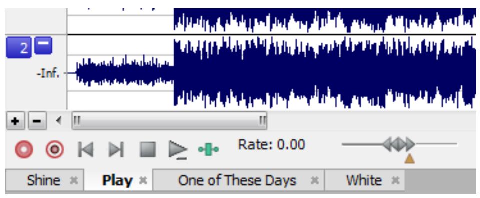
Option	Description
Allow Wave renders up to 4 GB	Select this check box to enable support for WAV files up to 4 GB. Clear the check box for compatibility with other software applications.
Warn when metadata cannot be saved in the file	Select this check box if you want to be prompted to save metadata to a separate file if it cannot be saved within the media file. When the check box is cleared, metadata will automatically be saved to a separate file if necessary.
Automatically reopen file after Save As	Select this check box if you want to automatically reopen files when you save to a different format. Changes in bit depth, channels, or compression format will result in reopening and will allow you to listen to any changes in sound quality. Clear the check box and select the **Prompt to open new file after Save As** check box if you want to be prompted to open the saved file in a new data window. When both check boxes are cleared, Sound Forge software does nothing after saving to a different format. If you're saving a file to several compressed formats, clearing these check boxes prevents you from having to reopen the file after saving each format.
Prompt to open new file after Save As	When the **Automatically reopen files after Save As** check box is cleared, select this check box if you want the application to prompt you to open the destination file to a new data window after saving a sound file to a different format. Opening the file in a new data window will allow you to hear any changes in quality between the original file and the result of the Save As operation.
Allow Undo past Save	When this check box is selected, your undo history is maintained until you close the data window (or exit the application) so you can undo edit operations even if you've saved your file. When this check box is selected, quick file saving may not be available. **Tip:** *If you want to be able to undo edit operations even after closing and reopening your file, save a Sound Forge project.*
Show shuttle controls on Data Window transport	When this check box is selected, **Rewind** and **Forward** buttons will appear on each data window's transport controls.
Show record controls on Data Window transport	When this check box is selected, **Arm** () and **Record** () buttons will appear on each data window's transport controls.
Show waveform while recording	When this check box is selected, the waveform is drawn while you're recording audio.
Show free storage space on Status Bar	When this check box is selected, the total amount of free disk space available on your specified temporary drive is displayed on the status bar. Use the **Temporary files and record folder** box at the bottom of the General tab to set the folder that will be used for temporary files and recorded data.
Spacebar and F12 Play/Pause instead of Play/Stop	Select this check box if you want the F12 and spacebar keyboard shortcuts to toggle between Play and Pause mode. In this mode, the cursor will maintain its position.
Default to slow scroll when drag selecting	In some very fast computers, automatic scrolling while selecting is too fast to use accurately. When this option is turned on, drag-selecting will cause a slow scroll. **Tip:** *Click the right mouse button while selecting to toggle slow scrolling.*
Allow Ctrl+drag style cursor scrub in data windows	When this check box is selected, you can hold Ctrl while dragging the cursor to scrub in data windows. **1.** Hover over the cursor and press Ctrl. The mouse pointer changes to a pan/scrub icon when you click.  **2.** Drag left or right to scrub playback.
Allow Ctrl+drag style zoom in data windows	When this check box is selected, you can hold Ctrl and drag in a data window to zoom to a selection. **Tip:** *When the check box is selected, you can still use Ctrl+drag to paste a selection; press and hold the Ctrl key after you start dragging the selection.*

Option	Description
Prompt for marker and region names	When this check box is selected, an edit box is displayed so you can name markers and regions as you place them.
Warn when Paste or Mix formats do not match	Select this check box if you want to be warned before pasting or mixing data that has different sample rates or bit depths. Pasting or mixing data of different formats may produce unintended results.
Autoupdate BWF Origination Time Reference	Select this check box if you want to update **OriginationTimeRef** metadata when adding or deleting sound data at the beginning of a Broadcast Wave Format file. When the check box is cleared, **OriginationTimeRef** metadata is unaffected when you add or delete sound data at the beginning of the file. *For more information about Broadcast Wave Format metadata, please see Broadcast Wave window (Ctrl+Alt+M, 4) on page 55.*
Auto-power MIDI keyboard window	Select this check box if you want to open the MIDI device assigned to the MIDI keyboard (if it is not already open) when you click a key on the MIDI keyboard. You may want to turn off this option if you are using the same MIDI output device for MIDI synchronization or for your sequencer. If this option is turned off, you need to click the **On** button on the keyboard prior to using it to send notes.
Allow floating windows to dock	When this check box is selected, windows will automatically be docked when you drag them to the edges of the Sound Forge workspace. You can hold the Ctrl key while dragging a window to prevent it from docking. When this check box is cleared, windows will not dock unless you hold the Ctrl key.
Recently used file list	Select the check box if you want to display a list of recently used files on the File menu. Use the edit box to specify the number of files you want to display.
Temporary files and record folder	Specify a folder for storing temporary files and recorded audio, or click the **Browse** button to specify a new folder. Using temporary file space allows you to edit very large files and keeps Sound Forge from using large portions of RAM on your computer. Your temporary directory must have enough space to accommodate the total size of all files you plan to edit along with space for any clipboard data and undo buffers. If you change the temporary storage folder, you will have to restart Sound Forge for the change to take effect.

Display tab

The **Display** tab allows you to specify options for adjusting the appearance of the Sound Forge window.

Description	Option
Default sound file window height	Drag the slider to specify the default data window height for a sound file. This magnification level is used when you load a sound file or create a new window.
Default video strip height	Drag the slider to specify the default height of the video strip when you open a video file.
Peak ratio default for new sound files	Choose a ratio from the drop-down list to specify the zoom ratio above which the application will use a peak file instead of the original file to draw the waveform. If you notice problems with waveform scrolling, try decreasing this setting so it is less than your current zoom ratio. To calculate the size of the resulting peak files, divide the size of the file by the peak ratio. For example, a 100 MB sound file will need a 0.39 MB (100/256) peak file when using 1:256.
Normal zoom ratio	Choose a zoom ratio from the drop-down list to specify the default horizontal magnification. This magnification level is used when you load a sound file, create a new window, or use the **Zoom Normal** command. High values show more data, and small values show more detail.

Description	Option
Custom zoom ratio 1 **Custom zoom ratio 2**	Choose a zoom ratio from the drop-down list to specify a custom level of horizontal magnification. This zoom ratio will be used when you perform any of the following actions: • Click the **Custom Zoom 1** or **Custom Zoom 2** button on the Navigation toolbar. • From the **View** menu, choose **Zoom Time**, and then choose a **Custom Zoom** command from the submenu. • Right-click in a data window, choose **Zoom** from the shortcut menu, and then choose a **Custom Zoom** command from the submenu.
Tabs for maximized data windows	Choose a setting from the drop-down list to choose whether you want to display tabs to help you browse maximized data windows: • Choose **None** if you do not want to display tabs. You can navigate data windows by choosing a window from the **Window** menu or by pressing Ctrl+Tab. • Choose **Top** to display tabs above the waveform display.  • Choose **Bottom** to display tabs below the waveform display.  When tabs are displayed, you can click a data window's tab to bring it to the foreground. When you have multiple data windows open and maximized, you can drag files to a specific data window. While dragging a file or selection, hover over a data window's tab to bring it to the foreground. You can then drop the file or selection in the desired data window.

Color preference for	The color preferences section allows you set a custom color for a variety of graphics within the Sound Forge interface.
	1. Choose a screen element from the **Color preference for** drop-down list.
	2. Set the color of the selected item:
	  • Drag the **Hue** slider to change the color of the selected item.
	  • Drag the **Saturation** and **Brightness** sliders to adjust the intensity of the selected color.
	
	  Note: *When adjusting the display color for channel waveforms, the* **Saturation** *and* **Brightness** *sliders are not available. To adjust saturation and brightness for all channels, choose* **Wave: All Channels** *from the* **Color preference for** *drop-down list and then adjust the controls.*
	
	  • Click the **Default** button to restore a custom color to the default setting.
	3. Click the **OK** button.

Icon color saturation	Drag the slider to adjust the color intensity of icons in the Sound Forge window. Drag to the left to decrease the color saturation, or drag to the right to increase it.

Description	Option
Icon color tint	Drag the slider to adjust the amount of tinting that is applied to the icons in the Sound Forge window. Drag the slider to the right to add an average of the title bar colors to the icons. Drag to the left to decrease the amount of tinting applied. **Tip:** *You can use the Personalization control panel to change your active window title bar colors.* • *In Windows Vista, open the Personalization control panel and click* **Window Color and Appearance***. In the Appearance Settings dialog, click the* **Advanced** *button and choose* **Active Title Bar** *from the* **Item** *drop-down list.* • *In Windows 7, open the Personalization control panel and click* **Window Color***. In the Window Color and Appearance dialog, choose* **Active Title Bar** *from the* **Item** *drop-down list.*

Editing tab

The **Editing** tab allows you to specify preferences for editing and undo operations.

Option	Description
Disable triple-clicking to select all sound file data	Select this check box if you don't want to select all data when you triple-click in a data window. You might want to select this option if triple-clicks are falsely detected when you make a selection and then try to perform a drag operation. Otherwise, decrease Windows' double-click threshold time. When this check box is cleared, you can triple-click anywhere in a data window to select all data.
Disable auto-snapping below 1:4 zoom ratios	Select this check box if you do not want selections to snap to time or zero-crossings when the data window zoom ratio is less than 1:4. This is useful if you commonly zoom in fully to adjust selection points manually yet still want to use automatic snapping when zoomed out.
Force loop bar to match selection	Select this check box if you want the loop region to always match the current time selection. Clicking to position the cursor in a data window will clear the loop region. When no loop region exists and looped playback is enabled, the entire data window will play looped. When the check box is selected, the behavior is similar to Sound Forge 8.0. Clear the check box if you want to be able to position the cursor without clearing the loop region.
Update loop bar on Mark In/Out	Select this check box if you want the loop bar in a data window to be updated when you mark the beginning or end of a selection. When the check box is cleared, the loop bar isn't updated until after you've marked both ends of the selection. This check box is not available when **Force loop bar to match selection** is selected.
Auto-crossfade Mix with selection	When this check box is selected, the Fade In and Fade Out settings for the Mix tool will pay attention to the destination selection and file length when mixing between files. This setting has no effect in the following situations: • When the material you are mixing does not overlap either end of the destination selection or the end of the destination file. • When no selection exists in the destination. • When you mix data within a single data window. When the check box is cleared, Fade In and Fade Out setting are not affected by the selection in the Mix destination.
Drag & drop auto rise delay	Drag this slider to specify the time before a window underneath the cursor becomes active during drag-and-drop operations.

Option	Description
Snap to zero-crossing slope	Use this drop-down list to specify how zero-crossings are detected when you choose Snap to Zero: • **Negative Slope** Zero-crossings are detected only on a negative slope. • **Any Crossing** Zero-crossings are detected on both positive and negative slopes. • **Positive Slope** Zero-crossings are detected only on a positive slope. **Tip:** *It is usually best to use either* **Positive Slope** *or* **Negative Slope** *so that noticeable pops and clicks are not generated by cutting data.*
Zero-cross scan time	Specify the maximum time (in samples) that will be used to search for the next zero-crossing.
Zero-cross level threshold	Specify the sample value below which data will be considered a zero-crossing. **Note:** *Setting this value above zero can compensate for DC offset. However, if possible, you should remove DC offset first.*
Pencil tool maximum zoom ratio	Choose a setting from the drop-down list to specify the maximum zoom ratio at which the Pencil tool will be available.
JKL / shuttle speed	Choose a setting from the drop-down list to set the speed that will be used for scrubbing the timeline with the JKL keys or with a multimedia controller.
Global media cache	Specify the amount of RAM you want Sound Forge Pro to reserve for media recently read from or written to disk. Reserving excessive amounts of RAM may decrease overall performance. To turn off the cache, choose 0.
Wet Gain (dB)	Type a value in the **Wet Gain** box (or use the spinner) to set the default level of the processed signal that will be mixed into the output.
Dry Gain (dB)	Type a value in the **Dry Gain** box (or use the spinner) to set the default level of the unprocessed signal that will be mixed into the output.
Fade In	Type a value in the **Fade In** box (or use the spinner) to set the default length of the fade in between the processed and unprocessed signal. Click the **Fade Curves** button ( ) and choose a curve type from the menu to set the speed of the fade in.
Fade Out	Type a value in the **Fade Out** box (or use the spinner) to set the default length of the fade out between the processed and unprocessed signal. Click the **Fade Curves** button ( ) and choose a curve type from the menu to set the speed of the fade out.

Labels tab

The **Labels** tab allows you modify the default names that are assigned to data windows, regions, and markers.

Editing default data window names

The Window labels section of the **Labels** tab allows you to modify the names that are assigned to new data windows when you create a new data window or choose **Create a new window for each take** from the **Mode** drop-down list in the Record dialog.

1. From the **Options** menu, choose **Preferences**, and click the **Labels** tab.
2. Select the **New window prefix** check box and type a prefix in the box if you want to display a name in the new window's title bar. Clear the check box if you do not want to include a prefix (if you want to number windows only, for example).
3. Select the **Use counter and start at** check box and type a number in the box if you want to number new data windows.
4. Select the **Insert leading zeros in field width of** check box and specify a field width if you want to use leading zeros in window names. For example, if you specify a field width of 3, windows numbered 1 to 99 would be numbered 001 to 099.
5. Click the **OK** button.

Editing default region names

The Region Labels section of the **Labels** tab allows you to modify the names that are assigned to regions when you insert regions or choose **Multiple takes creating regions** from the **Mode** drop-down list in the Record dialog.

1. From the **Options** menu, choose **Preferences**, and click the **Labels** tab.
2. Select the **Label Regions** check box to display text labels for regions in the data window when you insert regions or choose **Multiple takes creating regions** from the **Mode** drop-down list in the Record dialog.
3. Adjust additional settings as necessary:

Item	Description
Prefix	Type a prefix in the box if you want to assign a name to new regions. Clear the check box if you do not want to include a prefix (if you want to number regions only, for example).
Use counter and start at	Select this check box and type a number in the box if you want to number new regions.
Insert leading zeros in field width of	Select this check box and specify a field width if you want to use leading zeros in region names. For example, if you specify a field width of 3, regions numbered 1 to 99 would be numbered 001 to 099.

4. Click the **OK** button.

Editing default marker names

The Marker labels section of the **Labels** tab allows you to modify the names that are assigned to markers when you insert markers during playback or recording.

1. From the **Options** menu, choose **Preferences**, and click the **Labels** tab.
2. Select the **Label Markers** check box to display text labels for markers in the data window when you insert markers.
3. Adjust additional settings as necessary:

Item	Description
New marker prefix	Type a prefix in the box if you want to assign a name to new markers. Clear the check box if you do not want to include a prefix (if you want to number markers only, for example).
Use counter and start at	Select this check box and type a number in the box if you want to number new markers.
Insert leading zeros in field width of	Select this check box and specify a field width if you want to use leading zeros in marker names. For example, if you specify a field width of 3, markers numbered 1 to 99 would be numbered 001 to 099.

4. Click the **OK** button.

File Types tab

The **File Types** tab allows you to indicate which types of files you want to associate with Sound Forge software. When file is associated with Sound Forge software, you can double-click a sound file in the Windows Explorer and it will open for editing.

1. Select a file type from the list. The **File association details** box near the bottom of the tab displays information about the selected file type, as well as the current association.
2. Select the check box for each sound file format you want to associate with Sound Forge software, or clear the check box to remove a file association.
3. Click the **OK** button.

MIDI/Sync tab

The **MIDI/Sync** tab allows you to specify preferences for MIDI and synchronization.

Item	Description
Output	Choose a MIDI device from the drop-down list to specify the MIDI output device for synchronization when Generate MIDI Timecode is enabled.
Input	Choose a MIDI device from the drop-down list to specify the MIDI input device for synchronization and triggering when Trigger from MIDI Timecode is enabled. This is the device through which Sound Forge will receive all MIDI triggering and synchronization input, including SMPTE/MTC, MIDI triggers, and Regions/Playlist triggers.
Bound record time on SMPTE record sync	When this check box is selected, Sound Forge software will not allow recording beyond the specified end time. This ensures that your record length is exact regardless of any inaccurate timecode.
Use internal timer for SMPTE generation	Select this check box if you want to use the internal timer for SMPTE generation rather than position values reported by the sound card driver. Since many sound cards do not report their position accurately, it is usually better to use the internal timer for SMPTE generation. Choose a value from the **Internal timer resolution** drop-down list to specify the internal timer accuracy used for generating SMPTE. Low values produce more accurate SMPTE generation, but may also decrease system performance.
Use free-wheel for SMPTE loss	Select this check box to stop playback if the incoming MIDI timecode signal stops. When this check box is not selected, Sound Forge playback will continue until the user stops playback manually. In the **Free-wheel time** box, specify the amount of time that Sound Forge playback will continue after the incoming MIDI timecode signal stops. If timecode starts again during this time, playback will continue. In the **Free-wheel slack** box, specify how fast the software should expect timecode updates before going into Free-wheel mode. If you have a fast computer, this value can be set to a lower value if you want to stop playback immediately when timecode is interrupted.
Apply offset to generated SMPTE	Select this check box to specify an offset that will be added to the time displayed in the Sound Forge play counter. For example, if you want to generate MIDI timecode starting at 01:00:00:00, instead of inserting 1 hour of silence at the beginning of your sound file, you can specify that amount in this box. When using Record Sync, you'll often want to set this value to the **Enable MTC/SMPTE Input Synchronization Start** time. The Sound Forge ruler and play counter will not display this offset.

Previews tab

The **Previews** tab allows you to specify options for previewing files.

Item	Description
Limit non-realtime previews to	Select this check box and specify the length of audio that will be used when generating a preview. Low values decrease the amount of time needed to generate a preview when tuning effects or processing values.
Pre-roll	Select this check box and specify how many seconds of unprocessed audio will be played before the processed selection. Use this to listen to the transition from unprocessed to processed data. **Tip:** **Pre-roll** *can be toggled on and off by right-clicking the dialog and choosing* **Pre-roll** *from the shortcut menu.*
Post-roll	Select this check box and specify how many seconds of unprocessed audio will be played after the processed selection. Use this to listen to the transition from processed to unprocessed data. **Tip:** **Post-roll** *can be toggled on and off by right-clicking the dialog and choosing* **Post-roll** *from the shortcut menu.*
Reactive previewing	Select this check box to automatically recalculate and play back the preview buffer if the parameters of an effect change. This allows for immediate feedback of the effects of a change. This option is most useful when using a fast computer, limiting preview times, and not using processor-intensive effects.
Audio event locator: **Pre-roll/Loop time**	Use the **Pre-roll** and **Loop time** settings to control how the audio event locator plays audio: • In the **Pre-roll** box, specify the amount of data played prior to the cursor position. • In the **Loop time** box, specify the amount of time that will loop when you stop the cursor while clicking and dragging in the overview bar. To use the audio event locator, hold Ctrl, click the overview bar, and drag the mouse. Similar to a scrub control, playback follows mouse movement and loops around the cursor position when the mouse is still. Playback stops when the mouse button is released.
Cut preview configuration: **Pre-roll/Post-roll**	Use the **Pre-roll** and **Post-roll** settings to control the amount of data that is played back when you choose **Preview Cut/Cursor** from the **Transport** menu: • In the **Pre-roll** box, specify the amount of data played prior to the selection or cursor position. • In the **Post-roll** box, specify the amount of data played after the selection or cursor position.
Play Looped adjust pre-roll	When **Loop Playback** mode ( ) is turned on and you make a selection during playback, playback is pre-rolled from the end of the selection to help you tune long loops. Specify the number of seconds before the end of the selection that you would like to pre-roll.
Playlist pre-roll	Enter a value in the edit box or use the up and down arrows to specify the amount of pre-roll that will be used when playing entries in the Playlist/Cutlist window. This allows you to easily hear the transition from one region to another without having to play all the way through the first region.

Status tab

The **Status** tab allows you to specify preferences for displaying information in the status bar.

Item	Description
Default frames per second	The default frame rate used to calculate frame values. Frame values are useful when trying to synchronize sound with animation. Most animation players specify a playback frame rate at which video frames are shown to the user. If you are using an animation that has a frame rate of 15.0 frames per second, you would set the frame rate to 15.0. When status values are displayed, they will be shown in values of frames. This allows you to find the frame to which a given point in the sound file corresponds.
Default beats per measure	The number of beats in each measure for displaying in measures and beats. For example, 2/4 time would have two beats per measure. This setting will be also be used in the Edit Tempo dialog.
Default beats per minute	The number of beats per minute, that is, the tempo of a song for displaying lengths. This setting will be also be used in the Edit Tempo dialog.
RMS level scan time	The amount of sound data surrounding the cursor used to calculate the RMS level in the Levels toolbar.
Peak level scan time	The amount of sound data surrounding the cursor used when searching for a peak level to display in the Levels toolbar.
0 VU (+4 dBu) level	Choose a setting from the drop-down list or type a value in the box to calibrate the VU/PPM meters to their associated levels on the peak meters. VU meters display sound in dB VU, where 0 VU is a reference level, and there is headroom above 0 VU. The Sound Forge peak meters display peaks in dB FS (decibels relative to full scale). In digital audio, there is no headroom above 0 dB FS. Choosing a setting from this drop-down list subtracts a nominal dB value from the VU meters so that a signal displayed on the VU meters remains slightly below 0 dB on the peak meters.
VU meter integration time	Type a value in the box to set the amount of data surrounding the cursor that will be used to calculate levels in the VU meters. This setting has no effect on the PPM scales, which use fixed integration times: • UK PPM: 10 ms • EBU PPM: 10 ms • DIN PPM: 5 ms • Nordic PPM: 5 ms
Enable surround processing for files with 6 channels	Select this check box if you want to treat audio with six or more channels as surround audio when measuring loudness (a gain of ~1.5 dB is applied to the left and right surround channels). When the check box is cleared, all channels contribute equally to the loudness measurement.
Open editor when new loudness log is generated	Select this check box of you want to automatically open loudness log files in your default text editor when you choose **Tools** > **Generate Loudness Log**. *For more information, see Generating a loudness log on page 141*.

Toolbars tab

The **Toolbars** tab allows you to specify which toolbars you want to display.

Display or hide toolbars

Select the check box to display a toolbar; clear a check box to hide a toolbar.

Display or hide ToolTips

Select the **Show ToolTips** check box if you want to display pop-up descriptions when the mouse is held over certain items.

Customizing a toolbar

1. From the **View** menu, choose **Toolbars**. The Preferences dialog appears with a list of available toolbars.
2. Select the check box for a toolbar and click **Customize**. The Customize Toolbar dialog is displayed.
3. Use the controls in the Customize Toolbar dialog to add, remove, or rearrange the buttons on the selected toolbar. Click **Reset** to restore the toolbar to its default setting.
4. Click the **OK** button.

CD Settings tab

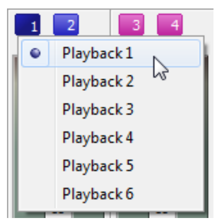
The CD Settings tab allows you to specify settings for burning and extracting audio from CDs.

Item	Description
Use strict Red Book specification for DAO validation	Select this check box if you want to be notified prior to burning a disc-at-once CD if anything about your CD project is against strict Red Book standards. These warnings are not critical, and in most cases you will not write an unreadable disc if you proceed. Clearing this check box will not suppress critical warnings that will result in an unreadable disc.
Use SPTI Direct for CD burning	Select this check box if you want to use SPTI (SCSI Pass-Through Interface) to communicate with your CD burning drive.
Overwrite CD Text autodetection results	If the software incorrectly detects that your CD recorder is not able to write CD Text, select this check box to turn on CD Text writing for your drive. **Note:** *Check the documentation provided with your CD recorder to determine whether the drive is able to write CD Text.*
Include Wide SCSI devices when searching for drives	Select this check box if you want to scan for wide SCSI CD drives when you attempt to extract data from or burn CDs. When the check box is cleared, the application will not scan for wide SCSI devices, which can increase compatibility with some USB device drivers that incorrectly identify themselves as wide SCSI.
Skip drive database; autodetect drive capabilities on startup	When the check box is cleared, an internal configuration file is used to determine your drive's capabilities. If you encounter problems burning CDs, select this check box, and your drive to determine its capabilities when the application starts.
Automatically detect CD length	Select this check box if you want to automatically detect the length of your blank CDs when you insert them in your drive. When this check box is cleared, the **Default CD length (minutes)** setting is used, and you can click the **CD Time Remaining** box in the status bar to update the available time on a CD.
Default time between CD tracks (seconds)	Type a value in the edit box to specify the length of time that is inserted between disc-at-once CD tracks.
Default CD length (minutes)	Type the default length for CD media. This length is used if the software has not yet scanned your CD drive or if no CD is in your drive. This length is used to calculate the amount of time remaining on the disc that is displayed in the status bar.

Audio tab

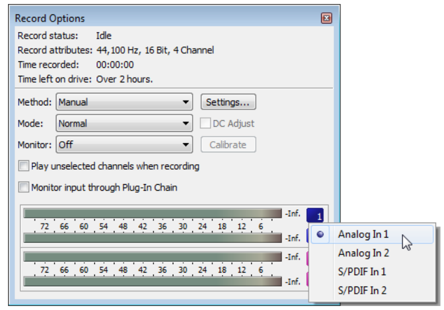
The **Audio** tab allows you to specify playback and recording options.

Basic audio preferences

Item	Description	
Audio device type	Choose a driver type from the drop-down list. • **Microsoft Sound Mapper** - The default setting. Allows the Sound Mapper to choose appropriate playback and recording devices. • **Windows Classic Wave** - Allows you to choose a specific audio device using a classic Wave driver. • **ASIO** - Allows you to choose a specific audio device using a low-latency ASIO driver.	
Playback	Click the Playback tab to adjust playback routing and buffering settings.	
	Playback device routing	The Channel and Device columns indicate which audio output will be used to play each channel in a multichannel file. To assign a channel to a different output, click the Device entry and choose a new output from the drop-down list. **Tip:** *To change a channel's output device using the Channel Meters window, click the channel number and choose a new output port from the menu:* 
	Playback buffering (seconds)	Specifies the total amount of buffering that is used during playback. The larger the number, the more buffering is performed during playback. This value must be as low as possible without gapping. To set it, start at .25 and play back a typical song. Move some of the track faders. If the playback gaps, try increasing this slider in small increments until the gapping goes away. If you simply cannot get playback to be free of gapping, you need to install more RAM in your computer so you can increase buffering, buy a faster access hard drive, or minimize the number of audio plug-ins you are trying to use simultaneously.
Record	Click the Record tab to adjust record input routing and buffering settings.	

<table>
<tr><th>Item</th><th colspan="2">Description</th></tr>
<tr><td></td><td>Recording device routing</td><td>The Channel and Device columns indicate which audio output will be used to record each channel in a multichannel file. To assign a channel to a different input, click the Device entry and choose a new input from the drop-down list.

Tip: To change a channel's input in the Record Options window, click the channel number and choose a new input port from the menu:


</td></tr>
<tr><td></td><td>Record buffering (seconds)</td><td>Specifies the total amount of buffering that is used during recording.

If you use your computer for other tasks while recording, increasing this setting can reduce the likelihood of those tasks interrupting recording.</td></tr>
<tr><td>Advanced</td><td colspan="2">Click this button to open the Advanced Audio Configuration dialog. For more information on these options, see below.</td></tr>
<tr><td>Default All</td><td colspan="2">Click to restore the Audio tab to the default settings.</td></tr>
</table>

Advanced audio preferences

You can click the **Advanced** button on the **Audio** tab to access the advanced audio preferences.

Setting	Description
Audio devices	This list contains all of the audio devices that are installed in your computer. Select a device from the list to set the options below for that device.
Interpolate position	When this check box is selected, the software will attempt to compensate for inaccurate devices by interpolating the playback or recording position. If you notice that your playback cursor is offset from what you are hearing, enable this option for the playback device.
Position bias	If the position of playback or recording does not match what you hear after you enable **Interpolate position**, you can attempt to compensate using the **Position bias** slider. Moving this slider will offset the position forward or backward to compensate for the inaccuracies of the device.
Do not pre-roll buffers before starting playback	When this check box is selected, the software will not create buffers prior to starting playback. Some devices do not behave properly if this check box is cleared. If your audio stutters when you start playback try selecting this check box.
Audio buffers	Drag the slider to set the number of audio buffers that will be used. Adjusting this setting can decrease gapping or help you synchronize the input and output.
Buffer size	Choose a setting from the drop-down list to indicate the buffer size you want to use. Choose **MME** to use the **Playback buffering** setting on the **Audio** tab in the Preferences dialog. For example, if you choose **MME** from the **Buffer size** drop-down, set the **Audio buffers** slider to 5, and set **Playback buffering** to 0.35 seconds, five 0.07-second buffers are created. If you choose 1024 from the **Buffer size** drop-down and set the **Audio buffers** slider to 5, five 1024-byte buffers are created.
Priority	Choose a setting from the drop-down list to set the priority that is assigned to your audio buffers. Increasing the buffers' priority can help you attain smoother playback, but it can also adversely affect other processes.

Video tab

Use the **Video** tab to specify preferences for displaying video.

Item	Description
Frame numbering on thumbnails	Determines how individual frame information, located in a box at the lower left-hand corner of each frame, will be displayed in the video strip when frame numbering is turned on. The frame information box can include **Frame Numbers** or **Media Timecode**.
Allow pulldown removal when opening 24p DV	Select this check box if you want to remove pulldown when you open 24 fps progressive-scan DV video files. When the check box is cleared, Sound Forge software will read 24p video as 29.97 fps interlaced video (60i).
Deinterlace method	Choose a setting from the drop-down list to determine how Sound Forge software separates the two fields that make up a video frame when you render to a progressive format: • **Blend Fields** Maintains the data in the two fields by blending them together. This method can produce a smooth, motion-blurred image. • **Interpolate** Deletes one field and uses the remaining field to interpolate the deleted lines. This produces sharper images than **Blend Fields** but can introduce jagged motion or stair-stepping artifacts.
Resample source video when rendering to a higher frame rate	Select this check box if you want to interpolate video frames when you render to a frame rate that is greater than the source file's frame rate.

Item	Description
External monitor device	Choose a device from the drop-down list to configure an IEEE-1394 device for use with an external monitor. Sound Forge will send your video output to this device when you click the **External Monitor** button () in the Video Preview window. More information on this device are displayed in the **Details** pane. You can make additional preview playback adjustments near the bottom of the Video tab once you've selected an external monitor device: • If necessary, you can adjust the video to display properly on your external monitor. Choose the desired format from the drop-down list. • If audio and video do not play back in synchronization, drag the **Sync offset** slider to specify a frame offset to restore synchronization.
Details	Displays information about the device selected in the **External monitor** drop-down list.
If project format is invalid for DV output, conform to the following	If your source media does not conform to DV standards, choose a setting from the drop-down list. The video is adjusted to display properly on your external monitor.
Sync offset (frames)	If your audio is not synchronized with your external monitor, you can configure an offset for your hardware. Drag the slider to synchronize audio and video. This setting affects synchronization for previewing on an external monitor. Audio and video synchronization in the Sound Forge workspace is not affected.

VST Effects tab

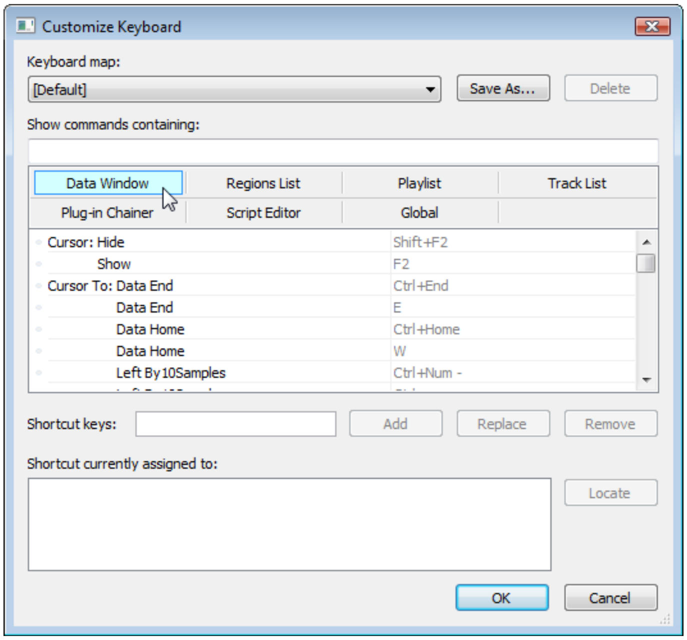
The **VST Effects** tab allows you to specify where your VST plug-ins are installed.

Item	Description
Default VST search folder	This is the folder in which the application looks for VST effects during startup.
Alternate VST search folder 1	Enter a path in the edit box or click **Browse** to indicate where the application can find VST effects.
Alternate VST search folder 2	Enter a path in the edit box or click **Browse** to indicate where the application can find VST effects.
Select VST effects to be available as audio plug-ins	Lists the VST effects that are currently available. Select a VST effect's check box to make it available for use as a plug-in. **Note:** *When you use a VST plug-in, the software will lock it for the remainder of your session. A lock is displayed next to the check box to indicate that the plug-in cannot be removed until you close and restart the application.*

Customizing keyboard shortcuts

From the **Options** menu, choose **Customize Keyboard** to customize the keyboard shortcuts available in the Sound Forge interface.

The **Shortcut keys** box displays the currently assigned shortcut keys for the selected command. Click a tab in the middle of the dialog to choose which shortcuts you want to see.



Editing or creating shortcuts

1. From the **Options** menu, choose **Customize Keyboard**. The Customize Keyboard dialog appears.
2. Click a tab in the middle of the dialog to indicate the type of command you want to assign to a keyboard shortcut.
3. Select a command in the list.

> **Tip:** *You can type a word in the* **Show commands containing** *box to filter the list of commands to display only commands that contain the word you typed.*

4. Click the **Shortcut keys** box and press the key combination you want to assign to the selected command.

> **Tip:** *If you type a key combination that has already been assigned to another command, the* **Shortcut currently assigned to** *box displays the existing assignment. To find the existing command, click the* **Locate** *button.*

5. Click the **Add** button to assign the key combination in the **Shortcut keys** box to the selected command.

Saving a keyboard mapping

1. From the **Options** menu, choose **Customize Keyboard**. The Customize Keyboard dialog is displayed.
2. Click the **Save as** button and type a name to save your current keyboard shortcuts to an .ini file in **C:\Users\<user name>\AppData\Roaming\Sony\Sound Forge Pro\11.0**.

> **Note:** *The AppData folder is not visible unless the* **Show hidden files and folders** *radio button is selected on the View tab of the Windows Folder Options control panel.*

You can use this file as a backup or to share your keyboard shortcuts with other Sound Forge users.

Deleting a keyboard mapping

1. From the **Options** menu, choose **Customize Keyboard**. The Customize Keyboard dialog is displayed.
2. Choose a mapping from the **Keyboard map** drop-down list and click the **Delete** button to remove the selected keyboard mapping.

> **Note:** *You cannot delete the default Sound Forge keyboard mapping.*

Importing or renaming a keyboard mapping

Copy a Sound Forge keyboard mapping .ini file to **C:\Users\<user name>\AppData\Roaming\Sony\Sound Forge Pro\11.0**.

> **Note:** *The AppData folder is not visible unless the* **Show hidden files and folders** *radio button is selected on the View tab of the Windows Folder Options control panel.*

The next time you start Sound Forge, the new keyboard mapping will be available from the **Keyboard map** drop-down list in the Customize Keyboard dialog.

> **Tip:** *If you want to edit a the name used to identify a keyboard mapping in the Customize Keyboard dialog, open the .ini file in a text editor and change the* **<Display Name>** *portion of the* **Name=<Display Name>** *entry. Save the .ini file and restart Sound Forge to use the new name.*

Resetting the default keyboard mapping

1. From the **Options** menu, choose **Customize Keyboard**. The Customize Keyboard dialog is displayed.
2. Choose **[Default]** from the **Keyboard map** drop-down list and click **OK** to restore the default configuration.

Appendix A

Shortcuts

This appendix contains information about the shortcuts you can use to make editing in Sound Forge® Pro software quicker and easier.

Keyboard shortcuts

The following shortcuts represent the default configuration. Your system may differ if you've used the Customize Keyboard window to customize your keyboard shortcuts. *For more information, see Customizing keyboard shortcuts on page 346*.

The available shortcut keys are arranged in tables according to function.

Project file shortcuts

Command	Keyboard Shortcut
Create a new data window	Ctrl+N
Create a new data window without displaying the New Window dialog	Ctrl+Shift+N
Open a sound file or project	Ctrl+O
Save modified sound data back to the file	Ctrl+S
Display File Properties window	Alt+Enter
Close the active data window	Ctrl+W
Exit Sound Forge	Alt+F4

Magnification and view shortcuts

Command	Keyboard Shortcut
Set input focus to the waveform display in the active window	Alt+0
Show/set input focus to the Explorer window	Alt+1
Show/set input focus to the File Properties window	Alt+2 —or— Alt+Enter
Show/set input focus to the Video Preview window	Alt+3
Show/set input focus to the Time Display window	Alt+4
Show/set input focus to the Channel Meters window	Alt+5
Show/set input focus to the Loudness Meters window	Alt+6
Show/set input focus to the Hardware Meters window	Alt+7
Show/set input focus to the Undo/Redo History window	Alt+8
Show/set input focus to the Spectrum Analysis window	Alt+9
Show/set input focus to the Plug-In Chain window	Ctrl+Alt+0
Show/set input focus to the Plug-In Manager window	Ctrl+Alt+1
Show/set input focus to the Keyboard window	Ctrl+Alt+2
Show/set input focus to the Script Editor window	Ctrl+Alt+3
Show/set input focus to the Loop Tuner window	Ctrl+Alt+4
Show/set input focus to the Record Options window	Ctrl+Alt+5
Show/set input focus to the Regions List	Ctrl+Alt+M, 0
Show/set input focus to the Playlist/Cutlist window	Ctrl+Alt+M, 1
Show/set input focus to the Track List window	Ctrl+Alt+M, 2
Show/set input focus to the ACID Properties window	Ctrl+Alt+M, 3
Show/set input focus to the Broadcast Wave window	Ctrl+Alt+M, 4
Show/set input focus to the CD Information window	Ctrl+Alt+M, 5
Show/set input focus to the Sampler Loops window	Ctrl+Alt+M, 6
Show/set input focus to the Summary Information window	Ctrl+Alt+M, 7
Tile the data windows vertically	Shift+F4
Restore the Sound Forge application window	Alt+F5
Recall window layout	Alt+Shift+D, then press 0-9
Save window layout	Ctrl+Alt+D, then press 0-9
Load default window layout	Alt+Shift+D, then press D
Load stereo recording window layout	Alt+Shift+D, then press R
Load Red Book authoring window layout	Alt+Shift+D, then press B
Load 5.1-channel video window layout	Alt+Shift+D, then press V
Cascade the data windows	Shift+F5
Restore the active data window	Ctrl+F5
Toggle playback scrolling on and off	F6
Toggle smooth playback scrolling on and off	Shift+F6
Go to the next data window	Ctrl+F6
Go to the previous data window	Ctrl+Shift+F6
Maximize all data windows	Ctrl+F10
Maximize the Sound Forge application window	Alt+F10
Show/hide windows docked at the bottom of the workspace	F11
Show/hide windows docked at the sides of the workspace	Shift+F11
Show/hide all docked windows (excluding floating window docks)	Ctrl+F11
Maximize the width of the active data window	Ctrl+Enter

Data window shortcuts

Command	Keyboard Shortcut
Select previous/next editing tool	D/Shift+D
Select normal edit tool	Ctrl+D
Select all data in the active window	Ctrl+A
Copy the selected data onto the clipboard	Ctrl+C
Paste the clipboard contents into a new data window	Ctrl+E
Mix data from the clipboard with the active window	Ctrl+M
Trim (crop) to the current selection	Ctrl+T
Paste data from the clipboard into the active window	Ctrl+V
Enable locking markers, regions, and envelope points to selection	Ctrl+L
Move (cut) the selected data onto the clipboard	Ctrl+X
Repeat last process, effect, or tool	Ctrl+Y
Undo the last action	Ctrl+Z
Redo the last undone action	Ctrl+Shift+Z
Clear (delete) the selected data; nothing will be placed on the clipboard	Delete
Note: *If the* **Treat as Cutlist** *command (available in the* **Edit** *menu,* **Playlist/Cutlist** *submenu) is selected, deleting a selection creates a region in the Cutlist window, but does not remove the selection.*	
Place a command marker at the current cursor position	C
Place a marker at the current cursor position	M
Place a region at the current cursor position	R
Place a region at the current cursor position (when the Event tool ( ) is selected)	Ctrl+Alt+R
Enable snapping	F8
Toggle auto snap to grid	Ctrl+F8
Toggle auto snap to markers	Shift+F8
Toggle auto snap to event edges	Ctrl+Shift+F8
Toggle auto snap to zero crossings	Ctrl+B
Insert/show/hide volume envelope	V
Insert/remove volume envelope	Shift+V
Insert/show/hide pan envelope	P
Insert/remove pan envelope	Shift+P
Create a loop from the current selection without displaying the Sampler Loops window	Alt+L
Create a loop from the current selection	Alt+Shift+L
Stop or cancel the current action (including playback)	Esc
Split event at cursor position (when the Event tool ( ) is selected)	S
Split selected events at region boundaries (when the Event tool ( ) is selected)	Ctrl+Alt+T

Cursor movement shortcuts

Command	Keyboard Shortcut
Move one pixel right/left	Right Arrow/Left Arrow
Go to end of file —or— Go to the next region, loop or marker boundary (if regions, loops, or markers exist in the file)	Ctrl+Right Arrow
Go to beginning of file —or— Go to the previous region, loop or marker boundary (if regions, loops, or markers exist in the file)	Ctrl+Left Arrow
Move one video frame right/left (available only if the data window contains a video file)	Alt+Right Arrow/Left Arrow
Move one sample right/left	Ctrl+Alt+Right Arrow/Left Arrow
Show the Go To dialog	Ctrl+G
Go to the first sample visible in the waveform display (or beginning of selection)	Home
Go to the first sample in the data window	Ctrl+Home
Go to the last sample visible in the waveform display (or end of selection)	End
Go to the last sample in the data window	Ctrl+End
Move 10% of the current view prior to the cursor position	Page Up
Move 100% of the current view prior to the cursor position	Ctrl+Page Up
Move 10% of the current view past the cursor position	Page Down
Move 100% of the current view past the cursor position	Ctrl+Page Down
Center the cursor in the waveform display	\ or .
Go to the next sample	+ (numeric keypad)
Move 10 samples past the current cursor	Ctrl+numeric keypad +
Go to the previous sample	- (numeric keypad)
Move 10 samples prior to the current cursor	Ctrl+numeric keypad -

Data selection shortcuts

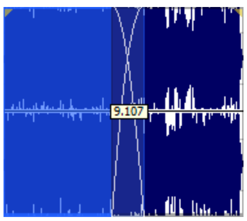
Command	Keyboard Shortcut
Show the Set Selection dialog	Ctrl+Shift+D
Select from the cursor to the next/previous screen pixel —or— Select next/previous event (when the Event tool () is selected)	Shift+Right/Left Arrow
Select from the cursor to the next/previous sample	Shift+Ctrl+Alt+Right/Left Arrow
Select from the cursor to the next/previous video frame (available only if the data window contains a video file)	Shift+Alt+Right/Left Arrow
Select from the cursor to the first sample visible in the waveform display —or— Select the first event (when the Event tool () is selected)	Shift+Home
Select from the cursor to the last sample visible in the waveform display —or— Select the last event (when the Event tool () is selected)	Shift+End
Select from the cursor to the first sample in the data window —or— Extend selection to the first event (when the Event tool () is selected)	Ctrl+Shift+Home
Select from the cursor to the last sample in the data window —or— Extend selection to the last event (when the Event tool () is selected)	Ctrl+Shift+End
Select from the cursor to 10% of the current view prior to the cursor position	Shift+Page Up
Select from the cursor to 10% of the current view past the cursor position	Shift+Page Down
Select 100% of the current view prior to the cursor position	Shift+Ctrl+Page Up
Select 100% of the current view past the cursor position	Shift+Ctrl+Page Down
Select from the cursor to the end of the file —or— Select from the cursor to the next region, loop or marker boundary (if regions, loops, or markers exist in the file) —or— Extend selection to the next event (when the Event tool () is selected)	Ctrl+Shift+Right Arrow
Select from the cursor to the beginning of the file —or— Select from the cursor to the previous region, loop or marker boundary (if regions, loops, or markers exist in the file) —or— Extend selection to the previous event (when the Event tool () is selected)	Ctrl+Shift+Left Arrow
Select from the cursor to the next sample	Shift+numeric keypad +
Select from the cursor to the previous sample	Shift+numeric keypad -
Select 10 samples past the current cursor	Shift+Ctrl+numeric keypad +
Select 10 samples prior to the current cursor	Shift+Ctrl+numeric keypad -
Snap to grid	T
Snap edge to grid	Shift+T
Snap to next zero crossing	Z
Snap edge to next zero crossing	Shift+Z
Switch the selection through the channels in a multichannel file	Tab/Shift+Tab
Shift current selection to the left by the length of the selection	<
Shift current selection to the right by the length of the selection	>
Cut the current selection length in half	;
Double the current selection length	'

Command	Keyboard Shortcut
Rotate audio	:
Restore previous five time selections	Backspace
Toggle last selection/cursor position	S or /
Create a loop from the current selection	Alt+L
Create a loop from the current selection without displaying the Sampler Loops window	Alt+Shift+L

Navigation and playback shortcuts

Command	Keyboard Shortcut
Save a view in cell <Number> where <Number> ranges from 1 to 8	Ctrl+Shift+<Number>
Restore a view using cell <Number> where <Number> ranges from 1 to 8	Ctrl+<Number>
Move cursor to corresponding marker or select corresponding region	0-9 keys (not numeric keypad)
Increase time magnification (zoom in)	Up Arrow or mouse wheel up
Decrease time magnification (zoom out)	Down Arrow or mouse wheel down
Increase level magnification	Shift+Up Arrow
Decrease level magnification	Shift+Down Arrow
Zoom to selection if a selection exists; otherwise Zoom In Full —or— Zoom event (when the Event tool ( ) is selected)	Ctrl+Up Arrow
Zoom normal (zooms to default zoom ratio set in Preferences)	Ctrl+Down Arrow
Display custom zoom ratio 1	1 (numeric keypad)
Display custom zoom ratio 2	2 (numeric keypad)
Pan data window up/down if zoomed in vertically	Ctrl+Shift+Up/Down Arrow
Switch cursor to opposite end of selection	5 (numeric keypad)
Set Mark In at the current cursor position	I
Set Mark Out at the current cursor position	O
Arm for recording	Ctrl+Shift+A
Start/stop recording	Ctrl+R
Toggle loops playback	Q
Play or Stop the contents of the data window in default mode	Spacebar or F12
Play All	Shift+Spacebar or Shift+F12
Play/Pause	Enter or Ctrl+F12
Switch play mode through Normal, Plug-In Chain, Play as Sample, and Play as Cutlist playback modes	X
Stop playback	Esc
Seek cursor on playback	F
Preview cut (skip selection on playback with pre-roll)	Ctrl+K
Play to cursor with pre-roll	Ctrl+Shift+K
Scrub playback	J, K, or L
Toggle playback scrolling on and off	F6
Toggle smooth playback scrolling on and off	Shift+F6
Generate MIDI timecode	F7
Trigger from MIDI timecode	Ctrl+F7

Event tool keyboard shortcuts

Command	Keyboard Shortcut
Select the previous/next editing tool (Edit tool, Magnify tool, Pencil tool, Event tool)	D or Shift+D
Select the next event	Shift+Right Arrow
Select the previous event	Shift+Left Arrow
Select the first event	Shift+Home
Select the last event	Shift+End
Extend the selection to the next event	Shift+Ctrl+Right Arrow
Extend the selection to the previous event	Shift+Ctrl+Left Arrow
Extend the selection to the first event	Shift+Ctrl+Home
Extend the selection to the last event	Shift+Ctrl+End
Split events at cursor	S
Split events at region boundaries	Ctrl+Alt+T
Toggle automatic crossfades on/off	Ctrl+Shift+X
Show/hide fade lengths between events: 	Ctrl+Shift+T
Toggle auto ripple on/off	Ctrl+Shift+R

Plug-In Chain shortcuts

Command	Keyboard Shortcut
Preview	Ctrl+P
Process selection	Ctrl+Shift+P
Bypass selected plug-in	Ctrl+B
Bypass plug-in chain	Ctrl+Shift+B
Save chain preset	Ctrl+S
Toggle Ignore/Mix/Insert Tail Data options	Ctrl+T
Add plug-ins to chain	Ctrl+E
Remove selected plug-ins	Ctrl+Delete
Select the next plug-in in the chain	Down Arrow
Select the previous plug-in in the chain	Up Arrow
Show the plug-in window	Ctrl+D
Show or hide the effect automation parameters	Ctrl+H
Show or hide envelope in data window	Shift+D
Enable or disable envelope	Shift+E
Select all plug-ins or effect automation parameters	Ctrl+A

Regions List shortcuts

Command	Keyboard Shortcut
Play or stop the active marker or region	Spacebar
Edit the active marker or region	Enter
Delete the active marker or region	Delete

Command	Keyboard Shortcut
Create region from the current selection	R Ctrl+I
Replicate selected region	Ctrl+D
Update region to match selection	Ctrl+U

Playlist/Cutlist shortcuts

Command	Keyboard Shortcut
Play or stop the active playlist entry	Spacebar
Edit the active playlist entry	Enter
Delete the active playlist entry	Delete
Add one to the active playlist entry play count	+ (plus sign) (not numeric keypad)
Subtract one from the active playlist entry play count	- (minus sign) (not numeric keypad)
Add or remove a stop point on the active playlist entry	Ctrl+E Ctrl+8
Toggle pre-roll on and off for the playlist	Ctrl+P / (forward slash) (not numeric keypad)
Add selected Regions List item to Playlist	Ctrl+I
Replicate selected playlist region	Ctrl+D

Script Editor shortcuts

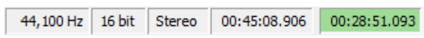
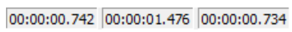
Command	Keyboard Shortcut
Create a new script	Ctrl+N
Open a script	Ctrl+O
Run script	Ctrl+R
Compile script	Ctrl+Shift+R
Save script	Ctrl+S
Find next instance of last-searched text	F3
Find previous instance of last-searched text	Shift+F3
Find next instance of the selected text	Ctrl+F3
Find previous instance of the selected text	Ctrl+Shift+F3

Drag-and-drop shortcuts

Drag-and-drop allows you to quickly perform operations crossing between open data windows, the Playlist/Cutlist window, the Regions List, and the time ruler.

Command	Function
Drag to New	To create a new file from the current selection, drag the selection to an open area of the Sound Forge desktop. **Note:** *You can also drag regions from the Regions List to the desktop.*
Drag Mix	To mix a selection, drag the selection from the source to the place where you want to mix the selected data. You can drag the selection to the same data window or another data window.
Drag Paste	To paste a selection, hold Ctrl and drag the selection from the source to the place where you want to paste the selected data. You can drag the selection to the same data window or another data window.
Drag to Regions List	To add the current selection to the Regions List, drag it to the Regions List. You can also drag regions from the Regions List to the Playlist.
Drag to Playlist	To add a region from the Regions List to a playlist, drag it from the waveform display or the Regions List to the Playlist/Cutlist window. You can also drag regions within a playlist to rearrange the playback order.
Drag to Time Ruler	To create a region, drag the current selection to the time ruler.
Drag to Track List	To create a disc-at-once CD track, drag a selection to the Track List window.
Drag CD Track	From the **Options** menu, choose **Drag-and-Drop Editing**, and then choose **CD Track** from the submenu if you want to create disc-at-once tracks during drag-and-drop editing. Choosing this command has the same effect as toggle-clicking the right mouse button while dragging until the cursor is displayed as a mouse pointer with a CD ().

Mouse shortcuts

Command	Function
Select All	Double-click the waveform display to select the entire sound file. Triple-click when regions, loops or markers are present (if the **Disable triple-clicking to select all sound file data** check box on the Editing tab in the Preferences dialog is cleared).
Zoom Time and Level	Double-click the level ruler to zoom the current selection vertically and horizontally. If no selection exists, the entire waveform display data is zoomed. Double-click again to zoom out to the full amplitude and to the normal horizontal magnification.
Return Control Value to Default	Double-click a slider, fader, or spinner to reset it to its default value.
Fine Tune Control Value	To fine tune a trackbar, fader, or spinner, hold the right and left mouse buttons (or hold the Ctrl key) when dragging.
Preview	Shift-click the **Preview** button to hear the original sound. This is equivalent to enabling the **Bypass** check box.
Main Status Bar	Double-click the **Sample Rate**, **Sample Size**, or **Channels** box at the bottom of the Sound Forge window to edit their values. 
Selection Status Bar	Double-click the **Selection Start** box in a data window to type a new value. Double-click the **Selection End** or **Selection Length** box to type a new value. 
Go to Marker	Click a marker tag ( ) to move the cursor to the marker's position.
Set Selection to Region/Loop	Double-click a region ( )or loop tag ( ) in the ruler to change the current selection to the region or loop end points.
Play Normal Button (on playbar)	Ctrl+click the **Play Normal** button ( ) to Preview Cut (skip selection) on playback. Ctrl+Shift+click to play to cursor with pre-roll.
Slow and Fast Selection Scroll Toggle	When making a selection past the end or beginning of the waveform display, click the right mouse button (while holding down the left mouse button) to toggle between fast scrolling and slow scrolling.
Zoom in horizontally	Rotate the mouse wheel forward.
Zoom out horizontally	Rotate the mouse wheel back.
Zoom in/out vertically	Ctrl + rotate the mouse wheel forward or back.
Scroll left/right 10% of the current view prior	Shift + rotate the mouse wheel forward or back.
Move cursor left or right (move current selection point if selection exists)	Ctrl + Shift + rotate the mouse wheel forward or back.

Appendix B

Sound Forge and the Microsoft Audio Compression Manager

The Microsoft Audio Compression Manager (ACM) is a standard interface for audio compression in Windows. This interface allows applications such as Sound Forge software to use compression algorithms provided by other companies.

Sound Forge® software fully supports audio compression through the ACM. This enables you to use any ACM-compatible compression. The best part of this support is you don't have to learn anything new to use it! Sound Forge software transparently opens compressed .wav files and provides all available compression formats for .wav files in the Save As dialog.

Audio data compression and decompression

The first piece of the ACM allows you to compress and decompress audio data. Audio compression is used to decrease the amount of data required to represent a sound; this ultimately results in smaller sound files. However, there are drawbacks to using audio compression on your sound files:

- Most audio compression algorithms will degrade the quality of the sound. This is referred to as lossy compression since information contained in the sound is lost when it is compressed. The amount of sound degradation is dependent on the compression algorithm.
- Audio compression requires more processing time than uncompressed data. The amount of processing time is dependent on the algorithm, as well as your hardware setup. As a result, opening and saving compressed files will usually take longer than uncompressed files.
- Compressed files are not as portable as uncompressed files. If you are distributing files in a compressed format, you must ensure the person receiving the files can use them. Also, not all audio software can use compressed .wav files, which could make using other programs with Sound Forge software inconvenient.

In Sound Forge software, any compressed .wav file can be opened as long as a compatible ACM driver is installed and enabled. If there is no compatible ACM driver available for a compressed .wav file, Sound Forge software will inform you of the problem when you try to open it.

Saving compressed .wav files is as simple as choosing the compression algorithm in the Format drop-down list of the Save As dialog. Once a file has been saved as compressed, Sound Forge software will always save changes to the file using the selected compression algorithm; you do not need to reselect the compression format each time you save. However, you can change the compression format or revert to an uncompressed format at any time with the Save As dialog.

Transparent playback and recording of non-hardware-supported audio files

The Microsoft Sound Mapper allows audio data formats that are not directly supported by your sound card to be played and recorded. Sound Forge software lets you use the Sound Mapper by selecting it for playback and recording on the Audio tab of the Preferences dialog.

Using the Sound Mapper with uncompressed files

The primary use of the Sound Mapper for uncompressed sound files is for your convenience. You don't have to convert the sound to a supported format before you listen to it:

- If, for example, you have a sound file that is recorded at a nonstandard sample rate such as 22,257 Hz, and the closest sample rate that your sound card supports is 22,050 Hz, then the sound file normally cannot be played. You would have to change the sample rate of the file to 22,050 Hz before you could play it. However, changing the sample rate (without resampling) would cause the sound to play at a lower pitch. Using the Sound Mapper, you can play this sound file correctly without resampling the file first. The Sound Mapper will map the sound to the best format possible and perform the resampling in real time.
- The Sound Mapper will allow you to play 16- or 24-bit sounds on an 8-bit sound card, play stereo sounds on a mono sound card, and record stereo files on a mono-only sound card. However, when you use the Sound Mapper to record in stereo from a mono source, the mono input of your sound card is simply duplicated in both channels--the Sound Mapper cannot create something that is better than what the sound card can supply.

Using the Sound Mapper with compressed files

The Sound Mapper allows you to play (and sometimes record) compressed sound files—even on sound cards that do not directly support compression—so you can play a sound file that is compressed with Microsoft ADPCM or the DSP Group's TrueSpeech without decompressing the file first.

The Sound Mapper cannot always record compressed sound files because compressing sound data can be very processor intensive: the amount of time required is dependent on the compression algorithm and how it is implemented. Decompressing sound data is almost always faster than compressing the same sound data.

Sound Forge software does not play and record compressed sound files directly. Rather, all compression and decompression is performed while opening and saving the sound files. Sound Forge software saves compressed sound files using the best quality possible—something that cannot always be done in real time. Saving compressed sound files with Sound Forge software will usually sound better than those recorded with audio compression.

Notes:

- *The Open dialog allows you to preview compressed .wav files if you have an appropriate ACM driver is installed. However, you must have your Default playback device set to the Sound Mapper for this to work.*
- *When saving uncompressed audio data to a compressed format with the Save As dialog, it is a good idea to close the file and reopen it after saving. Since Sound Forge software performs the compression and decompression during saving and loading you will not be able to hear what the file sounds like with compression until after you've saved and reloaded the file.*

Appendix C

SMPTE Timecode

The Society of Motion Picture and Television Engineers (SMPTE) timecode may be one of the most misunderstood concepts among individuals within the music industry. The problem with SMPTE timecode formats is that they may mean different things to people in the audio and video fields. What follows is a brief description of each SMPTE timecode format.

Important: *When synchronizing audio to video, it is crucial that the SMPTE timecode format used in the sequencer or digital audio workstation is the same as the SMPTE timecode striped onto the video. This guarantees that the SMPTE times on the video screen and computer monitor synchronize during playback.*

SMPTE 25 EBU (25 fps, Video)

SMPTE 25 EBU timecode runs at 25 fps (frames per second), and matches the frame rate used by European Broadcasting Union (EBU) television systems.

SMPTE 25 EBU format is used for PAL DV/D1 video projects.

SMPTE Drop Frame (29.97 fps, Video)

SMPTE Drop Frame timecode runs at 29.97 fps, and matches the frame rate used by NTSC television systems (North America, Japan).

SMPTE Drop Frame format is used for NTSC DV/D1 video projects.

Both SMPTE Drop and SMPTE Non-Drop run at 29.97 fps. In both formats, the actual frames are not discarded, but they are numbered differently. SMPTE Drop removes certain frame numbers from the counting system to keep the SMPTE clock from drifting from real (“wall clock”) time. The time is adjusted forward by two frames on every minute boundary except 0, 10, 20, 30, 40, and 50. For example, when SMPTE Drop time increments from 00:00:59.29, the next value is 00:01:00.02.

SMPTE Non-Drop Frame (29.97 fps, Video)

SMPTE Non-Drop Frame timecode runs at a rate of 29.97 fps. This leads to a discrepancy between real (“wall clock”) time and the SMPTE time, because there is no compensation in the counting system as there is in SMPTE Drop Frame.

SMPTE Non-Drop format is used for NTSC D1 video projects that are recorded on master tapes striped with Non-Drop timecode.

SMPTE 30 (30 fps, Audio)

SMPTE 30 is an audio-only format and runs at exactly 30 fps. SMPTE 30 is commonly used when synchronizing audio applications such as multitrack recorders or MIDI sequencers. This format is not used when working with video.

SMPTE Film Sync (24 fps)

The SMPTE Film Sync time format runs at 24 fps (frames per second). This frame rate matches the standard crystal-sync 16/33 mm film rate of 24 fps.

Appendix D

Using CSOUND, MTU, IRCAM, BICSF, and EBICSF Files

Although Sound Forge® Pro software supports a large number of sound file formats directly, it does not support the CSOUND, MTU, IRCAM, BICSF or EBICSF file types. However, you can use the Sound Forge Raw File Type capabilities to extract sound data from these file types.

About IRCAM files

The IRCAM or IRCAM-Gross format consists of a 1024-byte header prior to the audio data. This header contains standard information like the number of channels, sampling rate, and data format, but can also contain the name of the sample and comments. This format is used by the MTU system and these files are frequently referred to as MTU files. IRCAM files support two types of data formats: 16-bit linear PCM and floating point data.

About BICSF and EBICSF files

BICSF and EBICSF files (Berkeley/IRCAM/CARL Sound File or Extended BICSF) are extensions of the IRCAM format. Instead of using the standard IRCAM header, these files replace the first 28 bytes of the header with a standard NeXT/Sun header. This allows the IRCAM format to store additional information in its 1024-byte header, while also allowing the files to be read by software that supports the NeXT/Sun file format, such as Sound Forge software.

Opening files

BICSF and EBICSF files

When reading BICSF and EBICSF files, the software identifies them as NeXT/Sun files. This is because the header of the BICSF file has been modified to allow it to be read as a NeXT/Sun file. These files are read as long as they are in one of the supported NeXT/Sun data formats.

IRCAM, CSOUND and MTU files

To read these formats, users must import them as Raw data files. This is best accomplished by configuring the parameters in the Raw File Type dialog and saving them as presets. The Raw File import function allows these files to be opened providing they are stored in 16-bit linear format. Sound Forge software does not open floating point format IRCAM files.

Opening an IRCAM file

1. From the **File** menu, choose **Open**. The Open dialog appears.
2. Specify **Raw Audio** from the **Files of type** drop-down list.
3. Select an IRCAM file to open and click **Open**. The Raw File Type dialog appears.
4. Configure the following parameters:
   - Specify a sample rate from the **Sample rate** drop-down list.
   - In the **Sample type** area, select the **16-bit PCM** radio button.
   - In the **Format** area, select the **Signed** radio button.
   - Select the appropriate **Byte order** radio button.
   - Choose a setting from the **Channels** drop-down list to select the number of channels stored in the file.
   - Set the **Header** value to **1024 bytes.**
   - Set the **Trailer** value to **0 bytes.**
   - To automatically use these settings to open all Raw files, select the **Remember my preference and apply it in the future** check box.

Tip: *If you select the* **Remember my preference and apply it in the future** *check box, Sound Forge will bypass the Raw File Type dialog. However, you can access and change these settings from the Open dialog by selecting a Raw file and then clicking the* **Custom** *button.*

5. Click **OK**.

Notes:

- *The settings you choose for opening the file — with the exception of the sample rate — will be used when you click* **Save** *or save the file using the* **Default Template** *setting in the Save As dialog as long as the number of channels in the source file matches the number of channels in the file you're saving. The sample rate will be determined from the source file.*
- *If you do not always use the same settings for reading raw files, make sure the* **Keep media files locked** *check box is selected on the* **General** *tab of the Preferences dialog. Otherwise, the individual settings will be lost if you have multiple raw files open and switch away from the Sound Forge window.*

6. Click **Save As**. The Save Preset dialog is displayed.
7. Enter a name for the preset in the **New preset name** box and click **OK**. The preset is saved and the Raw File Type dialog is displayed.
8. Click **OK**. The file opens.

Remember that the byte order of files generated by CSOUND is not constant. CSOUND executables for PC generate files that use Little Endian byte ordering, while CSOUND for other platforms tends to generate files with Big Endian ordering. In addition, MTU files use Big Endian byte ordering. Sony Creative Software Inc. recommends initially trying a file in Big Endian.

Note: *You may want to save presets for byte ordering, as well as mono or multichannel, when receiving CSOUND files from a number of source computers.*

Saving files

You cannot save these files in their original format. You must select a Sound Forge-supported file format.

To save files for use with software that supports the BICSF/EBICSF format, use the NeXT/Sun format. This format does not save the additional information found in BICSF/EBICSF files, but allows the data to be read as a NeXT/Sun file.

Appendix E

Glossary

A-Law

A-Law is a compounded compression algorithm for voice signals defined by the Geneva Recommendations (G.711). The G.711 recommendation defines A-Law as a method of encoding 16-bit PCM signals into a nonlinear 8-bit format. The algorithm is commonly used in United States telecommunications. A-Law is very similar to µ-Law; however, each uses a slightly different coder and decoder.

Acoustic Signature

The acoustic signature of a system is data containing all of the sound characteristics of a system. This includes such things as reverb time, frequency response and other timbral qualities. Impulse files used by Acoustic Mirror can be thought of as acoustic signatures.

Activation Number

This number is based on the Computer ID number of the computer on which Sound Forge software is installed. Each computer has a unique number, similar to a license plate. An activation number is created based on that number. When you register the software, Sony will generate an activation number for you. Once the activation number is entered, the software will not time out. Since the activation number is based on the Computer ID, it is important that you have Sound Forge software installed on the computer where you will be using it.

See also Computer ID on page 368.

ActiveX

A Microsoft technology that enables different programs to share information. ActiveX extends Microsoft Windows-based architecture to include Internet and corporate intranet features and capabilities. Developers use it to build user interactivity into programs and World Wide Web pages.

Adaptive Delta Pulse Code Modulation (ADPCM)

A method of compressing audio data. Although the theory for compression using ADPCM is standard, there are many different algorithms employed. For example, Microsoft's ADPCM algorithm is not compatible with the International Multimedia Association's (IMA) approved ADPCM.

Advanced Streaming Format (ASF)

See Windows Media Format on page 381.

Aliasing

A type of distortion that occurs when digitally recording high frequencies with a low sample rate. For example, in a motion picture, when a car's wheels appear to slowly spin backward while the car is quickly moving forward, you are seeing the effects of aliasing. Similarly, when you try to record a frequency greater than one half of the sampling rate (the Nyquist Frequency), instead of hearing a high pitch, you may hear a low-frequency rumble. *See also Nyquist Frequency on page 374*.

To prevent aliasing, an anti-aliasing filter is used to remove high frequencies before recording. Once the sound has been recorded, aliasing distortion is impossible to remove without also removing other frequencies from the sound. This same anti-aliasing filter must be applied when resampling to a lower sample rate.

Amplitude Modulation

Amplitude Modulation (AM) is a process whereby the amplitude (loudness) of a sound is varied over time. When varied slowly, a tremolo effect occurs. If the frequency of modulation is high, many side frequencies are created that can strongly alter the timbre of a sound.

Analog

When discussing audio, this term refers to a method of reproducing a sound wave with voltage fluctuations that are analogous to the pressure fluctuations of the sound wave. This is different from digital recording in that these fluctuations are infinitely varying rather than discrete changes at sample time. *See also Quantization on page 376*.

Attack

The attack of a sound is the initial portion of the sound. Percussive sounds (drums, piano, guitar plucks) are said to have a fast attack. This means that the sound reaches its maximum amplitude in a very short time. Sounds that slowly swell up in volume (soft strings and wind sounds) are said to have a slow attack.

Audio Compression Manager (ACM)

The Audio Compression Manager, from Microsoft, is a standard interface for audio compression and signal processing for Windows. The ACM can be used by Windows programs to compress and decompress .wav files. *See also Wave on page 381*.

Audio Event Locator

The Audio Event Locator is similar to a scrub function. However, rather than playing the sound file at a slow speed, it loops playback around the cursor position. This position can be selected by dragging the cursor around in the Sound Forge Overview window.

Audio Interchange File Format (AIFF)

An audio file format developed by Apple Computer.

ASX File

ASF Stream Redirector file. *For more information, see Redirector File on page 376*.

Attenuation

A decrease in the level of a signal.

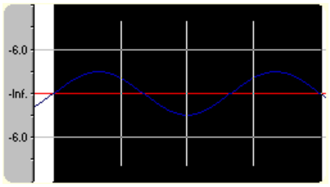
Bandwidth

When discussing audio equalization, each frequency band has a width associated with it that determines the range of frequencies that are affected by the EQ. An EQ band with a wide bandwidth will affect a wider range of frequencies than one with a narrow bandwidth.

When discussing network connections, bandwidth refers to the rate of signals transmitted or the amount of data that can be transmitted in a fixed amount of time (stated in bits/second): a 56 Kbps network connection is capable of receiving 56,000 bits of data per second.

Baseline

The baseline of a waveform is also referred to as the zero-amplitude axis or negative infinity. In the following image, the red line represents the baseline.



Beats Per Minute (BPM)

The tempo of a piece of music can be written as a number of beats in one minute. If the tempo is 60 BPM, a single beat will occur once every second.

Bit

The most elementary unit in digital systems. Its value can only be 1 or 0, corresponding to a voltage in an electronic circuit. Bits are used to represent values in the binary numbering system. As an example, the 8-bit binary number 10011010 represents the unsigned value of 154 in the decimal system. In digital sampling, a binary number is used to store individual sound levels, called samples.

Bit Depth

The number of bits used to represent a single sample. For example, 8- or 16-bit are common sample sizes. While 8-bit samples take up less memory (and hard disk space), they are inherently noisier than 16- or 24-bit samples.

Buffer

Memory used as an intermediate repository in which data is temporarily held while waiting to be transferred between two locations. A buffer ensures that there is an uninterrupted flow of data between computers. Media players may need to rebuffer when there is network congestion.

Bus

A virtual pathway where signals from tracks and effects are mixed. A bus's output is a physical audio device in the computer from which the signal will be heard.

Byte

Refers to a set of 8 bits. An 8-bit sample requires one byte of memory to store, while a 16-bit sample takes two bytes of memory to store

Channel Converter

The Channel Converter is a function that converts files from mono to stereo and stereo to mono with independent level control of the new channels. This function can also create interesting effects by converting stereo files to stereo with various levels and inversion of channels.

Channel Meters

The Channel Meters in Sound Forge software display the peak output levels of the sound file currently playing. These meters have selectable resolution and options to hold peaks and valleys.

Chorus

Chorusing is an effect created by combining a signal with a modulating, delayed copy of itself. This effect creates the illusion of multiple sources creating the same sound.

Clipboard

The clipboard is where sample data is saved when you cut or copy it from a data window. You can then paste, mix, or crossfade the sample data stored on the clipboard with another data window. This sample data can also be used by other Windows applications that support Sound data on the clipboard, such as Sound Recorder.

Clipping

Occurs when the amplitude of a sound is above the maximum allowed recording level. In digital systems, clipping is seen as a clamping of the data to a maximum value, such as 32,767 in 16-bit data. Clipping causes sound to distort.

Codec

Coder/Decoder: refers to any technology for compressing and decompressing data. The term codec can refer to software, hardware, or a combination of both technologies.

Compression Ratio (audio)

A compression ratio controls the ratio of input to output levels above a specific threshold. This ratio determines how much a signal has to rise above the threshold for every 1 dB of increase in the output. For example, with a ratio of 3:1, the input level must increase by three decibels to produce a one-decibel output-level increase:

Threshold	-10 dB
Compression Ratio	3:1
Input	-7 dB
Output	-9 dB

Because the input is 3 dB louder than the threshold and the compression ratio is 3:1, the resulting signal is 1 dB louder than the threshold.

Compression Ratio (file size)

The ratio of the size of the original uncompressed file to the compressed contents. For example, a 3:1 compression ratio means that the compressed file is one-third the size of the original.

Computer ID

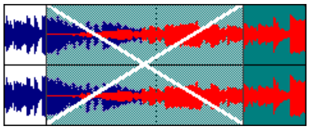
Each computer has a unique number, similar to a license plate. An activation number is created based on that number. Since the activation number is based on the Computer ID, it is important that you have Sound Forge software installed on the computer where you will be using it. The Computer ID is automatically detected and provided to you when you install the software.

> **Note:** *The Computer ID is used for registration purposes only. It doesn't give Sony access to any personal information and can't be used for any purpose other than for generating a unique activation number for you to use the software.*

See also Activation Number on page 365.

Crossfade

Mixing two pieces of audio by fading one out as the other fades in:



Crossfade Loop

Sometimes a sample loop cannot be easily created from the given source material. In these instances, a crossfade can be applied to the beginning and end of the loop to aid in the smooth transition between the two. The Crossfade Loop function provides a method of creating sampling loops in material that is otherwise difficult to loop.

Cutoff frequency

The cutoff frequency of a filter is the frequency at which the filter changes its response. For example, in a low-pass filter, frequencies greater than the cutoff frequency are attenuated, while frequencies less than the cutoff frequency are not affected.

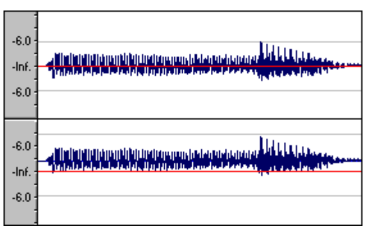
Data Window

Each opened sound file has its own data window. At the top of each data window is a title bar displaying either the title of the sample or the name of the file. Also in each data window are the waveform display, time and level rulers, playbar and other tools that give you information and allow you to navigate throughout the entire sound file.

DC Offset

DC offset occurs when hardware, such as a sound card, adds DC current to a recorded audio signal. This current results in a recorded waveform that is not centered around the baseline (-infinity). Glitches and other unexpected results can occur when sound effects are applied to files that contain DC offsets. Sound Forge software can compensate for this DC offset by adding a constant value to the samples in the sound file.

In the following example, the red line represents the baseline. The lower waveform exhibits DC offset; note that the waveform is centered approximately 2 dB above the baseline.



Decibel (dB)

A unit used to represent a ratio between two numbers using a logarithmic scale. For example, when comparing the numbers 14 and 7, you could say 14 is two times greater than the number 7; or you could say 14 is 6 dB greater than the number 7. Where did we pull that 6 dB from? Engineers use the equation dB = 20 x log (V1/V2) when comparing two instantaneous values. Decibels are commonly used when dealing with sound because the ear perceives loudness in a logarithmic scale.

In Sound Forge software, most measurements are given in decibels. For example, if you want to double the amplitude of a sound, you apply a 6 dB gain. A sample value of 32,767 (maximum positive sample value for 16-bit sound) can be referred to as having a value of 0 dB. Likewise, a sample value of 16,384 can be referred to having a value of -6 dB.

Device Driver

A program that enables Windows to connect different hardware and software. For example, a sound card device driver is used by Windows software to control sound card recording and playback.

Destructive Editing

Destructive editing is the type of editing whereby all cuts, deletes, mixes and other processes are actually processed to the sound file. Any time you delete a section of a sound file in Sound Forge software, the sound file on disk is actually rewritten without the deleted section.

See also Nondestructive Editing on page 374.

Digital Rights Management (DRM)

A system for delivering songs, videos, and other media over the Internet in a file format that protects copyrighted material. Current proposals include some form of certificates that validate copyright ownership and restrict unauthorized redistribution.

Digital Signal Processing (DSP)

A general term describing anything that alters digital data. Signal processors have existed for a very long time (tone controls, distortion boxes, wah-wah pedals) in the analog (electrical) domain. Digital Signal Processors alter the data after it has been digitized by using a combination of programming and mathematical techniques. DSP techniques are used to perform many effects such as equalization and reverb simulation.

Since most DSP is performed with simple arithmetic operations (additions and multiplications), both your computer's processor and specialized DSP chips can be used to perform any DSP operation. The difference is that DSP chips are optimized specifically for mathematical functions while your computer's microprocessor is not. This results in a difference in processing speed.

DirectX

A set of Application Program Interfaces designed by Microsoft for multimedia development. A DirectX plug-in, such as the Sony Noise Reduction DirectX Plug-In, uses the DirectX Media Streaming Services (DMSS) API. Because DMSS is a standard API, a DirectX plug-in can be used in any application that supports DMSS.

Dithering

Dithering is the practice of adding noise to a signal to mask quantization noise.

See also Quantization Noise on page 376.

Drag and Drop

A quick way to perform certain operations using the mouse. To drag and drop, you click and hold a highlighted selection, drag it (hold the left mouse button down and move the mouse) and drop it (let go of the mouse button) at another position on the screen.

Dynamic Range

The difference between the maximum and minimum signal levels. It can refer to a musical performance (high-volume vs. low-volume signals) or to electrical equipment (peak level before distortion vs. noise floor).

Endian (Little and Big)

Little and Big Endian describe the ordering of multi-byte data that is used by a computers microprocessor. Little Endian specifies that data is stored in a low-to-high byte format; this ordering is used by the Intel microprocessors. Big Endian specifies that data is stored in a high-to-low byte format; this ordering is used by the Motorola microprocessors.

Equalization (EQ)

Equalizing a sound file is a process by which certain frequency bands are raised or lowered in level.

Fast Fourier Transform (FFT) Analysis

A Fourier Transform is the mathematical method used to convert a waveform from the Time Domain to the Frequency Domain.

Since the Fourier Transform is computationally intensive, it is common to use a technique called a Fast Fourier Transform (FFT) to perform spectral analysis. The FFT uses mathematical shortcuts to lower the processing time at the expense of putting limitations on the analysis size.

The analysis size, also referred to as the FFT size, indicates the number of samples from the sound signal used in the analysis and also determines the number of discrete frequency bands. When a high number of frequency bands are used, the bands have a smaller bandwidth, which allows for more accurate frequency readings.

File Associations

This dialog allows you to associate sound file extensions (such as .wav, .au, .snd, etc.) with Sound Forge software. This dialog is opened from the File tab of the Preferences dialog.

File Format

A file format specifies the way in which data is stored. In Windows, the most common audio file format is the Microsoft .wav format. For information on the different file formats supported by Sound Forge software, click here.

Frame Rate

Audio uses frame rates only for the purposes of synchronizing to video or other audio. To synchronize with audio, a rate of 30 non-drop is typically used. To synchronize with video, 30 drop is usually used.

Frequency Modulation (FM)

Frequency Modulation (FM) is a process by which the frequency (pitch) of a sound is varied over time. Subaudio frequency modulation results in pitch-bending effects (vibrato). Frequency modulation within audio band frequencies (20 Hz - 20,000 Hz) creates many different side-band frequencies that drastically alter the timbre of the sound.

Frequency Modulation (FM) Synthesis

This type of synthesis relies on the principles of Frequency Modulation. The FM Synthesis tool allows you to use frequency modulation (FM) and additive synthesis to create complex sounds from simple waveforms.

In frequency modulation, the frequency of a waveform (the carrier) is modulated by the output of another waveform (the modulator) to create a new waveform. If the frequency of the modulator is low, the carrier will be slowly detuned over time. However, if the frequency of the modulator is high, the carrier will be modulated so quickly that many additional frequencies, or sidebands, are created.

In Sound Forge software, up to four waveforms (operators) can be used in a variety of configurations. Depending on the configuration, an operator can be a carrier, a modulator, or a simple, unmodulated waveform.

Frequency Spectrum

The frequency spectrum of a signal refers to its range of frequencies. In audio, the audible frequency range is between 20 Hz and 20,000 Hz. The frequency spectrum sometimes refers to the distribution of these frequencies. For example, bass-heavy sounds have a large frequency content in the low end (20 Hz - 200 Hz) of the spectrum.

Head-Related Transfer Function (HRTF)

Sounds are perceived differently depending on the direction the sound comes from. This occurs because of the echoes bouncing from your shoulders and nose and the shape of your ears. A head-related transfer function contains the frequency and phase response information required to make a sound appear to originate from a certain direction in 3-dimensional space.

Hertz (Hz)

The unit of measurement for frequency or cycles per second (CPS).

High-Pass Filter

A high-pass filter attenuates all frequencies below a cutoff frequency. It is usually used to remove low-frequency rumble from audio files.

Insertion Point

The insertion point (also referred to as the cursor position) is analogous to the cursor in a word processor. It is where markers or commands may be inserted depending on the operation. The insertion point appears as a vertical flashing black line and can be moved by clicking the left mouse button anywhere in the data window.

Inverse Telecine (IVTC)

Telecine is the process of converting 24 fps (cinema) source to 30 fps video (television) by adding pulldown fields. Inverse telecine, then, is the process of converting 30 fps (television) video to 24 fps (cinema) by removing pulldown.

See also Pulldown on page 375; Telecine on page 380.

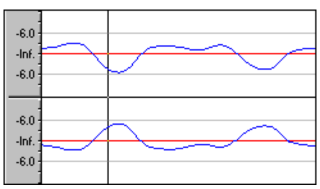
InterVoice Sound File Support

The InterVoice sound file format (.IVC), commonly used in telephony applications, is now supported and includes G.711 µ-Law and A-Law, G.721 ADPCM (32 kb/s) and G.723 ADPCM (24 kb/s) data formats.

Invert Data

Inverting sound data reverses the polarity of a waveform around its baseline. Inverting a waveform does not change the sound of a file; however, when you mix different sound files, phase cancellation can occur, producing a “hollow” sound. Inverting one of the files can prevent phase cancellation.

In the following example, the red line represents the baseline, and the lower waveform is the inverted image of the upper waveform.



Limiting

Limiting is essentially a hard compressor. Limiting is often used to keep signals from going above a certain level, but can also be applied to create heavily compressed effects. Limiting should only be performed on peaks; if the Threshold level is set too low, heavy distortion will occur.

See also Compression Ratio (audio) on page 368.

Loop

Loops are small audio clips that are designed to create a repeating beat or pattern. Loops are usually one to four measures long.

Low-Pass Filter

A low-pass filter attenuates all frequencies above a cutoff frequency. Low-pass filters can be used as anti-alias filters or for general tonal shaping.

Marker

A marker is an anchored, accessible reference point in a file. Markers are stored in the Regions List and can be used for quick navigation.

Media Control Interface (MCI)

A standard way for Windows programs to communicate with multimedia devices such as sound cards and CD players. If a device has an MCI device driver, it can easily be controlled by most multimedia Windows software.

Microsoft Sound Mapper

The Sound Mapper is a special device that attempts to select the most appropriate sound card (map) on which to play a sound, or it will translate the sound into a format that can be played on your sound card.

Mid-Side recording

Mid-side (MS) recording is a microphone technique in which one mic is pointed directly towards the source to record the center (mid) channel, and the other mic is pointed 90 degrees away from the source to record the stereo image. For proper playback on most systems, MS recordings must be converted to your standard left/right (also called AB) track.

MIDI

See Musical Instrument Device Interface (MIDI) on page 374.

MIDI Channels

MIDI allows for 16 discrete channels for sending data. When dealing with MIDI triggers, Sound Forge software needs to know what MIDI channel to look at for receiving the trigger. The channel this information is sent to in Sound Forge software depends on the device sending the MIDI messages.

MIDI Clock

A MIDI device-specific timing reference. It is not absolute time like MIDI Time Code (MTC); instead it is a tempo-dependent number of “ticks” per quarter note. MIDI clock is convenient for synchronizing devices that need to perform tempo changes mid-song.

MIDI Controllers

MIDI controllers are a specific type of MIDI message. Sound Forge software can use MIDI controllers to trigger events and playback of sound files. Consult your MIDI sending device to see what controller messages it sends.

MIDI Notes

MIDI notes are a specific type of MIDI message. Sound Forge software can use MIDI notes to trigger events and playback of sound files. Any MIDI sequencer or controller will send MIDI notes.

MIDI Port

A MIDI port is the physical MIDI connection on a piece of MIDI hardware. This port can be a MIDI in, out or through. Your computer must have a MIDI-capable card to output MIDI time code to an external device or to receive MIDI time code from an external device.

MIDI Time Code (MTC)

MTC is an addendum to the MIDI 1.0 specification and provides a way to specify absolute time for synchronizing MIDI-capable applications. MTC is essentially a MIDI representation of SMPTE time code.

Mix

Mixing allows multiple sound files to be blended into one file at user-defined relative levels.

Multiple-Bit-Rate Encoding

Multiple-bit-rate encoding allows you to create a single file that contains streams for several bit rates. A multiple-bit-rate file can accommodate users with different Internet connection speeds, or these files can automatically change to a different bit rate to compensate for network congestion without interrupting playback.

> **Note:** *To take advantage of multiple-bit-rate encoding, you must publish your media files to a Windows Media server or a RealServerG2.*

Musical Instrument Device Interface (MIDI)

A standard language of control messages that provides for communication between any MIDI-compliant devices. Anything from synthesizers to lights to factory equipment can be controlled via MIDI. Sound Forge software uses MIDI for synchronization purposes.

Noise-shaping

Noise-shaping is a technique which can minimize the audibility of quantization noise by shifting its frequency spectrum. For example, in 44,100 Hz audio quantization noise is shifted towards the Nyquist Frequency of 22,050 Hz.

Nondestructive Editing

This type of editing involves a pointer-based system of keeping track of edits. When you delete a section of audio in a nondestructive system, the audio on disk is not actually deleted. Instead, a set of pointers is established to tell the program to skip the deleted section during playback.

Normalize

Refers to raising the volume so that the highest level sample in the file reaches a user-defined level. Use normalization to make sure you are using all of the dynamic range available to you.

Nyquist Frequency

The Nyquist Frequency (or Nyquist Rate) is one half of the sample rate and represents the highest frequency that can be recorded using the sample rate without aliasing. For example, the Nyquist Frequency of 44,100 Hz is 22,050 Hz. Any frequencies higher than 22,050 Hz will produce aliasing distortion in the sample if no anti-aliasing filter is used while recording.

Object Linking and Embedding (OLE)

OLE is a technology developed by Microsoft to allow independent applications to behave as though they are tightly integrated. This allows objects such as Sound Forge audio files to be integrated into other applications such as a Microsoft Word document.

Overview

The Overview is the area on the data window directly under the title bar. The entire length of the overview represents the entire sound file. Cursor, selection, and position information is shown relative to the entire length of the sound file.

One-Shot

One-shots are RAM-based audio clips that are not designed to loop. Things such as cymbal crashes and sound bites could be considered one-shots. Longer files can be treated as one-shots if your computer has sufficient memory.

Pan

To place a mono or stereo sound source perceptually between two or more speakers.

Pause Time

Pause time is the space between CD tracks. This space may contain silence — as in a standard commercially produced CD — or can contain audio — as in a live performance captured on CD.

The Red Book standard calls for two seconds of pause time, but you can edit the default pause time on the CD Settings tab of the Preferences dialog.

Peak Data File

The file created by Sound Forge software when a file is opened for the first time. This file stores the information regarding the graphic display of the waveform so that opening a file is almost instantaneous. This file is stored in the directory where the audio file resides and has an .sfk extension. If this file is not in the same directory as the audio file or is deleted, it will be recalculated the next time you open the file.

Pixel Aspect

The pixel aspect determines whether the pixels are square (1.0) which refers to computers, or rectangular (settings other than 1.000) which typically refers to televisions. The pixel aspect ratio is unrelated to the frame's aspect ratio.

Playlist

The Playlist is a list of regions set to play in a specific order. The Playlist allows for nondestructive editing and rearranging of a sound file quickly and easily. Multiple versions of the playlist can be saved in an external playlist file for easy comparison.

Pre-roll/Post-roll

Pre-roll is the amount of time elapsed before an event occurs. Post-roll is the amount of time after the event. Pre and post-roll have various uses in Sound Forge software. Pre-roll can be added to a crossfade preview to listen to the sound before the crossfade begins to give context to it. Pre-roll can also be used in the Playlist to hear previous regions when playback is initiated from the middle of the Playlist.

Preset

A preset calls up a bulk setting of a function in Sound Forge software. If you like the way you adjusted the EQ but do not want to have to spend the time getting it back for later use, save it as a preset. All presets show up in the drop-down list on the top of most function dialogs in Sound Forge software.

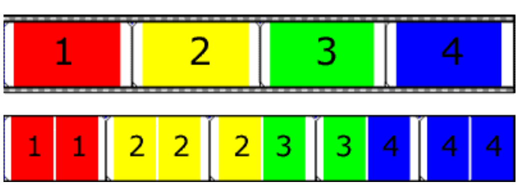
Punch-In

Punching-in during recording means automatically starting and stopping recording at user-specified times.

Pulldown

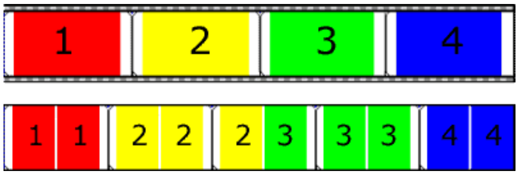
In telecine conversion, fields are added to convert 24 fps film to 30 fps video.

In 2-3 pulldown, for example, the first frame is scanned into two fields, the second frame is scanned into three fields, and so on for the duration of the film. 2-3 pulldown is the standard for NTSC broadcasts of 24p material. Use 2-3 pulldown when printing to tape, but not when you intend to use the rendered video as source media. Removing 2-3 pulldown is inefficient because the pulldown fields that are created for frame 3 span two frames:



24 fps film (top) and resulting NTSC video with 2-3 pulldown fields (bottom)

Use 2-3-3-2 pulldown when you plan to use your rendered video as source media. When removing 2-3-3-2 pulldown, Sound Forge software simply discards frame three and merges the pulldown fields in the remaining frames:



24 fps film (top) and resulting NTSC video with 2-3-3-2 pulldown fields (bottom)

Pulse Code Modulation (PCM)

PCM is the most common representation of uncompressed audio signals. This method of coding yields the highest fidelity possible when using digital storage. PCM is the standard format for .wav and .aif files.

Q Subcode

Compact disc players use the Q channel to display the music playing time. The Q channel is broken down into three modes:

- **Mode 1** Contains the running times from both the beginning of the disc (total disc time) and the beginning of the track (track relative time).
- **Mode 2** Identifies the track number, who recorded the track, where it was recorded and in what year.
- **Mode 3** Identifies UPC media catalog number for the disc.

A special mode of Q data is stored within the lead-in area. This Q data contains information on two- or four- channel format, copy protection, and pre-emphasis.

Quantization

Quantization is the process by which measurements are rounded to discrete values. Specifically with respect to audio, quantization is a function of the analog-to-digital conversion process. The continuous variation of the voltages of a analog audio signal are quantized to discrete amplitude values represented by digital, binary numbers. The number of bits available to describe these values determines the resolution or accuracy of quantization. For example, if you have 8-bit analog-to-digital converters, the varying analog voltage must be quantized to 1 of 256 discrete values; a 16-bit converter has 65,536 values.

Quantization Noise

Quantization noise is a result of describing an analog signal in discrete digital terms (see quantization). This noise is most easily heard in low-resolution digital sounds that have low bit depths and sounds like a shhhhh-type sound while the audio is playing. It becomes more apparent when the signal is at low levels, such as during a fade out.

Reactive Preview

Reactive previews allow for the adjustment of parameters in a function dialog while the preview is playing. When a parameter is changed, the preview will automatically rebuild and continue playback.

Real-Time Streaming Protocol (RTSP)

A proposed standard for controlling broadcast of streaming media. RTSP was submitted by a body of companies including RealNetworks and Netscape.

Redirector File

A metafile that provides information to a media player about streaming media files. To start a streaming media presentation, a Web page will include a link to a redirector file. Linking to a redirector file allows a file to stream; if you link to the media file, it will be downloaded before playback begins.

Windows Media redirector files use the .asx or .wax extension.

Region

A region in Sound Forge software is a subsection of a sound file. You can define any number of regions in a sound file which are stored in the Regions List.

Regions List

The Regions List is simply the list containing all of the regions and markers defined within the sound file. From this list you can preview and edit the regions as well as drag them to the Playlist or to the desktop to create new files from them.

Resample

The act of recalculating samples in a sound file at a different rate than the file was originally recorded. If a sample is resampled at a lower rate, sample points are removed from the sound file, decreasing its size, but also decreasing its available frequency range. Resampling to a higher sample rate, Sound Forge software will interpolate extra sample points in the sound file. This increases the size of the sound file, but does not increase the quality. When downsampling, be aware of aliasing.

See also Aliasing on page 365.

Root Mean Square (RMS)

The Root Mean Square (RMS) of a sound is a measurement of the intensity of the sound over a period of time. The RMS level of a sound corresponds to the loudness perceived by a listener when measured over small intervals of time.

Ruler, Level

The level ruler is the area on a data window to the left of the waveform display. It shows the vertical axis units as a percentage or in decibels.

Ruler, Time

The time ruler is the area on a data window above the waveform display. It shows the horizontal axis units as well as marker, region, and loop tags.

Ruler Tags

Ruler tags are the small tab-shaped controls on the time ruler that represent the location of markers, regions, and loop points in the waveform display.

Sample

The word sample is used in many different (and often confusing) ways when talking about digital sound. Here are some of the different meanings:

- A discrete point in time which a sound signal is divided into when digitizing. For example, an audio CD-ROM contains 44,100 samples per second. Each sample is really only a number that contains the amplitude value of a waveform measured over time.
- A sound that has been recorded in a digital format; used by musicians who make short recordings of musical instruments to be used for composition and performance of music or sound effects. These recordings are called samples. In this Help system, we try to use sound file instead of sample whenever referring to a digital recording.
- The act of recording sound digitally, i.e. to sample an instrument means to digitize and store it.

Sample Dump

A sample dump is the process of transferring sample data between music equipment. Because of the large amounts of data required to store digital sound, sample dumps may take a very long time when using the MIDI Sample Dump Standard (SDS). However, when using the faster SCSI MIDI Device Interface (SMDI) protocol, sample dumps can be performed many times faster.

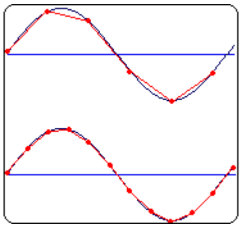
Sample Dump Standard (SDS)

The MIDI Sample Dump Standard is a way to transfer samples between music equipment. Samples transferred with SDS are sent across MIDI cables at the MIDI data rate of 31,250 Hz baud. SMDI is a much faster sample transfer method for musicians.

Sample Rate

The Sample Rate (also referred to as the Sampling Rate or Sampling Frequency) is the number of samples per second used to store a sound. High sample rates, such as 44,100 Hz provide higher fidelity than lower sample rates, such as 11,025 Hz. However, more storage space is required when using higher sample rates.

In the following example, each red dot represents one sample. Because the lower waveform is represented by twice as many samples as the top waveform, the samples are able to better approximate the original waveform.



Sample Size

See Bit Depth on page 367.

Sample Value

The Sample Value (also referred to as sample amplitude) is the number stored by a single sample. The number stored by a single sample:

- In 32-bit audio, these values range from -2147483648 to 2147483647.
- In 24-bit audio, they range from -8388608 to 8388607.
- In 16-bit audio, they range from -32768 to 32767.
- In 8-bit audio, they range from -128 to 127.

The maximum allowed sample value is often referred to as 100% or 0 dB.

Sampler

A sampler is a device that records sounds digitally. Although, in theory, your sound card is a sampler, the term usually refers to a device used to trigger and play back samples while changing the sample pitch.

Secure Digital Music Initiative (SDMI)

The Secure Digital Music Initiative (SDMI) is a consortium of recording industry and technology companies organized to develop standards for the secure distribution of digital music. The SDMI specification will answer consumer demand for convenient accessibility to quality digital music, enable copyright protection for artists' work, and enable technology and music companies to build successful businesses.

SCSI MIDI Device Interface (SMDI)

SMDI is a standardized protocol for music equipment communication. Instead of using the slower standard MIDI serial protocol, it uses a SCSI bus for transferring information. Because of its speed, SMDI is often used for sample dumps.

Shortcut Menu

A context-sensitive menu that appears when you click on certain areas of the screen. The functions available in the shortcut menu depend on the object being clicked on as well as the state of the program. As with any menu, you can select an item from the shortcut menu to perform an operation. Shortcut menus are used frequently in Sound Forge software for quick access to many commands.

Sign-Bit

Data that has positive and negative values and uses zero to represent silence. Unlike the signed format, twos complement is not used. Instead, negative values are represented by setting the highest bit of the binary number to one without complementing all other bits. This is a format option when opening and saving RAW sound files.

Signed

Data that has positive and negative twos complement values and uses zero to represent silence. This is a format option when opening and saving raw sound files.

Signal-to-Noise Ratio

The signal-to-noise ratio (SNR) is a measurement of the difference between a recorded signal and noise levels. A high SNR is always the goal.

The maximum signal-to-noise ratio of digital audio is determined by the number of bits per sample. In 16-bit audio, the signal to noise ratio is 96 dB, while in 8-bit audio its 48 dB. However, in practice this SNR is never achieved, especially when using low-end electronics.

Small Computer Systems Interface (SCSI)

SCSI is a standard interface protocol for connecting devices to your computer. The SCSI bus can accept up to seven devices at a time including CD ROM drives, hard drives and samplers.

Society of Motion Picture and Television Engineers (SMPTE)

SMPTE time code is used to synchronize time between devices. The time code is calculated in hours:minutes:second:frames, where frames are fractions of a second based on the frame rate. Frame rates for SMPTE time code are 24, 25, 29.97 and 30 frames per second.

Sound Card

The sound card is the audio interface between your computer and the outside world. It is responsible for converting analog signals to digital and vice-versa. There are many sound cards available on the market today, covering the spectrum of quality and price. Sound Forge software will work with any Windows-compatible sound card.

Status Format

The status format is the format by which Sound Forge software displays the time ruler and selection times. These include: Time, Seconds, Frames and all Standard SMPTE frame rates. The status format is set for each sound file individually.

Streaming

A method of data transfer in which a file is played while it is downloading. Streaming technologies allow Internet users to receive data as a steady, continuous stream after a brief buffering period. Without streaming, users would have to download files completely before playback.

Telecine

The process of creating 30 fps video (television) from 24 fps film (cinema).

See also Inverse Telecine (IVTC) on page 372; Pulldown on page 375.

Tempo

Tempo is the rhythmic rate of a musical composition, usually specified in beats per minute (BPM).

Threshold

A threshold determines the level at which the signal processor begins acting on the signal. During normalization, levels above this threshold are attenuated.

Time Format

The format by which Sound Forge software displays the time ruler and selection times. These can include: Time, Seconds, Frames and all standard SMPTE frame rates.

Trim/Crop

Trim/Crop is a function that will delete all data in a sound file outside of the current selection. This is a necessary function when dealing with samples to be played by a sampler to get rid of blank time at the beginning and ending of samples.

µ-Law

µ-Law (mu-Law) is a companded compression algorithm for voice signals defined by the Geneva Recommendations (G.711). The G.711 recommendation defines µ-Law as a method of encoding 16-bit PCM signals into a nonlinear 8-bit format. The algorithm is commonly used in European and Asian telecommunications. µ-Law is very similar to A-Law, however, each uses a slightly different coder and decoder.

Undo Buffer

This is the temporary file created before you do any processing to a sound file. This undo buffer allows you to rewrite previous versions of the sound file if you decide you don't like changes you've made to the sound file. This undo buffer is erased when the file is closed or the **Clear Undo/Redo History** command is selected.

Undo/Redo

These commands allow you to change a project back to a previous state, when you don't like the changes you have made, or reapply the changes after you have undone them.

Undo/Redo History

This is a list of all of the functions that have been done to a file that are available to be undone or redone. Undo/Redo History gives you the ability to undo or redo multiple functions as well as preview the functions for quick review of the processed and unprocessed material. This list can be displayed from within the View menu.

Unsigned

Data that has only positive values and uses half the maximum value to represent silence. This is a format option when opening and saving raw sound files.

Virtual MIDI Router (VMR)

A software-only router for MIDI data between programs. Sound Forge software uses the VMR to receive MIDI time code and send MIDI clock. No MIDI hardware or cables are required for a VMR, so routing can only be performed between programs running on the same PC. Sony supplies a VMR with Sound Forge software called the Sony Virtual MIDI Router.

Wave

An digital audio standard developed by Microsoft and IBM. One minute of uncompressed audio requires 10 MB of storage.

Waveform

A waveform is the visual representation of wave-like phenomena, such as sound or light. For example, when the amplitude of sound pressure is graphed over time, pressure variations usually form a smooth waveform.

Waveform Display

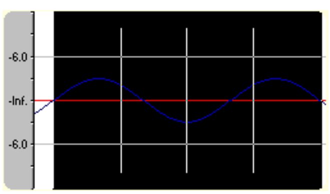
Each event shows a graph of the sound data waveform. The vertical axis corresponds to the amplitude of the wave. For 24-bit audio, they range from -8388608 to 8388607. For 16-bit sounds, the amplitude range is -32,768 to +32,767. For 8-bit sounds, the range is -128 to +127. The horizontal axis corresponds to time, with the leftmost point being the start of the waveform. In memory, the horizontal axis corresponds to the number of samples from the start of the sound file.

Windows Media Format

Microsoft's Windows Media file format that can handle audio and video presentations and other data such as scripts, URL flips, images and HTML tags.

Zero-Crossing

A zero-crossing is the point where a fluctuating signal crosses the baseline.



By making edits at zero-crossings with the same slope, the chance of creating glitches is minimized.

Zipper Noise

Zipper noise occurs when you apply a changing gain to a signal, such as when fading out. If the gain does not change in small enough increments, zipper noise can become very noticeable. Fades are accomplished using 64-bit arithmetic, thereby creating no audible zipper noise.

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