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Shortcuts

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Keyboard Shortcuts

The following sections contain comprehensive lists of keyboard shortcuts available in Sound Forge.

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See also

[Shortcuts](#)

General Keyboard Shortcuts

File Commands

Press	To
Alt+Enter	Show the Properties dialog for the active data window
Control+N	Create a new data window
Control+O	Open a sound file into a new data window
Control+S	Save modified sound data back to the file
Control+W	Close the active data window
Alt+F4	Exit Sound Forge

Window Management Commands

Press	To
Alt+0	Set input focus to the waveform display in the active window.
Alt+1	Set input focus to the Regions List.
Alt+2	Set input focus to the Playlist.
Alt+3	Set input focus to the MIDI Keyboard window.
Alt+4	Set input focus to the Audio Device Mixer window.
Alt+5	Set focus to the Time Display window
Alt+6	Set focus to the Video Preview window
Alt+7	Set focus to the Play Meter window
Alt+8	Set input focus to the Undo/Redo History window
Alt+F5	Restore the Sound Forge application window.
Alt+F10	Maximize the Sound Forge application window.
Control+F5	Restore the active data window.
Control+F6	Go to the next data window.
Control+F10	Maximize the active data window.
Control+Shift+F6	Go to the previous data window.
Shift+F4	Tile the data windows vertically.
Shift+F5	Cascade the data windows.

See also

[Shortcuts](#)

Edit Commands

Press

Control+A
Control+C
Control+E
Control+F
Control+K
Control+Shift+K
Control+M
Control+T
Control+U
Control+V
Control+X
Control+Y
Control+Z
Control+Shift+Z
Delete

To

Select all data in the active window.
Copy the selected data onto the clipboard.
Paste the clipboard contents into a new data window.
Crossfade data from the clipboard with the active window.
Preview Cut or Clear in the active window.
Play to cursor with pre-roll.
Mix data from the clipboard with the active window.
Trim (crop) to the current selection.
Toggle Undo buffer creation for the active window.
Paste data from the clipboard into the active window.
Move (cut) the selected data onto the clipboard.
Repeat last process, effect or tool.
Undo the last action.
Redo the last undone action
Clear (delete) the selected data; nothing placed on the clipboard.

See also

[Shortcuts](#)

Misc. View, Option and Special Commands

Press	To
S	Toggle current selection on and off.
Control+B	Toggle Snap to Zero on and off for the active data window.
Control+Enter	Maximize the width of the active data window.
Control+D	Show the Set Selection dialog.
Control+G	Show the Go To dialog.
Control+L	Show the Loop Tuner for the active data window.
Control+R	Record new data into a data window.
Escape	Stop or cancel the current action (including playback).
F1	Open the Sound Forge Help file to the Contents page.
F10	Make the menu bar active.
Shift+F10	Display a shortcut menu.
F6	Toggle playback scrolling on and off
Shift+F6	Toggle smooth playback scrolling on and off
F8	Toggle Drag and Drop Snapping on and off
F7	Turn on MIDI input sync/trigger
Shift+F7	Turn on MIDI output sync

See also

[Shortcuts](#)

Cursor Movement

Press	To move to
Right Arrow	The next screen pixel.
Left Arrow	The previous screen pixel.
End	The last sample visible in the waveform display.
Home	The first sample visible in the waveform display.
Control+End	The last sample in the data window.
Control+Home	The first sample in the data window.
Page Up	10% of the current view past the cursor position.
Page Down	10% of the current view previous to the cursor position.
Control+Right Arrow	2% of the current view past the cursor position.
Control+Left Arrow	2% of the current view previous to the cursor position.
Numpad +	The next sample.
Numpad -	The previous sample.
Control+Numpad +	10 samples past the current cursor.
Control+Numpad -	10 samples previous to the current cursor.

When Regions, Loops or Markers Exist in the File:

Control+ Left Arrow	The previous region, loop or marker boundary.
Control+ Right Arrow	The next region, loop or marker boundary.

See also

[Shortcuts](#)

Selecting Data

Press

Shift+Right Arrow
Shift+Left Arrow
Shift+End
Shift+Home
Control+Shift+End
Control+Shift+Home
Shift+Page Up
Shift+Page Down
Control+Shift+Right Arrow
Control+Shift+Left Arrow
Shift+Numpad +
Shift+Numpad -
Control+Shift+Numpad +
Control+Shift+Numpad -

To select from the cursor to

The next screen pixel.
The previous screen pixel.
The last sample visible in the waveform display.
The first sample visible in the waveform display.
The last sample in the data window.
The first sample in the data window.
10% of the current view past the cursor position.
10% of the current view previous to the cursor position.
2% of the current view past the cursor position.
2% of the current view previous to the cursor position.
The next sample.
The previous sample.
10 samples past the current cursor.
10 samples previous to the current cursor.

When Regions, Loops or Markers Exist in the File:

Control+Shift+Left Arrow The previous region, loop or marker boundary.
Control+Shift+Right Arrow The next region, loop or marker boundary.

See also

[Shortcuts](#)

Navigation and Playback

Press	To
Control+<Number>	Save a view in cell <Number> where <Number> ranges from 1 to 8.
<Number>	Restore a view using cell <Number> where <Number> ranges from 1 to 8.
Up Arrow	Increase time magnification (zoom in closer to data).
Down Arrow	Decrease time magnification (zoom out farther from data).
Shift+Up Arrow	Increase level magnification
Shift+Down Arrow	Decrease level magnification
Control+Up Arrow	Zoom Selection if a selection exists, Zoom In Full if no selection.
Control+Down Arrow	Zoom Normal (zooms to default zoom ratio set in Preferences).
Control+Shift+Up Arrow	Pan data window up if zoomed in vertically.
Control+Shift+Down Arrow	Pan data window down if zoomed in vertically.
Numpad 5 or *	Switch cursor to opposite end of selection
Tab	Cycle stereo selection from left channel to right channel to both channels
Shift+Tab	Cycle stereo selection from both channels to right channel to left channel.
[(open bracket)	Set Mark In at the current cursor position.
] (open bracket)	Set Mark Out at the current cursor position.
M or \ (backslash)	Drop a marker at the current cursor position.
C or . (period)	Center the cursor in the waveform display.
Spacebar	Play or Stop the current data window in default mode.
Shift+Spacebar	Play All or Pause the current data window.
Control+Spacebar	Switch playmode between Normal, Looped and As Sample.
Enter	Pause playback; keep the cursor at the current position.
Escape	Stop playback of the current data window.
Control+K	Preview Cut (skip selection on playback with pre-roll).
Control+Shift+K	Play to cursor with pre-roll.

See also

[Shortcuts](#)

Regions List Shortcuts

Press	To
Spacebar	Play or Stop the active marker or region.
Enter	Edit the active marker or region.
Delete	Delete the active marker or region.
` (grave accent)	Cycle through the Regions List display formats.
R	Create region from the current selection
Shift+R	Create region without bringing up dialog.
L	Create loop from the current selection
Shift+R	Create loop without bringing up dialog.

See also

[Shortcuts](#)

Playlist Shortcuts

Press

Spacebar

Enter

Delete

+ (plus sign)

- (minus sign)

* (asterisk)

/ (forward slash)

` (grave accent)

To

Play (stop) from the active Playlist entry.

Edit the active Playlist entry.

Delete the active Playlist entry.

Add one to the active Playlist entry play count.

Subtract one from the active Playlist entry play count.

Add or remove a Stop Point on the active Playlist entry.

Toggle Pre-roll on and off for the Playlist.

Cycle through the Playlist display formats.

See also

[Shortcuts](#)

Shortcut Menus

Shortcut menus are menus that appear when you right-click on certain areas of the screen. They allow you to quickly access important functions. Try right-clicking on any area in Sound Forge...chances are that a Shortcut Menu will appear.

[Sound Forge Desktop](#)

[Waveform Display](#)

[Playbar Selection Status](#)

[Regions List](#)

[Playlist](#)

[Time Ruler](#)

[Display Properties](#)

[Level Ruler](#)

[Level Toolbar](#)

[Marker Tag](#)

[Region Tag](#)

[Loop Tag](#)

[Status Bars](#)

[Wave, MIDI, and Auxiliary Mixers](#)

[Keyboard MIDI Out](#)

[Play Meters](#)

[Preview Configuration](#)

[Record Dialog](#)

[Time Display](#)

[Video Strip](#)

[Video Preview](#)

[Summary Information Picture](#)

[Edit Tool Selector](#)

See also

[Shortcuts](#)

Sound Forge Desktop Shortcut Menu

This shortcut menu is reached by right-clicking on the desktop workspace of Sound Forge. This menu has options for sound file management on your workspace. You can open recently used files from the *Recent Files* option. The *Preferences* dialog can quickly be reached from this menu as well.

See also

[Shortcut Menus](#)

Waveform Display Shortcut Menu

This shortcut menu is reached by right-clicking on the waveform display of a sound file. It is a way of quickly reaching the following commands and dialogs: *Zoom In Full/Normal/Out Full*, *Paste*, *Crossfade*, *Mix*, *Cursor Snap to Time/Zero*, *Cursor Go To*, *Scroll Playback*, *Properties*.

See also

[Shortcut Menus](#)

Cursor|Set to Start

Moves the cursor to the start of the selection.

See also

[Waveform Display Shortcut Menu](#)

Cursor|Set to End

Moves the cursor to the end of the selection.

See also

[Waveform Display Shortcut Menu](#)

Playbar Selection Status Shortcut Menu

This shortcut menu is reached by right-clicking on any of the playbar's selection status fields. It allows you to set the current status format.

See also

[Shortcut Menus](#)

Regions List Shortcut Menu

This shortcut menu is reached by right-clicking on the *Regions List*. It allows you to perform the following *Regions List* operations: *Add, Delete, Edit, Replicate, Split, Update, Markers to Regions, Open, Save As, Copy onto Clipboard*.

See also

[Shortcut Menus](#)

Playlist Shortcut Menu

This shortcut menu is reached by right-clicking on the *Playlist*. It allows you to perform the following Playlist operations: *Add, Delete, Edit, Replicate, Stop Point, Pre-roll Playback, Open, Save As, Copy onto clipboard, Convert to New.*

See also

[Shortcut Menus](#)

Time Ruler Shortcut Menu

This shortcut menu is reached by right-clicking on the *Time Ruler*. It allows you to reach the *Add Marker/Region* or *Edit Sample* dialog for adding regions, markers, and loop points, as well as changing the status format.

See also

[Shortcut Menus](#)

Display Properties Shortcut Menu

This shortcut menu is reached by right-clicking on the *Edit Tool* selector at the upper left-hand corner of the data window. It allows you to set options described under *Properties|Display* in the File Menu.

See also

[Shortcut Menus](#)

Level Ruler Shortcut Menu

This shortcut menu is reached by right-clicking on the *Level Ruler*. It allows you to change the labeling used, as well as various level zooming operations.

[Zoom Out Full](#)

[Zoom Level|Window](#)

[Zoom Level|Selection](#)

[Label in Percent](#)

[Label in dB](#)

See also

[Shortcut Menus](#)

Level Ruler|Label in Percent

When checked, the *Level Ruler* will be labeled in percent.

See also

[Level Ruler Shortcut Menu](#)

Level Ruler|Label in dB

When checked, the *Level Ruler* will be labeled in decibels.

See also

[Level Ruler Shortcut Menu](#)

Level Toolbar Shortcut Menu

This shortcut menu is reached by right-clicking on the *Levels* toolbar. It sets the status of the display for the toolbar. The *Levels* toolbar can be set to one of these formats:

[Integer](#)

[Percent](#)

[dB](#)

[Peak](#)

[RMS](#)

See also

[Shortcut Menus](#)

Level Toolbar|Integer

Displays the sample value at the cursor as an integer. The range is from -32768 to 32767 in 16-bit audio and -128 and 127 in 8-bit audio.

See also

[Level Toolbar Shortcut Menu](#)

Level Toolbar|Percent

Displays the sample value at the cursor as a percentage. The range is from -100 to 100%.

See also

[Level Toolbar Shortcut Menu](#)

Level Toolbar|dB

Displays the sample value at the cursor as a dB value. 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).

See also

[Level Toolbar Shortcut Menu](#)

Level Toolbar|Peak

Displays the peak sample value in the vicinity of the cursor as a dB value. The amount of time which is scanned by default is 50 milliseconds. This can be set on the *Status* page of the *Preferences* folder.

See also

[Level Toolbar Shortcut Menu](#)

Level Toolbar|RMS

Displays the Root Mean Square (RMS) of the sample values in the vicinity of the cursor as a dB value. RMS is a more accurate measurement of loudness than the peak value. The amount of time which is scanned by default is 50 milliseconds. This can be set on the *Status* page of the *Preferences* folder.

The maximum value of 0 dB is only achieved by a square wave (or DC offset) of maximum amplitude. A maximum amplitude sine wave reaches only -3 dB. Values below -96 dB can be achieved by combination of silence and very low-levels, and only complete silence will register negative infinity.

See also

[Level Toolbar Shortcut Menu](#)

Marker Tag Shortcut Menu

This shortcut menu is reached by right-clicking on a marker tag in the ruler.

[Go To](#)
[Delete](#)
[Edit](#)
[Update](#)

See also

[Shortcut Menus](#)

Region Tag Shortcut Menu

This shortcut menu is reached by right-clicking on a region tag in the ruler.

[Edit](#)

[Select](#)

[Delete](#)

[Update](#)

[Split](#)

See also

[Shortcut Menus](#)

Loop Tag Shortcut Menu

This shortcut menu is reached by right-clicking on a loop tag in the ruler.

[Update](#)

[Edit](#)

[Delete](#)

[Select](#)

See also

[Shortcut Menus](#)

Status Bar Shortcut Menus

These shortcut menus allow you to change the playback rate, sample size, channels (*Stereo/Mono*), and status format by right-clicking on the status bar fields.

[Sample Rate](#)

[8-Bit](#)

[16-Bit](#)

[Mono](#)

[Stereo](#)

[Status](#)

See also

[Shortcut Menus](#)

Status Bar|Sample Rate

Use this shortcut menu to select the playback rate of the current sound file. Note that if you do not *Resample* the data, the pitch will change also.

See also

[Status Bar Shortcut Menu](#)

Status Bar|Status

Use this shortcut menu to select the time or sample [status format](#) used.

See also

[Status Bar Shortcut Menu](#)

Status Bar|8-Bit

Use this shortcut menu to change the bits per sample of a sound file to 8.

See also

[Status Bar Shortcut Menu](#)

Status Bar|16-Bit

Use this shortcut menu to change the bits per sample of a sound file to 16.

See also

[Status Bar Shortcut Menu](#)

Status Bar|Mono

Use this shortcut menu to make the sound file contain only one channel.

See also

[Status Bar Shortcut Menu](#)

Status Bar|Stereo

Use this shortcut menu to make the sound file contain two channels.

See also

[Status Bar Shortcut Menu](#)

Wave, MIDI, and Auxiliary Mixer Shortcut Menus

This shortcut menu is reached by right or left-clicking on the *Wave*, *MIDI*, or *Aux* headings in the *Mixer* window. They allow you to select the *Wave*, *MIDI*, or *Auxiliary* device that the volume fader controls. Many sound cards do not allow for fader controls, in which case this feature will not work.

See also

[Shortcut Menus](#)

Keyboard MIDI Out Shortcut Menu

This shortcut menu is reached by right or left-clicking on the *MIDI Out* button on the *Keyboard*. It allows you to set the device to which the *MIDI Keyboard* will transmit.

See also

[Shortcut Menus](#)

Play Meters Shortcut Menu

Right-click on the *Play Meters* to reach this menu will allows you to select the *Play Meter* resolution and other options including: *Reset Clip*, *Show Labels*, *Hold Peaks*, *Hold Valleys* and *Always on Top*.

[Reset clip](#)

[Resolution](#)

[Show Labels](#)

[Show Peaks](#)

[Show Valleys](#)

[Always on Top](#)

See also

[Shortcut Menus](#)

Preview Configuration Shortcut Menu

This shortcut menu is reached by right-clicking on any unused area in a dialog which has Preview capabilities.

[Left](#)

[Right](#)

[All](#)

[Pre-Roll](#)

[Post-Roll](#)

[Reset All](#)

[Save](#)

[Configure](#)

See also

[Shortcut Menus](#)

Preview|Pre-roll

When enabled, sound data preceding the current selection is played before the processed data during previews.

See also

[Preview Configuration Shortcut Menu](#)

Preview|Post-roll

When enabled, sound data following the current selection is played after the processed data during previews.

See also

[Preview Configuration Shortcut Menu](#)

Preview|Left

Process left channel only when using a stereo sound file.

See also

[Preview Configuration Shortcut Menu](#)

Preview|Right

Process right channel only when using a stereo sound file.

See also

[Preview Configuration Shortcut Menu](#)

Preview|Both

Process both channels when using a stereo sound file.

See also

[Preview Configuration Shortcut Menu](#)

Preview|Reset All

Resets all of the dialog's parameters to their default values.

See also

[Preview Configuration Shortcut Menu](#)

Preview|Save Current

Saves the current dialog settings to the *Untitled* preset.

See also

[Preview Configuration Shortcut Menu](#)

Preview|Configuration

Takes you to the *Preview Configuration* dialog.

See also

[Preview Configuration Shortcut Menu](#)

Record Dialog Shortcut Menu

This shortcut menu is reached by right-clicking anywhere in the Record window. The Remote Record shortcut menu is similar, with options not available in the dialog available from the menu.

- [Reset clip](#)
- [Resolution](#)
- [Show Labels](#)
- [Show Peaks](#)
- [Show Valleys](#)
- [Calibrate DC Adjust](#)
- [Review Pre/Post Roll](#)
- [Go to](#)
- [Selection](#)
- [Window](#)
- [New Window](#)

See also

- [Shortcut Menus](#)

Record Meters|Reset Clip

This function resets the clip indication on the *Record Meters*. After a clip occurs, the clip indication will remain lit until the *Reset Clip* function is invoked. You can also click on the clip light or levels to clear it.

See also

[Options Menu](#)

Record Meters|Resolution

These settings display the minimum and maximum readings for the *Record Meters*. Changing the resolution of the meters is helpful for metering sound files of varying dynamism.

[-90 to 0 dB](#)

[-78 to 0 dB](#)

[-60 to 0 dB](#)

[-42 to 0 dB](#)

[-24 to 0 dB](#)

See also

[Shortcut Menus](#)

Record Meters|-90 to 0 dB

Determines the bottom-most level that will be displayed. A value of -90 dB is the absolute minimum peak value in 16-bit audio, corresponding to a sample value of +/- 1.

See also

[Record Dialog Shortcut Menu](#)

Record Meters|-78 to 0 dB

Determines the bottom-most level that will be displayed. A value of -78 dB (0.012%) will be the smallest value displayed.

See also

[Record Dialog Shortcut Menu](#)

Record Meters|-60 to 0 dB

Determines the bottom-most level that will be displayed. A value of -60 dB (0.1 %) will be the smallest value displayed.

See also

[Record Dialog Shortcut Menu](#)

Record Meters|-42 to 0 dB

Determines the bottom-most level that will be displayed. A value of -42 dB (0.8 %) will be the smallest value displayed.

See also

[Record Dialog Shortcut Menu](#)

Record Meters|-24 to 0 dB

Determines the bottom-most level that will be displayed. A value of -24 dB (0.6 %) will be the smallest value displayed.

See also

[Record Dialog Shortcut Menu](#)

Record Meters|Show Labels

This option turns on or off the markings on the *Record Meters*.

See also

[Record Dialog Shortcut Menu](#)

Record Meters|Hold Peaks

This option turns on or off the markings to indicate the highest peaks in the sound file.

See also

[Record Dialog Shortcut Menu](#)

Record Meters|Hold Valleys

This option turns on or off the markings to indicate the lowest peaks in the sound file.

See also

[Record Dialog Shortcut Menu](#)

Gap Detection|Ignore

If a gap is detected while recording, no action is taken.

See also

[Record Dialog Shortcut Menu](#)

Gap Detection|Mark

If a gap is detected while recording, a marker is dropped at the estimated location.

See also

[Record Dialog Shortcut Menu](#)

Gap Detection|Stop

If a gap is detected while recording, recording is stopped.

See also

[Record Dialog Shortcut Menu](#)

Blinking Status

Enables the blinking record status underneath the record meters.

See also

[Record Dialog Shortcut Menu](#)

Record|Calibrate DC Adjust

Calculates a new DC Offset to use from the incoming audio.

See also

[Record Dialog Shortcut Menu](#)

Record|Review Pre/Post Roll

Enables or disables Pre-Roll and Post-Roll on playback.

See also

[Record Dialog Shortcut Menu](#)

Record|Go to

Takes you to the Go To dialog, which allows you to set a Record start time.

See also

[Record Dialog Shortcut Menu](#)

Record|Selection

Takes you to the Selection dialog, which allows you to set a Record start and end time.

See also

[Record Dialog Shortcut Menu](#)

Record|Window

Takes you to the Record Window dialog, which allows you to set a new Data Window to record to.

See also

[Record Dialog Shortcut Menu](#)

Record|New Window

Sets Record to write data to a new sound file.

See also

[Record Dialog Shortcut Menu](#)

Record|Aggressive Update

When enabled, the Record meters will respond much more rapidly to incoming audio. However, with some sound cards, this can lead in some cases to peaks not being detected. If you want complete security that your recording never reached 0 dB, turn this off.

See also

[Record Dialog Shortcut Menu](#)

Time Display Shortcut Menu

This shortcut menu is reached by right-clicking on the *Time Display* window. It allows you to set the display format for the *Time Display*.

[Time Display|Playlist Position](#)

[Time Display|Position](#)

[Time Display|Passive Update](#)

[Time Display|Position](#)

See also

[Shortcut Menus](#)

Video Strip Shortcut Menu

This shortcut menu is reached by right-clicking on the video strip above the waveform display.

Video Strip|Copy

Copies the selected frame as a bitmap to the clipboard. You could, for example, then paste it into the *Summary* information folder (*Properties* in the File menu) by right-clicking over the shot you want to copy.

[Video Strip|Copy](#)

[Video Strip|Go To](#)

[Video Strip|Animate](#)

[Video Strip|Number Frames](#)

See also

[Shortcut Menus](#)

Video Strip|Go To

Moves the cursor to the beginning of the frame.

See also

[Video Strip Shortcut Menu](#)

Video Strip|Animate

Displays the frame closest to the current cursor position during editing and playback instead of a static overview frame.

See also

[Video Strip Shortcut Menu](#)

Video Strip|Number Frames

Displays frame number on each frame.

See also

[Video Strip Shortcut Menu](#)

Video Preview Shortcut Menu

This shortcut menu is reached by right-clicking on the *Video Preview* window. It allow you to set various options of the video display. A complete list and explanation of these options is available in the Reference section under the Options menu item.

See also

[Video Strip Shortcut Menu](#)

Summary Information Picture Shortcut Menu

This shortcut menu is reached by right clicking on the picture area in File|Properties|Summary Information. It allows for copy, paste, and clear operations on the image.

See also

[Shortcut Menus](#)

Summary Information|Copy

Use this to copy the current summary information image onto the clipboard.

See also

[Summary Information Shortcut Menu](#)

Summary Information|Paste

Use this to embed a picture in a sound file by pasting a bitmap, cursor, or icon from the clipboard.

See also

[Summary Information Shortcut Menu](#)

Summary Information|Clear

Use this to delete a picture from a sound files embedded summary information.

See also

[Summary Information Shortcut Menu](#)

Edit Tool Selector Shortcut Menu

This shortcut menu is reached by right clicking on the *Edit Tool Selector*. The *Edit Tool Selector* is the icon on the upper left hand side of the data window which allows you to switch between the Edit, Magnify, and Pencil tools.

The *Edit Tool Selector* shortcut menu allows you to enable and disable various parts of the Data Window, such as scroll bars and rulers.

See also

[Shortcut Menus](#)

Edit Tool Selector|Status

Use this to show or hide the Playbar in the current data window.

See also

[Edit Tool Selector Shortcut Menu](#)

Edit Tool Selector|Time Ruler

Use this to show or hide the Time Ruler in the current data window.

See also

[Edit Tool Selector Shortcut Menu](#)

Edit Tool Selector|Time Zoom

Use this to show or hide the Time Zoom spinner in the current data window.

See also

[Edit Tool Selector Shortcut Menu](#)

Edit Tool Selector|Overview

Use this to show or hide the Overview bar in the current data window.

See also

[Edit Tool Selector Shortcut Menu](#)

Edit Tool Selector|Video

Use this to show or hide the Video Strip in the current data window.

See also

[Edit Tool Selector Shortcut Menu](#)

Edit Tool Selector|Level Ruler

Use this to show or hide the Level Ruler in the current data window.

See also

[Edit Tool Selector Shortcut Menu](#)

Edit Tool Selector|Level Zoom

Use this to show or hide the Level Zoom spinner in the current data window.

See also

[Edit Tool Selector Shortcut Menu](#)

Edit Tool Selector|Scrollbar

Use this to show or hide the Scrollbar in the current data window.

See also

[Edit Tool Selector Shortcut Menu](#)

Edit Tool Selector|Data Window Only

When this is checked, only the Waveform Display will be shown in the selected data window. All other controls such as rulers and scrollbars will be hidden.

See also

[Edit Tool Selector Shortcut Menu](#)

Drag and drop shortcuts

Using drag and drop allows you to quickly perform operations crossing between open data windows, the *Playlist*, the *Regions List*, and the *Time Ruler*.

[Drag to New](#)

[Drag Mix](#)

[Drag Crossfade](#)

[Drag Paste](#)

[Drag to Regions List](#)

[Drag to Playlist](#)

[Drag to Time Ruler](#)

See also

[Shortcuts](#)

Drag to New

To create a new file from the current selection, drag the selection to an open area of the Sound Forge desktop. You can also drag regions from the *Regions List* to the desktop.

See also

[Drag and Drop Shortcuts](#)

Drag Mix

To mix a selection into another data window, drag the selection from the source to the destination window. Processing begins at the point where you drop the region in the destination file. When mixing a mono selection into a stereo file, you can mix the selection to both channels by dropping the selection on the destination centerline. Otherwise, the selection will be mixed into either the left or right channel only.

See also

[Drag and Drop Shortcuts](#)

Drag Crossfade

To crossfade a selection into another data window, drag the selection from the source to the destination window. Processing begins at the point where you drop the region in the destination file. Press the *Control* key while in the destination window. When crossfading a mono selection into a stereo file, you can crossfade the selection to both channels by dropping the selection on the destination centerline. Otherwise, the selection will be crossfaded into either the left or right channel only.

See also

[Drag and Drop Shortcuts](#)

Drag Paste

To paste a selection into another data window, drag the selection from the source to the destination window. Processing begins at the point where you drop the region in the destination file. Press the *Alt* key while in the destination window. When pasting a mono selection into a stereo file, you can paste the selection to both channels by dropping the selection on the destination centerline. Otherwise, the sound data will be placed in either the left or right channel only, with silence in the other.

See also

[Drag and Drop Shortcuts](#)

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*.

See also

[Drag and Drop Shortcuts](#)

Drag to Playlist

To add a region in the *Regions List* to the *Playlist*, drag it from the waveform display or the *Regions List* to the *Playlist*. You can also drag regions within the *Playlist* to rearrange the playback order.

See also

[Drag and Drop Shortcuts](#)

Drag to Time Ruler

To create a regions, drag the current selection to the *Time Ruler*.

See also

[Drag and Drop Shortcuts](#)

Other Mouse Shortcuts

[Select All](#)

[Zoom Both Time and Level](#)

[Magnify Mode](#)

[Return Control Value to Default](#)

[Fine Tune Control Value](#)

[Preview](#)

[Main Status Bar](#)

[Selection Status Bar](#)

[Go To Marker](#)

[Set Selection to Region/Loop](#)

[Edit Region or Marker](#)

[Edit Playlist](#)

[Play Normal Button](#)

[Slow and Fast Selection Scroll Toggle](#)

[Lock Loop/Region length while dragging](#)

[Ruler scrolling speed control](#)

See also

[Shortcuts](#)

Select All

Double-click on the waveform display to select the entire sound file. Triple-click when regions, loops or markers are present.

See also

[Other Mouse Shortcuts](#)

Zoom Both Time and Level

Double-click on the *Level Ruler* to zoom the current selection both vertically and horizontally. If no selection exists, the entire waveform display data is zoomed. Double-click again to *Zoom Out Full* vertically and *Zoom Normal* horizontally.

See also

[Other Mouse Shortcuts](#)

Magnify Mode

To zoom in to a section, select an area while holding the *Control* key. This will switch Sound Forge to *Magnify* mode, which zooms to fit the selected area.

See also

[Other Mouse Shortcuts](#)

Return Control Value to Default

Double-click on a trackbar, fader, or spinner to set it to its default value.

See also

[Other Mouse Shortcuts](#)

Fine Tune Control Value

To fine tune a trackbar, fader, or spinner, hold down both right and left mouse buttons when dragging. This can also be done if you hold down the *Control* key while dragging.

See also

[Other Mouse Shortcuts](#)

Preview

Shift-click on the *Preview* button to hear the original sound. This is equivalent to checking the *Bypass* check box.

Control-click on the *Preview* button to reach the *Preview Configuration* dialog.

See also

[Other Mouse Shortcuts](#)

Main Status Bar

Double-click on the *Sample Rate, Sample Size, or Channels (Stereo/Mono)* field to reach the *Properties* dialog.

See also

[Other Mouse Shortcuts](#)

Selection Status Bar

Double-click on the left-most field to reach the *Go To* dialog. Double-click on the other two fields to reach the *Set Selection* dialog.

See also

[Other Mouse Shortcuts](#)

Go To Marker

Double-click on a marker tag in the ruler to move the cursor to the marker's position.

See also

[Other Mouse Shortcuts](#)

Set Selection to Region/Loop

Double-click on a region or loop tag in the ruler to change the current selection to the region or loop end points.

See also

[Other Mouse Shortcuts](#)

Edit Region or Marker

Double-click on a region or marker in the *Regions List* to reach the *Edit Region/Marker* dialog.

See also

[Other Mouse Shortcuts](#)

Edit Playlist

Double-click on a *Playlist* entry to reach the *Edit Playlist* dialog.

See also

[Other Mouse Shortcuts](#)

Play Normal Button

Control-click to *Preview Cut* (skip selection) on playback. *Control*+shift-click to play to cursor with pre-roll.

See also

[Other Mouse Shortcuts](#)

Slow and Fast Selection Scroll Toggle

When making a selecting past the end or beginning of the waveform display, while holding down the left mouse button, click on the right mouse button to toggle between fast scrolling and slow scrolling.

See also

[Other Mouse Shortcuts](#)

Lock Loop/Region length while dragging

When dragging a loop or region with the Region or Loop Ruler tags, holding down the Shift key will move both start and end points together. This preserves the region or loop duration.

See also

[Other Mouse Shortcuts](#)

Ruler scrolling speed control

When dragging the Time Ruler for horizontal scrolling, once you move the mouse outside of the data window, the further you move the mouse, the faster the scrolling occurs. The speed can also be controlled by moving the mouse up and down while scrolling.

See also

[Other Mouse Shortcuts](#)

Sound Forge for 32 Bit Windows

This section discusses details specific to the 32 Bit version of Sound Forge 4.0.

Sound Forge for 32 Bit Windows is the first **true 32 Bit flat model** digital sound editor available for the PC. The performance differences from a 16 Bit sound editor are amazing. Processing is often 2 to 3 times faster with the 32 Bit version of Sound Forge.

Sound Forge for 32 Bit Windows has been designed to run on the following platforms:

- Microsoft Windows 95.
- Microsoft Windows NT 3.5 or later. Alpha AXP, MIPS, PowerPC and x86 processors are supported.

When running Sound Forge for 32 Bit Windows on the Windows NT operating system, some limitations may be present. *These limitations result from a lack of compatible hardware and software available for 32 Bit Windows or are a direct result of the Windows NT Operating System implementation.* These limitations are summarized below.

Limitations under Microsoft Windows NT

The following is a list of limitations of Sound Forge for 32 Bit Windows when running under Windows NT:

1. The Sonic Foundry Virtual MIDI Router (VMR) is not supported under Windows NT.
2. Not all sound and MIDI cards have drivers available for Windows NT. Contact your manufacturer for availability of Windows NT drivers.
3. Synchronization with 16 bit applications (sequencers) running in the Windows on Windows (WOW) layer will not perform as well as running the 16 bit version of Sound Forge under Windows 3.x with a 16 bit sequencer.

What's new in version 4.0

Below are just a few of the new features in Sound Forge 4.0. This is not a comprehensive list of changes but simply a few of the key features that you might want to check out. Many of these new features are ones that our users have requested. Keep those suggestions coming!

General Features

Direct Mode editing

You can now perform edits directly on the original file, meaning that file opens and saves are almost instantaneous. In previous versions of Sound Forge, a backup of the file being opened was always created. For very large files, creating a backup could take more than a few minutes. The downside of working directly on a file is that the original data is immediately changed whenever you make an edit. This is why it's so nice to have multiple undo buffers. To work on a file directly, check the Operate directly on the sound file option while opening a file in the *File Open* dialog. If you don't want to work directly on files, simply leave the option unchecked.

Multi-level undo/redo

There is now no limit to the number of undo operations available. Use the *Undo* and *Redo* options in the Edit menu to restore the sound file to any edited stage. The *Undo/Redo History* window (View menu) makes it easy to select from all the previously existing states and in most cases you can even preview what the undo or redo will sound like.

AVI file support

To open a Video for Windows AVI file, select AVI as the file type from the *File Open* dialog. The video frames will be shown on top of the Waveform Display in the Video Strip. To view the video during playback, open the *Video Preview* window (View Menu).

Super fast screen redraws

Need we say more?

Auto-Scroll during playback

When this option is turned on, the sound data being played will always be displayed on the screen. If your computer is not fast enough to redraw the window, you can disable this feature in the Options menu.

Level Zoom (vertical) and scrolling ruler

The Level Ruler on the left hand side of the Waveform Display is useful for estimating sample values in percentage or decibels. It is especially useful now that you can zoom and scroll vertically for viewing very low level signals. To zoom in vertically, use the *Level Zoom* spinner on the lower left hand corner of the Data Window or hit the *Shift-Up Arrow* or *Shift-Down Arrow* keys. Once you're zoomed in, dragging the Level Ruler allows you to view different ranges. Double-clicking on the Level Ruler centers the 0 axis.

Drag and Drop Mix/Paste/Crossfade shows destination position

When you drag from one sound file to another, a cursor and region now appear on the destination file that show where data will be mixed, pasted, or crossfaded. Remember, hold down the *Control* key before dropping to crossfade or the *Alt* key for pasting, otherwise a Mix will occur. You can also click with the right mouse button during dragging to quickly switch between modes.

Save and Open Workspaces

Any sound files open when you close Sound Forge will be automatically re-opened next time you open Sound Forge. This feature can be turned off from the *General Preferences* page. You can also save and load workspaces from the *Open* and *Save Workspace* dialogs (File menu). The workspace files (.sfw) store the files opened, position, size, and views.

Play meter

A very fast and accurate playback meter is useful for maximizing dynamic range and achieving the hottest

signal possible without distortion. To open the meter, select *Play Meters* from the View menu. The meter will indicate if clipping ever occurs. It even works while previewing effects.

Audio event locator

This feature allows you to find audio events by clicking and dragging the mouse on the Overview (the thin strip above the Time Ruler). Similar to a scrub, playback follows mouse movement across the Overview and loops when the mouse is still. Releasing the mouse stops playback and centers the cursor on the last start position in the Waveform Display.

Internet Audio Support (Java and RealAudio)

Updated support has been added for the NeXT/Sun .au format for working with Java scripts. We've also added RealAudio support for streaming audio over the Internet.

Send Mail and OLE Drag and Drop

To send a sound file via email to somebody, select *Send* from the File menu. Dragging a file outside of the Sound Forge workspace creates a true OLE object which can be embedded on any OLE container (such as Wordpad). To do so, you must drag from the OLE button in the lower right corner of the Data Window.

Effects/Processing/Tools

Reactive Previews

While listening to a preview, you can now tweak any parameter and a new preview will be generated automatically and played without the need to hit the *Preview* button again. You can also have a pre-roll and post-roll on the preview for listening to transitions. If you're into synchronization the previews now also generate MTC while playing.

Paragraphic EQ

The *Paragraphic EQ* is a new dialog which allows you to do four bands of parametric filtering plus Low and High-Pass filtering. It displays a Gain vs. Frequency graph which helps to visualize the final results. It also uses a new, fast, and accurate algorithm.

Reverb, Time compression/expansion, and Pitch change use new algorithms

These effects have been tremendously improved.

Crossfade Loop function.

This tool greatly simplifies the process of finding good sample loops.

Phaser and Wah-wah

Check them out along with Flange in the Effects menu.

Graphic Dynamics now uses graph

An input vs. output graph allows for very flexible dynamics processing.

Multi-band Dynamics

A four band dynamic compressor/limiter is indispensable for de-essing, de-popping, and many mastering applications.

Dithered/noise shaped fades

By far the most noticeable need for dithering is when fading from or to absolute silence. The *Graphic Fade* function now allows you to perform advanced dithering and noise-shaping for silky smooth fades. Of course, you can apply dithering to the entire file as well.

Stereo expansion and MS mixing

The *Graphic Pan* dialog now also allows for stereo expansion of Left/Right recordings and mixing of MS recordings.

Normalize

Normalize now allows you to use Peak or RMS levels and can perform intelligent compression to maximize the volume of your sound files.

Smooth/Enhance

Quickly add or remove high frequency content.

DTMF dial tone generator

Generate touch-tone triggers in Sound Forge.

Channel converter

Use this dialog to convert between mono and stereo formats or to adjust the left/right balance.

Graphic EQ.

It's faster and more accurate.

Mix and Crossfade

Both now include the ability to preview their results as well as the ability to fine tune the starting position.

Find

You can now find the maximum sample value in a range or file.

Chorus and Flange

Improvements have been made to apply smoother pitch modulation.

Resample

Includes a new and improved anti-alias filter as well as the ability to filter after upsampling.

Record

Sync support for Record dialog.

It is now possible to send and receive SMPTE/MTC time code while recording. Also, a SMPTE output offset can be set in the [MIDI/Sync Preferences](#).

Auto glitch-detect in record dialog

While recording, if Sound Forge detects a high probability of a glitch having occurred due to your computer not keeping up with the incoming data, it either stops recording or drops markers on the suspicious spot. These options are selected by right mouse clicking anywhere in the record dialog.

Drop Marker in record dialog

While recording, hitting the M key or pressing the *Drop Marker* button (next to the play button in the Record dialog) can be used to create a marker in the file.

DC Offset auto-correct during record

To calibrate Sound Forge's auto-DC offset correction, press the DC Adjust button in the record dialog while monitoring input. From then on, the detected sound card DC offset will be removed from all recordings.

Markers and Regions

Markers to Regions function

The Markers to Regions function converts all markers in a file to consecutive regions. This is handy after dropping markers while recording or during playback.

Split region function.

You can split a region into two by first putting the cursor where you wish the split to occur and selecting the region in the Regions List. Then, select Region List|Split from the Special menu.

Snap to markers and regions when using drag and drop

When doing drag and drop mix, crossfade, or pasting, if the cursor is near a marker or region in the destination file it will snap to it. You can also snap to major ticks on the time ruler as well as to a specific frame in the Video Strip.

Markers, Regions, and Loop points annotations

The name of markers, regions, and loops can be displayed right on the Waveform Display.

Miscellaneous Marker and Region List functions

Markers and Regions are now stored in the clipboard for when you Copy, Mix, or Create New.

Double-clicking on the Waveform Display selects from previous to next marker or loop/region.

Shift-Double Click on a marker moves the marker to the current cursor position.

Ctrl-Left or Ctrl-Right keys jump to the next marker, region, or loop if they exist.

Typing 'R' will create a region for the current selection.

Typing 'L' will create a loop for the current selection.

Regions are now named after the start and end position of the region rather than order of creation.

And more...

You can now drag-scroll the Waveform Display by dragging on the ruler, drag speed is determined by how far the mouse is moved beyond the end of the window horizontally or vertically.

You can drag selection edges with the mouse.

A new *Levels* toolbar displays Value, Percent, dB, Peak, or RMS Power around the cursor. The scan window size used is determined in the *Status Preferences*.

When playing in Looped Mode, playback always follows selection and cursor changes.

Support for the InterVoice file format for telephony applications.

Envelope controls now multi-point editing as well as the ability to display the sound data in the envelope control.

A free disk space display has been added to the status bar. You can turn this on or off in the Preferences section.

We now use the Windows 95 Explorer common dialogs for open and save. We've also added the ability to auto play files in the *Open* dialog.

Many new preference options have been added to allow you to customize the way you work.

Help Menu

Contents

Takes you to the on-line help table of contents. **Shortcut:** F1

Search for Help On

Use this to find help on a particular topic with the on-line help.

Keyboard Shortcuts

Displays a list of shortcuts and Sound Forge is loaded with them.

Troubleshooting

Use this before calling the Tech Support line.

Tip of the Day

Displays a new Sound Forge tip.

The Sonic Foundry Home Page

Takes you to our little plot in cyberspace.

About Sound Forge

Use this to display pleasantly puzzling information about Sound Forge, such as the software license owner, copyright and system information, program version and serial number, and the cool Sound Forge logo.

Tip of the Day

This command will bring up the *Tip of the Day* pages in the help file. These are helpful and handy tips to increase your productivity with Sound Forge.

The Sonic Foundry Home Page

This is a direct link to our home page on the World Wide Web. Invoking this command automatically launches your default web browser and takes you to the Sonic Foundry Home Page (<http://www.sfoundry.com>). Here you will find Downloadable demos and updates, technical information, and news of what's new with the folks at Sonic Foundry.

You can also send us comments, suggestions, questions, sound files, gossip, whatever...via email from our web page (or direct to our development team at develop@sfoundry.com). We'd love to hear how Sound Forge is being used and your thoughts on how it can be improved.

Search for Help on

Use this to find on-line information on a particular topic.

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Help file last updated: 08/15/96

Sound Forge Main Screen Basics

[Video tutorial](#)

This section will show you around the Sound Forge screen and teach you the most basic operations. We will cover:

[Using the Mouse](#)

[The Main Screen](#)

[Toolbars](#)

[The Data Window](#)

[ToolTips](#)

[Getting On-line Help](#)

Using the Mouse

Although using the mouse in Sound Forge is not required, it will make your editing sessions easier. Once you become familiar with Sound Forge you will probably want to use some of the built in shortcuts provided by the mouse and the keyboard. The following list of terms will help you when reading the manual.

Pointing	Moving the mouse pointer over an item is called pointing.
Clicking	Pointing to an item and quickly pressing and releasing the left or right mouse button is called clicking. The manual will let you know whether you need to right-click or left-click on an item to execute a specific function. When there is no specification as to left or right you can assume we mean left-clicking. Right-clicking is often used to reach Shortcut Menus.
Double-Clicking	Double-clicking is the same as clicking except instead of pressing and releasing the mouse button once you do it twice in quick succession. Double-clicking always indicates clicking twice with the left mouse button.
Triple-clicking	Triple-clicking is the same as clicking except instead of pressing and releasing the mouse button once you do it three times in quick succession. Triple-clicking always indicates clicking three times with the left mouse button. In Sound Forge, a triple-click in the data window selects all data.
Toggle-clicking	Toggle-clicking involves clicking the right mouse button while holding the left mouse button to toggle through various options. This is a great shortcut for procedures such as drag and drop and changing the <i>Magnify Tool</i> mode of operation.
Shift Clicking	Shift-clicking is holding down the <i>Shift</i> key on the keyboard while clicking the mouse. Shift clicking is used mainly to skip dialogs so that you can quickly repeat operations.
Control Clicking	Control -clicking is holding down the <i>Control</i> key on the keyboard while clicking the mouse. Control-clicking is used to modify the operation of a normal click.
Dragging	Holding down the mouse button while you move the mouse pointer is known as dragging. Dragging is used to quickly move sections of data between separate windows and to move trackbars, scrollbars, faders.
Slow-dragging	Holding down both the right and left mouse buttons while moving trackbars, and faders increases the resolution of the movement. This is especially useful when making fractional adjustments to parameters in a dialog box.
Dropping	After dragging an item, releasing the mouse button on top of another area is known as dropping. Dragging and dropping is used to speed up operations like mixing or moving regions within the Playlist.

The Main Screen

When you start Sound Forge, you see the main screen, or work space, where you will do all your editing. When you first open Sound Forge, no Data Windows are open and you will need to either open an existing sound file or create a new one. There are a variety of ways to do this, all of which are explained later in the manual. The following list briefly describes each part of the screen.

Program Title Bar	Shows the program name and the name of the currently active data window (if the data window is maximized within the workspace).
Data Window	Each opened sound file has its own data window. Data windows can be arranged, resized, or minimized.
Menu Bar	Shows the menu headings for the available functions. When no data windows are open, the Process, Effects, and Tools menus are not listed because these contain functions which require an open data window.
Status Bar	On the left, help and processing information is displayed. The fields on the right show the playback sample rate, sample size, mono/stereo, total length of the active data window, and the total free storage space. These fields can be edited by double-clicking or right-clicking on them (except for the free storage space, of course). When no data windows are open, the fields are blank.
Sound Forge Workspace	This is the background area behind the data windows. You can drag sections here to create new data windows.

Toolbars

The first time you run Sound Forge, two toolbars also appear on the screen: the Standard and *Transport* toolbars. The toolbars contain buttons which are used to quickly execute commands.

Standard Provides quick access to many commonly used Sound Forge File and Edit menu options.

Transport Provides the audio transport buttons: *Record, Play All, Play Normal, Pause, Stop, Go to Start, Rewind, Forward, and Go to End.*

These toolbars, like any other toolbar in Sound Forge, can be dragged and dropped to anywhere on the screen. Toolbars are also resizable and removable. When a toolbar is dragged to any side of the main window, it docks, or attaches to the side. If you drag a toolbar away from a side, it becomes a floating toolbar. To hide a floating toolbar, just click on its *Close* button.

A list of available toolbars can be displayed by selecting the *Preferences* command under the Options menu and looking in the *Toolbars* folder (or *Toolbars* in the View menu). To show a toolbar, just check the box next to the toolbar you wish to use and then select *OK*. The number of open toolbars and their position on the screen is entirely up to you.

The Data Window

Data Windows are the windows which contain sound data files. These windows contain a number of sub windows and controls which you will use in editing and viewing your sound data. Many parts of the Data Window lead to Shortcut Menus, dialogs, and other operations.

The following list briefly describes each part and possible mouse operations:

Title Bar	Shows the file name or, if present, the title stored in the <i>Summary</i> information of .WAV files. Double-click to maximize and restore the window.
Level Ruler	Shows the amplitude of the waveform. Right-click to reach the <i>Level Ruler</i> shortcut menu. Drag to shift the view up or down when zoomed in vertically.
Time Ruler	Shows the current location in the data window as well as ruler tags. Right-click to reach the <i>Time Ruler</i> shortcut menu. Drag to scroll the data window.
Ruler Tags	Used to indicate the position of region end points, loop end points, and markers. Right-click to reach the <i>Ruler Tag</i> shortcut menu. Drag to change their locations. Double-click anywhere in the region to select.
Edit Tool Selector	Use this button to toggle between the <i>Edit</i> , <i>Magnify</i> , and <i>Pencil Tools</i> . Note: The <i>Pencil Tool</i> requires that the zoom ratio be between 1:1 and 1:16.
Playbar	On the left side of the playbar are the following audio transport buttons: <i>Go to Start</i> , <i>Go to End</i> , <i>Stop</i> , <i>Play Normal</i> , <i>Play Looped</i> , and <i>Play as Sample</i> . To the right are the playbar's selection status fields.
Selection Status Fields	Show the beginning, end, and length of a selection. If no selection has been made, only the cursor position (also known as the insertion point) is displayed. Double-click on the left-most field to reach the <i>Go To</i> dialog. Double-click on either of the other two fields to reach the <i>Set Selection</i> dialog. Right-click to reach the <i>Status Format</i> shortcut menu.
Waveform Display	Shows a graphical representation of the sound file. The horizontal axis represents time (marked in the <i>Time Ruler</i>) and the vertical axis represents amplitude (marked in the <i>Level Ruler</i>). Right-click anywhere on the waveform display to reach the <i>Waveform Display</i> shortcut menu.
Position Scroll Bar	Use this to scroll the sound file forward and backward in time to see parts of the file not currently visible in the waveform display.
Overview	Allows for quick navigation and playback of any part of the file. Overview also shows the fraction of the waveform being shown on the waveform display, as well as the selected region. Left-click to move the cursor. Double-click to center the cursor in the waveform display. Right-click to start playback or pause. Left-click and drag to activate the <i>Audio Event Locator</i> .
Time Zoom Resolution	(Also called <i>Zoom Ratio</i>) Specifies the number of samples of data represented by each point on the screen horizontally. This determines the length of time which is shown in the waveform display. With a small resolution value (1:1, 1:2, 1:4, ...), a shorter length of time is displayed.
Time Zoom In/Out	Use this to change the zoom resolution for the time axis. Between these buttons is a spinner that allows you to continuously change the time zoom resolution.
Level Zoom In/Out	Use this to change the zoom resolution for the level (vertical) axis. Between these buttons is a spinner that allows you to continuously change the level zoom resolution.
Maximize Width	Pressing this button stretches the width of the data window to fit within the

Sound Forge workspace.

OLE Drag Source

Drag from this section of the data window into another application such as Microsoft Word, to link the sound file to other documents.

ToolTips

If you keep the mouse over a toolbar button or status bar field for more than one second, you will notice a small box appear next to the mouse pointer containing some text. This text is a quick description of the function of the button or field. Using *ToolTips* is an easy way to learn your way around the Sound Forge screen. If you wish, you can disable this function by selecting *Toolbars* under the View menu and unchecking *Show ToolTips*.

When you press and hold a button in a toolbar or select a menu item, you will see a one line description of the command in the left side of the status bar. This gives you a little more descriptive information about the function of a button. If you release the mouse button outside of a toolbar or menu item, the toolbar command will not be performed.

Getting On-line Help

On-line help is available in a variety of ways. You can:

Go straight to the Help table of contents by selecting *Contents* under the Help menu, or pressing *F1*. Once there, you can search for your help topic and also get information on using the on-line help.

Press the *Help* button or the *F1* function key in a dialog. This will take you straight to the help information on the dialog.

Press the *F1* function key while selecting a menu item to get information on the command.

You will find that on-line help is extremely useful as a quick reference guide. Since many topics are cross linked, its easy to navigate through all of the help material available on a topic. Once you have become familiar with using Sound Forge, you'll probably never have to refer to this manual to find information. All you'll need is the on-line help.

Data Window Basics

With Sound Forge, you can have multiple sound data files open simultaneously on the screen. Each file has its own corresponding Data Window which shows you a graphical representation of the waveform and other information about the file. This section covers the following topics:

[Opening an Existing File](#)

[Playing a File](#)

[Playing a Section](#)

[Using the Transport vs. the Data Window Playbar](#)

[Creating a New Window](#)

[Active Windows vs. Inactive Windows](#)

[Copying Data to a New File](#)

[Saving a File](#)

Opening an Existing File

To open an existing sound file, do the following:

1. Select the *Open* command from the File menu. Sound Forge displays the *Open* dialog.
2. Use the *Open* dialog to select and open a file. You can preview .WAV files by pressing the *Play* button in the *Open* dialog. Pressing the *Play* button once more stops the playback. You can also have the files play when selected by checking the *Auto play* box.

You can use the *Files of type* list to select different types of data files. If Sound Forge recognizes the type of a file when it is highlighted in the files list, it will display its information in the bottom half of the dialog.

Open the file TUTOR1.WAV now. This file is located in the directory where you installed Sound Forge.

You will now see the data window containing TUTOR1.WAV. This file is a recording of someone saying "Wow, sound editing just gets easier and easier."

Notice how even though the file name is TUTOR1.WAV the name of the data window is Wow, sound editing... This is because we have stored a title for the file within the .WAV file itself. Sound Forge allows you to embed descriptive titles as well as copyright information and other text fields for any .WAV file.

Sound files in Sound Forge can have a title that is different from the file name. To edit the title, use the *Summary* page of the *Properties* command dialog (File menu). You can quickly get to the *Summary* dialog by pressing *Alt+Enter*.

Playing a File

Once you have opened a file, you can hear it by clicking on the *Play All* button on the *Transport* toolbar.



While the file is playing, a pointer will move along the data window showing the current position. You will also see the current position in the first selection status field in the playbar.

Play the TUTOR1 file now. You will hear the words Wow, sound editing just gets easier and easier.

You can play from any point in the window by simply moving the cursor. Move your mouse pointer to the silence section right after the "Wow" portion of the file. Left-click (don't drag, we don't want a selection yet), and you will see the flashing cursor in the silence area after the word "Wow." Click the *Play* button on the *Transport* toolbar and now you will only hear the "sound editing just gets easier and easier" part.

If you don't hear anything you may have made a small selection by accidentally dragging the mouse. To tell if you've made a selection just look at the three right hand status fields in the data window status bar. If only the first one is filled, then you don't have a selection. If they all have values then you need simply click in the data window again to clear the selection.

Playing a Section

You can also play portions of the sound data by making a selection on the waveform display with the mouse. To do this, click and drag the left mouse button starting at the section of silence prior to "Wow." Drag the mouse until you are in the portion after the "Wow." Notice that as you drag the data, the background appears in another color. Let the mouse button up and you will now have a highlighted section of data. Again click the *Play* button. You should now hear just the "Wow" portion of the file.

While making a selection, and after selecting, you will notice that the three selection status fields on the right side of the playbar have values. These values show you the start (*Selection Left*), end (*Selection Right*), and length (*Selection Length*) of the selection you have made. When no selection has been made, the cursor position (also called the insertion point) is shown.

Let's clear the selection and put the cursor back at the start of the file. To do this click on the *Go to Start* button in the *Transport* toolbar (the fourth button from the left).

Using the Transport versus the Data Window Playbar

There are a number of ways to play your sound files within Sound Forge. The most common method is to use one of the *Play* buttons located either on the *Transport* toolbar or the playbar in the data window. Let's take a look at both of them.



The Playbar buttons

On the left-side of the playbar at the bottom of every data window are six buttons. These buttons allow you to play the sound data in a variety of ways. The first two buttons allow you to set the cursor to the start or end of the current file. These are the *Go to Start* and *Go to End* buttons. The next button is the *Stop* button which you can use at any time to stop playback.

The last three buttons play the sound in three different ways and also set the default play mode at the same time. The first button is the *Play Normal* button. This button plays the currently selected section of data. If there is no selection, pressing the *Play Normal* button plays from the current cursor position to the end of the file.

The final button is the *Play as Sample* button. This button is used to play a sound file as if it was present in a sampler. This means the file will play from the start until it reaches the sustaining loop defined for the file and then will loop in the sustaining loop for the number of loops defined. If there are no loops defined the file plays straight through. This is known as a one shot sample.

Whenever you play a file from the playbar, you will notice a small line that appears underneath the play arrow in the corresponding *Play* button. This is the current play mode which will be used whenever you select the *Play* button in the *Transport* toolbar (or hit the *Spacebar*). *Play Normal* is the current play mode in the playbar graphic shown above.

To change the current play mode, press the corresponding playbar button or use *Control+Spacebar*.

The Transport Toolbar

The *Transport* toolbar has seven buttons.



The *Transport* toolbar buttons are described below from left to right:

Record	Brings up the <i>Record</i> dialog which allows you to record to either a new or existing window. Shortcut: <i>Control+R</i>
Play All	Plays the entire sound file from beginning to end. This allows you to hear the entire sound even if you have a selection or the cursor positioned somewhere other than the start of the data. Shortcut: <i>Shift+Spacebar</i>
Play	Plays the file using the current play mode. To set the play mode select one of the three play buttons on the playbar as described above. The last button used becomes the active play mode. Shortcut: <i>Spacebar</i>
Pause	Works like the <i>Stop</i> button in that it stops play of the current file. However when selecting <i>Pause</i> the cursor is placed at the current play position rather than back to where it was when play was selected. Shortcut: <i>Enter</i> during playback
Stop	Stops play and positions the cursor back to where it was prior to selecting play. Shortcut: <i>Spacebar</i> during playback

Go to Start

Places the cursor at the beginning of the sound file. **Shortcut:** *Control+Home*

Rewind

Shuttles the cursor backward in the sound file. **Shortcut:** *Control+Left Arrow*

Forward

Shuttles the cursor forward in the sound file. **Shortcut:** *Control+Right Arrow*

Go to End

Places the cursor at the end of the sound file. **Shortcut:** *Control+End*

Creating a New Window

To create a new Data Window do the following:

1. Select the *New* command from the *File* menu. Sound Forge displays the *New Window* dialog.
2. Select the new data format in the dialog.

Since we are already working with the TUTOR1 file let's create a new window that has the same data format as TUTOR1.

In the *New Window* dialog set the *Sample size* to 16-bit, the *Channels* to *Mono*, and the *Sample rate* to 44,100. When you click the *OK* button a new window will appear titled *Sound2*. This is an empty data window into which we are going to place data from TUTOR1.

Active Windows vs. Inactive Windows

When you have multiple windows on the screen, only one window is considered the active window. This is the window on which you are currently working. Any operations you perform will only affect this window.

To make a window the active window click anywhere on the window with the left mouse button. The window's title bar will change to the color you have defined as the active window color in *Control Panel* in Windows.

Copying Data to a New File

Make Wow, sound editing... the active window by clicking on its title bar. If you don't still have the word Wow selected, select it again as we did above. From the Edit menu, select the *Copy* command. This will copy the sound data for Wow onto the clipboard.

Now make Sound2 the active window by clicking on its title bar (if the title bar is covered by another window, you can always use the Window menu to activate a data window). Select the *Paste* command from the Edit menu and the Wow data will appear in the Sound2 window. If you press the *Play* button, you can hear how our new file sounds with just the word Wow.

There are faster ways to copy data from one window to another, or from an existing window to a new window, but we will cover those in a later section.

Saving a File

To save a data window, you first need to make it the active window. Make sure the Sound2 window is active.

1. Select the *Save* command from the File menu. Since the Sound2 window is a new file, Sound Forge displays the *Save As* dialog. If the file had been opened or previously saved by Sound Forge the file would be saved immediately.
2. Type a file name into the *File name* field in the *Save As* dialog and select the *OK* button. Let's save the new sound file, Sound2, as MYFIRST Sound Forge will add the file extension, .WAV, automatically.

The *Save As* dialog also allows you to change file types, data format, and set summary information fields. This is covered in a later section.

Simple Editing and Navigation

This section will show you the basic editing operations of Sound Forge.

[Common Edit Operations](#)

[Making a selection](#)

[Copy, Paste, Cut, Undo, Redo](#)

[Cropping](#)

[Simple Mixing](#)

[Status Formats](#)

[Magnification and Zooming](#)

Common Edit Operations

The edit operations used most often include cut, copy, paste, delete, mix, and trim/crop. Most of these make use of the Clipboard, which is a temporary storage area which can also be used to move data from one window to another. The following list provides a brief description of each operation:

Cut	Deletes a selected portion of data and copies it on to the Clipboard.
Copy	Copies a selected portion of data on to the Clipboard.
Clear	Deletes a selected portion of data but doesn't copy it on to the Clipboard.
Trim/Crop	Deletes all data in a window except the selected section.
Paste	Inserts the contents of the Clipboard into a Data Window at the current cursor position.
Mix	Mixes the contents of the Clipboard with the current data in a window starting at the current cursor position.
Crossfade	Crossfades the contents of the clipboard with the current data in a window starting at the current cursor position.

To show how these operations are used, we will once again be using the TUTOR1.WAV file. If it is not currently open please open the file now as described in the previous sections.

Making a Selection

If TUTOR1 is not the active window, activate it by clicking on its title bar. To make a selection, you must first make sure that you are using the *Edit Tool*. To get to the *Edit Tool*, look under the Edit menu for the *Tool* item (it should already be checked) and select *Edit*. Next, select the word "Wow" with the mouse by clicking and dragging like we did in the previous section. You can verify you have the right section of data by pressing the *Play* button to hear it.

Copy, Paste, Cut, Undo, and Redo

Copying Data on to the Clipboard

Once you have a selection, you can use the *Copy* command from the Edit menu. This will copy the selected data onto the clipboard. You will see no change since the *Copy* command does not change the data, it only copies it to the clipboard.

Pasting Data from the Clipboard

Now move the cursor to the beginning of the file by selecting the *Go to Start* button on the playbar. To insert the contents of the clipboard into the file, use the *Paste* command from the Edit menu. You should now see the data for the word "Wow" appear at the beginning of the window. Press the *Play* button just to make sure. You should hear Wow Wow Sound editing just gets easier and easier.

You have just made your first edit in Sound Forge! There are much quicker ways to do what we have just done with Sound Forge, but we'll get to that in the Advanced Editing section.

Pasting Data to Another Window

Data on the clipboard remains there until it is replaced by another operation which places data onto the clipboard. Therefore, you can continue pasting the data anywhere you want.

To demonstrate this, create a new window as described in the previous section by selecting the *New* option from the File menu. Now, select the *Paste* command from the Edit menu once more. You should now have a new data window with the data for the word Wow. You can also do this by selecting the *Paste to New* function in the *Paste Special* item of the Edit menu. This creates a new window and fills it with the clipboard contents in one easy step.

Cutting Data

To cut data, first you need to select a section of data you want to cut. Select one of the Wow words (you should have two if you have been following the examples) from TUTOR1.

Now select the *Cut* command from the Edit menu. This will remove the selected data and place it on to the clipboard.

Undoing an Edit Operation

After any edit operation you can cancel it by selecting the *Undo* command from the Edit menu.

Let's undo the cut we just made. Select *Undo Cut* from the Edit menu and you will see the original two Wow words in the TUTOR1 window.

Redoing an Edit Operation

If you change your mind again and decide you really liked an edit you undid, simply select *Redo* from the Edit menu. Doing this re-performs the edit operation you last undid. In this case selecting *Redo* would re-cut the second Wow from the TUTOR1 window.

Let's redo the cut. Select *Redo Cut* from the Edit menu and you will see the second Wow has been cut from the file again.

<p>The Undo/Redo History window (View menu) also allows you to restore your sound file to previous states by undoing or redoing operations. For more information, see the Reference section.</p>
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Trimming edges (crop)

Trim (also called *Crop*) allows you to single out a section of data and cut everything else out of the window except that section. This is a handy feature since you can keep using the *Play* button to hear selections until you have just the right amount and then get rid of everything else with the *Trim/Crop* command in the Edit menu.

By now you should be getting used to selecting data on the screen, so select the "Wow, sound editing just gets easier" section in the TUTOR1 window, but don't select the second easier. Remember you can use the *Play* button to hear how the selection sounds at any time.

Once you have the selection, select the *Trim/Crop* item found in the Edit menu. After cropping you will have only "Wow, sound editing just gets easier" left in the window.

At this point let's close the Wow, sound editing... window. Either click on the close box of the TUTOR1 window, or select the *Close* command from the File menu. You will be asked whether you want to save the changes you have made to TUTOR1. Select the *No* button since we don't want to keep the changes we made to TUTOR1.

Also close any other windows you may have open, like MYFIRST.WAV.

Mixing

Mixing is a powerful and useful edit operation which you will use often. Mixing allows you to combine two sounds together into one window so you can create complex sound effects.

This time we are going to open two files, TUTOR1.WAV and TUTOR2.WAV. Open them now as we've shown you in the previous sections.

TUTOR2 is a file which contains the sound of a snare drum and crash cymbal. We are going to mix this sound with TUTOR1; the "Wow, sound editing..." window.

Before we begin mixing, let's make sure the *Status Format* is set to *Time* to make finding the mixing points a little easier. To do this select the item *Time* from *Status Format* under the Options menu. You will need to do this for both data windows since Sound Forge keeps track of the format type for each individual window.

To make the windows easier to view during the mix operations you may want to maximize Sound Forge by pushing the *Maximize Window* button in the upper right-hand corner and selecting the *Tile Vertically* item from the Window menu. This will arrange the windows vertically fully utilizing the Sound Forge workspace and make things easier to see.

If you activate each of the TUTOR windows you will notice that the length of TUTOR1 is 5.0 seconds long and TUTOR2 is 3.0 seconds long. For this example, we want the drum hit to occur just before the "Wow." We could just copy the drum hit sound on to the clipboard and then paste it before the "Wow", but this would increase the length of TUTOR1 to 8.0 seconds. So instead we'll use the *Mix* command.

Select all the data in *Drum Hit* by making it the active window and then double-clicking in the waveform display with your left mouse button (you could also use the *Select All* command under the Edit menu). Now copy the data onto the clipboard by selecting the *Copy* command from the Edit menu.

Make TUTOR1 the active window and then select the *Go to Start* button on the playbar. This will put the cursor at the beginning of TUTOR1. Select the *Mix* command from the *Paste Special* option in the Edit menu.

The *Mix* dialog now appears. Keep both levels at 0 dB and select *OK*.

You will see that the drum hit sound has been mixed into the TUTOR1 window and the length of TUTOR1 is still 5.0 seconds. Press the *Play* button to hear the results.

Select the *Undo Mix* command from the Edit menu to put TUTOR1 back to its original state.

Now let's mix the drum hit sound closer to the Wow portion of TUTOR1. The Wow occurs at about 0.8 seconds into TUTOR1, so move your cursor in the TUTOR1 window to approximately 0.8 seconds. You do this by clicking with the left mouse button in the data window of TUTOR1 and watching the cursor position status field on the status bar (the left most field). You don't have to be exact. Once you've positioned the cursor, select the *Mix* command again and then hit *Play* for the result. Notice how the drum hit sound and the Wow sound overlap each other.

Status Formats

By right-clicking on the *Time Ruler* and selection status fields on the playbar, files can be set to different formats. You can coordinate sound files with other events, or edit to a timing base you feel most comfortable using.

Lengths and positions can be displayed in a variety of formats including *Samples, Time, Seconds, Frames, Measures and Beats, and SMPTE*.

Selecting a Status Format

To select a format choose the *Status Format* item from the Options menu. This shows the ten different formats available. Choosing one of these options sets the status format for the current data window.

The available formats are:

Samples	Number of samples
Time	Hours, minutes, seconds and milliseconds
Seconds	Seconds and fractions there of
Time & Frames	Hours, minutes, seconds and frames as defined by the <i>Edit Frame Rate</i> (Special Menu)command
Absolute Frames	Frames and fractions there of
Measures & Beats	Measures, beats and 1/4's of a beat
SMPTE Non-Drop	SMPTE at 30 or 29.97 frames per second (fps) non-drop
SMPTE Drop	SMPTE at 30 fps with drop frames
SMPTE EBU	SMPTE at 25 fps
SMPTE Film Sync	SMPTE at 24 fps

Using the current file, TUTOR1, let's take a look at how the status formats affect values in the status display fields. First, select the *Samples* format from the *Status Format* menu under the Options menu.

Now select all of the data in the TUTOR1 window. To do this choose the *Select All* option from the Edit menu. This will select all data in the window. You can also select all of the data in a window by triple-clicking the left mouse button anywhere in the waveform display of the data window.

In the selection status fields on the playbar (at the lower right hand side of the data window) you should see the values of 0, 240,006, and 240,007. This means that the first selected sample (*Selection Left*) is sample 0, the last selected sample (*Selection Right*) is 240,006, and the total number of samples in the selection (*Selection Length*) is 240,007.

Now select the *Time* option from *Status Format* under the Options menu. You will see that all these values change to values specified in hours, minutes, and seconds rather than samples. You can see that a sound containing 240,007 samples with a sample rate of 44,100 Hz will play for 5.000 seconds. You can experiment with each of the status formats to see how each format is displayed.

Configuring Frames and Measures & Beats

When setting the status format to *Frames* or *Measures & Beats*, there is additional information you can provide to Sound Forge to customize how these values are displayed. The *Edit Frame Rate* dialog in the Special menu allows you to change the frames per second. In the *Edit Tempo* dialog, also in the Special menu, you can specify the *Beats per minute* and *Beats per measure* values used to calculate measures and beats. The default values for *Frames* and *Beats* are set on the *Status* page in the *Preferences* folder (Options menu).

Magnification and Zooming

When sound data is drawn on the screen, it is necessary to represent many samples of data for each horizontal point on the screen. There are almost always many more samples in a sound file than there are horizontal points (pixels) on the screen. Depending on the editing operations you need to use, you may want to view the entire file at once, or you may want to look only at a small portion in greater detail. This is where the magnification, or zoom ratio, comes in.

[Time Ruler Zooming](#)

[Level Ruler Zooming](#)

[Using Zoom Selection and Zoom Out](#)

[Magnify Tool](#)

Time Ruler Zooming

In the bottom right corner of the data window, just above the data window playbar, is the current time magnification ratio. The *Zoom Ratio* defines how many samples of sound data are squeezed into each horizontal point on the screen. This setting is shown as a value of 1:XX , where XX is the number of samples represented by each horizontal point on the screen.

For example, if the setting is 1:1 then each point on the screen is one sample. With this setting, a very short length of time is shown on the waveform display. You might not be able to see the cursor or selection because you are seeing it at such close range. If the setting is 1:1,024, then there are 1,024 samples represented by each point on the screen and a longer length of time can be seen. By dragging the *Time Ruler* right or left, you can easily scroll the data window. The further you drag in either direction, the faster the window will scroll.

If you right-click on the waveform display you can quickly select various <i>Zoom</i> commands from the <i>Data Window</i> shortcut menu.

When Sound Forge opens a file it sets the horizontal magnification to the value specified by the *Normal zoom ratio* set on the *Display* page in the *Preferences* folder, unless the file is too small.

To increase or decrease the *Zoom Ratio*, use the *Zoom In/Out* spinner control found on each data window next to the ratio. Clicking on either of the magnifying glasses increases or decreases the zoom ratio by single steps. If you drag the small rectangular button between the two magnifying glasses, you can spin the zoom ratio by large amounts.

Level Ruler Zooming

In the bottom left corner of the data window, just above the *Go To Start* button, is the spinner control for zooming in and out vertically in the data window. Zooming in along the level axis allows for more precise editing at the lower levels (where many edits start and end).

The labels for this axis can be either in decibels or in percent and are switched by right-clicking on the ruler. When zoomed in, only the lower level samples can be seen; the peaks of the waveform will have moved beyond the scope of the data window. To view these peaks, drag the *Level Ruler* up or down.

To optimize both the time and level scaling of a selection, double-click on the *Level Ruler*.
Double-click again to zoom out to the normal level.

Using Zoom Selection and Zoom Out

If you would like to quickly magnify a section of a data window horizontally, the easiest way is to use the *Zoom Time|Selection* command found in the View menu or in the *Data Window* shortcut menu. To use the *Zoom Time|Selection* command you must first select an area which you want to magnify. After selecting *Zoom Time|Selection*, Sound Forge will calculate the minimum zoom ratio which will allow the full selection to fit on the screen and then center the selection within the waveform display.

This will not affect the zoom ratio of the <i>Level Axis</i> . Only the <i>Zoom Selection</i> command in the <i>Level Ruler</i> shortcut menu will zoom the selection both vertically and horizontally.

To display all data in a window at one time, use the *Zoom Time|Out Full* command. This command sets the *Zoom Ratio* to the lowest value needed to display all of the data at once in the waveform display.

If you wish to set the zoom factor to its maximum magnification (1:1), use the *Zoom Time|In Full* command also found in the View menu or the *Data Window* shortcut menu (*Zoom In Full*). You will then be able to see the actual waveform oscillations which correspond to physical sound waves when played through your speakers.

Magnify Tool

Another way to zoom in to a particular section of the sound file is to use the *Magnify Tool*. You can select this tool from the Edit menu (*Tool|Magnify*) or by clicking on the *Edit Tool* selector (upper left-hand corner of data window) until the magnifying glass appears.

When the *Magnify Tool* is enabled, instead of making a selection, the mouse cursor will become a magnifying glass and a dotted square will appear. The zoom will take place within this dotted square. The three different modes of operation for the *Magnify Tool* can be selected by toggle-clicking (See Using the Mouse above). These allow you to time zoom, level zoom or both simultaneously. You must then return to the *Edit Tool* mode to do edit operations.

You can also enter this *Magnify* mode temporarily by holding down the *Control* key when making a selection. If you drag the mouse while still holding the *Control* key, you can select a region which will then be magnified to a best fit ratio within the waveform display.

Advanced Editing and Navigation

This section will show you how to use the advanced editing and navigation operations of Sound Forge:

[Making a selection using the Set Selection dialog](#)

[Extending the selection points with the mouse](#)

[Selecting with the keyboard](#)

[The Overview](#)

[Using the Go To dialog](#)

[Using Views](#)

[Drag And Drop Operations](#)

[Advanced Mixing options](#)

[Editing Stereo Files](#)

Making a selection using the Set Selection dialog

If you need to select data at specific points, the *Set Selection* dialog allows you to type in selection points or choose a selection from a list of regions. To reach the *Set Selection* dialog, choose the *Selection|Set* command from the Edit menu.

In the *Set Selection* dialog, you can modify the *Start*, *End*, *Length*, and *Channel* of the selection. Pre-determined regions can also be chosen from the *Selection* drop-down list.

Also, in this dialog you will find *Snap Zero* and *Snap Time* buttons. Pressing one of these buttons will force the current start and end selection points to their nearest zero-crossings or time increments, accordingly. A more thorough explanation of these functions can be found in the Reference chapter under *Selection* in the Edit menu section.

Many commonly used functions in Sound Forge can be performed in a number of different ways. For example, you can reach the *Set Selection* dialog by double-clicking on the playbar field to the left of *OLE*, or you can pick the *Selection* item from the *Waveform Display* shortcut menu. Sound Forge has been designed to allow you to choose how you want to work. It doesn't force you to work in a certain way. To avoid confusion in our demonstrations, we won't always tell you about every method of accomplishing the same task. Refer to the Sound Forge Reference chapter and the Appendix section on shortcut keys for more information on each feature.

Extending a Selection the Mouse

After selecting a section of sound data, you will sometimes find that the start or end points are not exactly where you want them to be. You could just re-select the data, but it is often difficult to get the start or end points just right.

Sound Forge allows you to update the selection by grabbing a selection edge and moving it. Once you have established a selection, place the mouse over the selection start or end point. You will notice that the cursor changes to a bi-directional arrow. Once this arrow appears, you can press the left mouse button and move the selection to a new position. The new position will be updated once you let up on the mouse button.

Snapping a Selection to Zero-crossings

If after making a selection, the end points jump somewhere else automatically, it's probably because you have *Auto Snap to Zero* on. You may want to enable *Auto Snap to Zero* (under the Options menu) so that any drag selections start and end points will always reside on a zero-crossing.

To make a current selection snap to zero-crossings, select *Edit|Selection|Snap to Zero* (or press Z on the keyboard).

If you'd like *Auto Snap to Zero* disabled only when zoomed in (1:1 or 2:1), select the *Disable auto-snapping below 1:4 zoom ratios* option in the *Editing* page in the *Preferences* folder.

Selecting with the keyboard

The keyboard selection controls allow you to quickly select data or update a selection accurately. For example, say you want to extend the selection end by a small amount. First make sure that the cursor is at the end of the selection. You can move the cursor from one end of the selection to another by pressing the *Home* (moves to the start) and *End* (moves to the end) keys. Use the C key to center the cursor in the waveform display. You can then use the *Shift+Right* arrow and *Shift+Left* arrow combinations to extend the selection end point by small increments.

The following list shows the available keyboard commands for selecting data.

Press	To select from cursor to...
<i>Shift+Right</i> arrow	the next screen pixel.
<i>Shift+Left</i> arrow	the previous screen pixel.
<i>Shift+End</i>	the last sample visible in the waveform display.
<i>Shift+Home</i>	the first sample visible in the waveform display.

Press	To select ...
<i>Shift+Page Up</i>	10% of the current view past the cursor position.
<i>Shift+Page Down</i>	10% of the current view previous to the cursor position.
<i>Shift+Control+Right</i> arrow	2% of the current view past the cursor position (see below).
<i>Shift+Control+Left</i> Arrow	2% of the current view previous to the cursor position (see below).
<i>Shift+Numpad +</i>	the next sample.
<i>Shift+Numpad -</i>	the previous sample.
<i>Shift+Control+Numpad +</i>	10 samples past the current position.
<i>Shift+Control+Numpad -</i>	10 samples previous to the current position.

In Windows 95, <i>Shift Numpad +</i> will not work unless the <i>Numlock</i> indicator on the keyboard is lit.
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Also, if there are regions, markers, or loops present in the data window:

Press	To select from cursor to...
<i>Shift+Control+Right</i> arrow	the next region start/end or marker.
<i>Shift+Control+Left</i> arrow	the last region start/end or marker.

As you can see from the above lists, Sound Forge has extensive keyboard support for selecting data. You will find that almost any operation has an equivalent keyboard shortcut which advanced users find invaluable. For a complete listing see Appendix A.

Let's take a quick look at how to use some of these keys. Open the file TUTOR1.WAV as we've done in previous sections. First, make sure that you're using the *Edit Tool* by selecting it in *Tool* under the Edit menu. Now make a selection that encompasses the word *Wow* with a generous amount of space on each

side of the word. Pressing the *Play* button you should hear a little bit of silence, *Wow*, and a little more silence.

Choose the *Zoom Selection* command from either *Zoom Level* or *Zoom Time* in the View menu. This will fit the selection on the screen with the best possible resolution. Now let's adjust the right side of the selection.

Grab the end of the selection with the mouse and drag to the right. Adjust the selection to be close to the end of the word *Wow* and let up on the mouse button. Now let's do the same thing with the keyboard. Hold down the *Shift* key and press the *right* arrow key. The selection will extend to the right by one screen pixel. If you hold down the *Shift* key and press the *left* arrow key, the selection will decrease by one pixel. One screen pixel is equal to the number of samples shown in the *Zoom Ratio* field. You can move the end of the selection in this manor until you've got the selection just right.

Now let's adjust the start of the selection. Since the cursor is at the end of the selection, any keys we press for selecting will adjust the end. So we need to put the cursor at the front of the selection. To do this, press the *Home* key. You will see the cursor jump to the start point of the selection. Now, use the selection keys to adjust the start of the selection.

A selection can also be updated on a per sample basis rather than a pixel basis. If you look in the above list you will see that the *Shift + Numpad + (plus)* and *- (minus)* keys are used for these operations. This allows you to get the exact selection you need without having to change the *Zoom Ratio*.

The Overview

While making selections and navigating through a sound file, you probably noticed the overview (the thin bar directly underneath the title bar) change. The overview represents the length of the entire sound file, as if you were zoomed out all the way. From the overview, you can determine what section of the entire sound file is being displayed, the selection made, and the cursor location.

Getting the Whole Picture

Zoom out all the way on the TUTOR1 window (*Zoom Out Full* in the *Data Window* shortcut menu) and make a selection over the word *Wow*. Notice that the entire overview has brackets above and below it, since the entire sound file is being displayed. Also notice how the selection and the cursor are displayed on the overview.

Now press the *Zoom In* button a few times (the large magnifying glass in the lower right-hand corner of the data window) and notice how the brackets become smaller. This corresponds to the smaller fraction of the entire sound file which you can see when you zoom in on the waveform display. The selection size, however, remains the same and does not move. Even when you can't see them on the waveform display, you always know where the cursor and the selection are in the sound file by referring to the overview.

You can also change the cursor position by left-clicking anywhere on the overview. As always, when you change the cursor, you lose your selection. Remember you can use *Toggle Selection* if you want to restore it.

Fast Navigation and Playback

When you left-click outside the bracketed region in the overview, you will not be able to see the cursor on the waveform display. However, double-clicking anywhere on the overview will move the cursor and center the waveform display to the selected position in the sound file.

You can also play back the sound file starting from the current cursor position by right-clicking anywhere on the overview. Right-clicking again pauses playback. Left-clicking in the overview also moves the cursor position. Also note that left-clicking on the overview during playback moves the cursor to the point where you clicked and continues playback at that point. These navigation tools make it very easy to find sections in large files.

As an example, say you wanted to move the cursor to the beginning of a phrase in a speech. You can right-click on the overview bar to begin playback and then left-click at different positions within the overview until you find the right start point. Once you find the start point, you can press the *Stop* button to stop playback. The cursor should now be positioned at the last spot on which you left-clicked in the overview; select *Center Cursor* from the Special menu and you are ready to edit.

Audio Event Locator

Holding down the left mouse button and dragging it in the overview bar will begin playback of small looped portions of audio around the cursor position. While this is not technically a scrub function, it serves a similar purpose i.e., it allows the user to hear brief segments of audio material in order to quickly scan or locate events within the sound file. The length of the *Audio Event Locator* loop segment can be adjusted in the *Preferences* dialog.

Using the Go To dialog

To quickly move the cursor to a specific point and center it in the waveform display, use the *Go To* command. You can reach the *Go To* dialog in a number of different ways:

Use the *Go To* item in the Edit menu

Right-click on the waveform display and select *Cursor|Go To* from the shortcut menu

Double-click on the left-most selection status field

Press Control+G

The *Go To* dialog is much like the *Set Selection* dialog described below. You can type in the location where you want to go or use one of the predetermined locations from the list. To create your own location marker, press the M key at any cursor position. You can also drop a marker at any time during playback and recording by pressing M. Refer to the Regions List and Playlist sections for more information on using markers.

Using Views

If you wish to save the selection points you've set, the zoom ratio, or the position of the waveform display, you can do so with *Views*. Sound Forge has the ability to store up to eight views for each data window. Each view stores the selection, cursor position, magnification, and position scroll bar placement. A saved view can later be instantly restored.

You can store and recall up to eight different views (1-8) in this manner. There is also a *Views* toolbar you can use (look in the *Toolbars* folder of the *Preferences* dialog) to set and restore views. When using the *Views* toolbar, all stored views are indicated by an underscore in each *View* button.

Follow these steps that demonstrate how to use views:

1. Open the file TUTOR1.WAV and select Wow.
2. To store the current view, press *Control+1*.
3. Now, zoom in and make a different selection anywhere you wish. Store this new view by pressing *Control+2*.
4. To return to the first view you stored, press *1*. Notice that the selection and zoom factor are restored.
5. To go to the second view you created, press *2*.

Drag And Drop Operations

You can perform a mix, paste, or crossfade by dragging a selection from one data window and dropping it on another.

To drag and drop:

1. Open two sound files and select *Tile Vertically* from the Window menu just to make them easier to work with. One will be called the source window and the other, the destination window.
2. Press and hold the left mouse button in the selected area of the source data window. While holding the mouse button down, move the cursor until it changes to an arrow dragging a little box.
3. Drag the cursor onto the destination window. You will see a shaded block that represents the range of the source material. You will also see that the little box accompanying the cursor now contains an M(Mix), C (Crossfade), or P (Paste). Position the block at the place in the destination window where you would like the mix, paste, or crossfade to occur.
4. Let go of the left mouse button. This will bring up the *Mix* dialog. To perform a crossfade instead of a mix, hold the *Control* key down while letting go of the mouse button. Holding down the *Alt* key when dropping performs a paste.

Tip: An easier way to select whether you will perform a mix, paste, or crossfade is to simply drag the block to the destination window and then while holding the left mouse button, click the right mouse button to toggle between the *Mix*, *Paste*, or *Crossfade* functions. You will see the block change according to the method; a mix will look like a solid block, a crossfade will look like an X (or perhaps, a butterfly), and a paste will be represented by a segmented vertical line rather than a block.

As we did in the previous section on mixing, open the two files, TUTOR1.WAV and TUTOR2.WAV. For this example, TUTOR1.WAV will be the destination window and TUTOR2.WAV the source window.

To make the windows easier to view while doing the mix operation you may want to select the *Tile Vertically* option from the Window menu.

Highlight all the data in TUTOR2 by making it the active window and triple-clicking in the waveform display. Now drag the selected section by pressing and holding the left mouse button in the data window of the TUTOR2 window. Notice that the cursor changes to a pointer with a small box when you begin dragging. Drag the block to the beginning TUTOR1 data window and let go of the mouse button to drop the section.

When you let go of the mouse in the TUTOR1 window, you will see a dialog which shows the TUTOR2.WAV window as the source and TUTOR1.WAV as the destination. Leave the levels at 0 dB, and select *OK*. You will see that again the drum hit sound has been mixed into the TUTOR1 window. Press the *Play* button to hear the results.

Drag and Drop Crossfading

Undo the last operation. Now select all of TUTOR2 and drag it toward the end of TUTOR1. Press and hold the *Control* key, you will see the crossfade block and by moving the block back and forth horizontally across the end of TUTOR1, you can adjust the crossfade time and see the crossfade expand and contract over the end of the destination file. Once you have determined where you would like the crossfade to begin, let go of the mouse button. This takes you to the *Crossfade* dialog. Select the *Normal crossfade* preset from the preset list at the top of the dialog and press *OK*. You should hear the end of TUTOR1 fade out as the beginning of TUTOR2 fades in.

Drag and Drop Pasting

Again, undo the last operation. Select all of TUTOR2 and drag it over to TUTOR1. Hold the *Alt* key down while you position the paste point at the beginning of TUTOR1. Now, let go of the mouse button. If everything was done correctly, you should have pasted the data from TUTOR2 at the insert position in TUTOR1.

Creating a New Window with Drag and Drop

A very useful feature of drag and drop is the ability to quickly create a new window from a selection. Make a selection in a data window and drop it on an empty area of the Sound Forge workspace. A new data window is automatically created containing the data from the selection.

Snap to Event in Drag and Drop Operations

Another useful feature of drag and drop is the ability to snap to markers, regions, time increments and other events in the destination window. All of the drag and drop operations mentioned above will snap (or gravitate) to points in the destination file depending on what events have been established in the file.

The following list shows possible *Snap to* points in the destination file.

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 <i>Time Ruler</i>
Video Frames	Start of block snaps to the start of video frames appearing in the video strip

Advanced Mixing options

In the *Mix* dialog, there are several options you can use to perform complex mixes. For example, when mixing vocals over music, you can have the music fade out and then fade back in when the vocal part is mixed. This is sometimes referred to as ducking. Follow these steps to perform voice-over ducking:

1. Open the file TUTOR1.WAV and select the word Wow.
2. Drag the Wow selection to an open area in the Sound Forge workspace to create a new data window.
3. In the *Summary* dialog within *Properties* (File menu), name this sound file Wow in the *Title* field and press *OK*.
4. Now, open the file TUTMUSIC.WAV. This file contains a short music clip to which we will mix the Wow sound.
5. Place the cursor in TUTMUSIC at about two seconds from the start of the file.
6. Select all of Wow and drag and drop it to the TUTMUSIC window.
7. In the *Mix* dialog, select the *Slow duck* preset. Notice that the *Destination Volume* is set to -6 dB and the times in the *Pre/Post-fade destination edges* are set to 0.5 seconds. This means that before and after the mix, the music will be faded to -6 over 0.5 seconds. Press *OK*.
8. Listen to the file, then undo the operation.
9. Once again, drag Wow to TUTMUSIC and select the *Slow duck* preset. This time, set the *Destination Volume* to -20 dB and press *OK*. The music will now fade out even lower. Both the source and destination volumes can be used to get the right mix of voice and music. The *Pre/Post-fade destination edges* control how fast the destination sound is faded out and back in.

Editing Stereo Files

When editing stereo files you have two channels of data on which to work. The upper channel is the left channel and the lower channel is the right channel. We will refer to them in both ways left (upper), and right (lower).

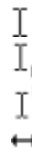
Selecting Data in Stereo Files

When selecting data in stereo files, Sound Forge allows you to select either the left channel, right channel or both channels for playing, editing, and effects processing.

When editing a stereo file, the waveform display showing the two channels is split into three logical sections for selection with the mouse. The upper quarter of the waveform display is the left channel hit section, the lower quarter is the right channel hit section, and the middle half selects both channels. When selecting data with the mouse, the cursor location determines what channel(s) will be selected.

Open the file TUTOR1.WAV and convert it to stereo by editing the *Format* under *Properties* in the File menu or by right-clicking on the *Channels* playbar field and selecting *Stereo*. Set the *Channels* to *Stereo* and press *OK*. Select *Both Channels* for the destination in the *Mono to Stereo* dialog. You should now have a stereo version of TUTOR1.WAV.

Move the mouse pointer near the top of the left channel and select the word "Wow." Notice the change to the left channel selection cursor, and that only the left channel of the data becomes highlighted. Now do the same thing but in the middle half of the window near the top of the right channel and the bottom of the left channel. This time you should see both channels being selected. Do this one more time near the bottom of the waveform display and you should see only the right channel being selected.



Cursor when selecting both channels of a stereo file or when in a mono file.

Cursor when selecting the left channel of a stereo file.

Cursor when selecting the right channel of a stereo file.

Cursor when extending a selection.

Toggling Channel Selections

Once you have made a selection in a stereo file you can switch between channel selections by pressing the *Tab* key. The *Tab* key will cycle between selections of *Left Channel*, *Right Channel*, and *Both Channels*. You can also set the channel selection by using the *Channel* drop-down list in the *Set Selection* dialog.

Previewing Channels

Selecting a single channel allows you to hear a preview of a single channel in the stereo file. For example double-click (triple-click if it contains regions or markers) in the TUTOR1 window to select all the data (or use *Select All* from the Edit menu). Press the *Play* button and listen to the clip. Next press the *Tab* key to toggle the channel selection into a single channel and press the *Play* button again. Do this one more time to hear the other channel.

Single Channel Editing

Stereo data files are tied together by the nature of their format. In other words, they always play together. This means that there are some edit operations, such as *Cut* and *Paste*, which you can't use on a single channel. It would leave one channel shorter or longer than the other. This is usually not a problem in real world editing situations. To shift a single channel in time by small amounts, you can use the *Delay/Echo* function.

You can copy a selection from a single channel to the clipboard by selecting the data in either the left or right channel and using the *Copy* command. This will place a mono clip on the clipboard. You can then paste the mono clip to a mono file, both channels of a stereo file, or you can mix it into a single channel or both channels of a stereo file. When mixing mono clipboard data to a stereo file you will be asked with a

dialog whether you wish to mix to a single channel or both channels.

Working with .AVI Files

Sound Forge allows you to open and save *Microsoft Audio and Video Interleave (.AVI)* files. This allows you to edit the audio tracks of a video with single frame accuracy. This section covers:

[Opening an .AVI File](#)

[Handling Multiple Streams](#)

[Video Strip](#)

[Video Preview](#)

[Attaching a Video to a Sound File](#)

[Saving an .AVI file](#)

[.AVI Video Compression](#)

[Compression Algorithms](#)

[Compression Settings](#)

Opening an .AVI File

To open an .AVI file, choose *Open* from the File menu and select *Video for Windows (*.avi)* from the *Files of type* list. Then, double-click on the file you wish to open. If the .AVI file being opened in Sound Forge contains exactly one video stream and one audio stream, it will open these streams automatically. However, if no audio is present an audio stream will be created containing only silence for the entire length of the video.

Handling Multiple Streams

.AVI files can contain more than one video or audio stream. A stream is not the same as a track in a multi-track video or audio editor. Multiple streams are most commonly used to contain different versions of a video or audio track. For example, you could have an .AVI file with one video stream and multiple audio streams in different languages. An .AVI player, such as Microsoft's *Media Player*, can detect which language version of Windows your computer is using and automatically play the correct audio stream for that language.

If more than one stream exists you have the option of choosing which one will be opened in Sound Forge. This is done in the *Video Stream* dialog. In this dialog, you must select a video and audio stream to open by using the *Stream Selectors* (black diamonds) to the left of each stream. Once you've selected one of each, press *OK*.

Video Strip

When you open an .AVI file, you will notice the video strip on top of the waveform display. When you're zoomed out, all of the frames in an .AVI will not be displayed. The left edge of the displayed frames are aligned with the audio in time. If you'd like to see the frame numbers on the frames, right-click on the video strip and select *Number Frames*.

To resize the video strip, drag its lower edge up or down.

To enable frame animation in the video strip, select *Animate* frames from the *Video Strip* shortcut menu. When this option is enabled, moving the cursor will cause the nearest frame to be displayed above it. During playback, you will see the active frame window play the video as the cursor moves past it.

Video Preview

The *Video Preview* window is opened by selecting *Video Preview* from the View menu. The frame closest to the cursor (during editing or playback) will always be displayed in this window.

At the top of the window, the original frame size and current display size (in parenthesis) are displayed, along with the frame rate and current frame number. An asterisk (*) by a frame number indicates that the frame is a key frame (see the section on compression below).

Attaching a Video to a Sound File

To attach a video to an opened sound file, select *Properties* from the File Menu and go to the *Video* folder. Pressing the *Attach* button will take you to the *Attach Video* dialog, where you can select an .AVI file to attach a video from. Press *OK* after selecting an .AVI with a video stream.

The *Video Properties* will now display the audio and video streams in the .AVI. An .AVI file can contain multiple streams, but in most cases you'll have one audio stream and one video stream.

The square checkbox to the left of each stream indicates which streams will be stored when you go to save your file. The black diamond next to it is the *Stream Selector*, which indicates which streams are currently being used in Sound Forge.

Press *OK* again and you should now see the video strip above the waveform display. Once you're done editing, you must save the file as an .AVI if you want to store the video.

Saving an .AVI file

To save an opened file in the .AVI file format, follow these steps:

1. In the *Save* or *Save As* dialog, select *Video for Windows (.avi)*. After naming the file, etc., press *Save*.
2. The *Video Save Options* dialog appears next. In here, you can select which streams to store in the .AVI file by checking the square checkbox to the left of each stream. In this dialog, it is also possible to edit the titles of the streams.
3. Pressing *OK* will take you to the *Compression Options* dialog. Here, you can select a compression scheme for each audio and video stream. To change the options for a stream, select it and press the *Options* button. By checking the *Interleave every _ Frames* option, you can specify how frequently the audio chunks are interwoven with the video.
4. If you pressed the *Options* button, yet another dialog, *Video Compression*, let's you select from different compression algorithms, and even go to another dialog to set specific compression configuration parameters. For more information on compression, see the next section. Selecting *No Recompression* saves the stream in its original format, which is the Sound Forge default.
5. Once you're done setting compression options, hitting *OK* several times will complete the process. If the video is long and/or the compression scheme selected is slow, saving the file might take a while. However, if you didn't change any video compression formats, the save will be much faster, since no recompression takes place.

.AVI Video Compression

If you thought audio files chewed up your hard-disk space quickly, wait until you start using video! CD quality audio takes up about 10 MB of hard-drive space per minute:

(60 seconds x 44,100 samples per second x 2 tracks for stereo x 2 bytes per 16-bit sample). On the other hand, a typical video for multimedia use will easily contain about 200 MB per minute, and this for a small window of video:

(60 seconds x 15 frames per second x 320 x 240 x 3 bytes for 24 bit pixels).

Using uncompressed video as a final distribution format is out of the question for most practical purposes. Video compression is a necessary evil.

Compression Algorithms

Many different compression algorithms exist and many more will continue to be created in the next several years. With all algorithms, trade-offs will have to be made between video quality, size reduction, and compression/decompression processing time. Compressing by large amounts quickly will generally produce visual artifacts such as jumpy or grainy video.

When saving an .AVI in Sound Forge, you can select from all of the installed compressors installed in your system. Included with Windows are Cinepak by Radius (cvid), Intel Indeo (IV32), and Microsoft Video 1 (CRAM). These can be selected from the Video Compression dialog (see section above).

Compression Settings

Many compressors use key frames during compression and decompression. A key frame is a frame in the video stream which is not inter-frame compressed. The following compressed frames are derived from the closest key frame. Having fewer key frames will make the video size smaller. However, it will take a lot more time to display the frames, specially when jumping around the end video.

When editing, .AVI files in Sound Forge, the display speed will suffer greatly if there is only one key frame in the entire file. In such cases, Sound Forge needs to scan all preceding data when drawing every single frame.

Also in the *Video Compression* dialog, you can specify the *Data Rate* of the video, which relates directly to the final size of the file. Different compressors do different things with this value, since its only an expected value. In most cases you should leave this unchecked. If you have very limited playback data rates, such as in a single-spin CD-ROM, you might then want to set this value. However, note that the *Compression Quality* control setting will also affect the output size.

Use the *Preview* button to get an idea of the video's output quality and compression amount.

Editing Audio in .AVI Files

When doing any operations on the soundtrack of an .AVI file, the video stream is always left untouched. For example, doing a cut, paste, time compression, ,etc. on a sound file will not change the duration of the video stream.

In Sound Forge, the video stream and audio stream will always start together, and you will see the video frames only where audio exists. Therefore, if you want to view the entire video stream, you should use *Insert Silence* to add enough audio to run the entire length of the video. You can then later mix in any needed sounds into the file.

Using the Playlist with .AVI Files

Once you've created and arranged regions in the *Playlist* (see a previous section), playing from the *Playlist* will keep the original synchronization between the audio and video. In other words, if you have a *Playlist* that jumps all over the file, both the audio and video will jump around.

In a sense, you can use the *Playlist* to perform basic video edits. For example, to remove unnecessary material from the beginning and end of the video, simply make a region containing only the video that you need in the *Playlist*. Playback from the *Playlist* will play only the selected regions in the specified order.

Creating a New .AVI File from the Playlist

After arranging the Playlist entries, you often want to create a new file containing the final arrangement. To do so, select *Playlist|Convert to New* from the Special menu. This will create a new file, which when played back will contain the exact sequence of audio and video that the current *Playlist* specifies.

When creating a new file, each region's start and end point must line up with a frame boundary. Region start points are moved backwards and end points moved forward, with silence being used to fill the frame. This can cause the output file to be slightly longer than the *Playlist* playback time.

To ensure that *Convert to New* works exactly like playing from the Playlist, make sure that all the region endpoints are frame aligned. This can be done by enabling *Auto Snap to Time* (Options menu) and using the *Time & Frames* or *Absolute Frames* status format before creating regions.

Editing Sound Formats

This section will show you how to deal with the many different sound formats.

[Changing sound formats](#)

[Mono to Stereo and Stereo to Mono Conversions](#)

[Converting 16-bit samples to 8-bit samples](#)

[Converting files](#)

[Embedding Summary Information in WAV files](#)

Changing Sound Formats

When you open a file or create a new window, the format for the window is shown on the first three status fields of the main status bar at the bottom of the program window. The format parameters consist of the *Sample Rate*, *Sample Size*, and number of *Channels* (Mono or Stereo). The last status fields display the *Total Length* of the sound data, and amount of *Free Storage* available.

Changing the sample rate will make your sound play faster or slower and with a higher or lower pitch. If you want the file to sound the same at a different sample rate, use the *Resample* function under the Process menu.

When you right-click on any of the three format fields, a shortcut menu will pop up at the current mouse location allowing you to change that parameter. You can also change all three parameters at one time by selecting *Properties|Format* under the File menu. The *Properties* dialog can also be invoked by double-clicking on any of the format fields, or by right-clicking on the waveform display of the data window and selecting *Properties* in the shortcut menu.

Mono to Stereo and Stereo to Mono Conversions

For demonstration purposes, open the file TUTOR1.WAV located in the directory where Sound Forge 4.0 was installed (in most cases this will be C:\Forge40). To convert the file from mono to stereo, right-click on the status field which has the word *Mono* and select *Stereo* from the shortcut menu.

When you do conversions from mono to stereo, you will be presented with a dialog which asks you where you want to put the data. Your choices are as follows:

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.

When performing stereo to mono conversions, the following options are available:

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.

For now let's put the data in the left channel so we can get a feel for how the function works. Click the radio button next to *Left Channel* and select the *OK* button.

You will now have data in the upper half of the data window (the left channel) and silence in the right channel. Press the *Play* button and you will hear "Wow, sound editing just gets easier and easier" in only the left channel.

If your card supports only mono data you will still be able to play stereo files if you set your playback device to Sound Mapper in the *Wave* section of *Preferences* in the *Options* menu.

For now let's go back to a mono sound file, but we'll do it in a different manner. Select the option *Properties* in the shortcut menu that appears when you right-click on the waveform display. In the *Properties* dialog select the *Format* tab. This page allows you to change all the parameters at once. Set the radio button for *Channels* to *Mono* and select the *OK* button. Select the *Left Channel* for the source channel in the *Stereo to Mono* dialog and we're back where we started.

The above steps were used for demonstration purposes. Simply using *Undo* would have been much more efficient.

Channel Converter

The *Channel Converter* (Process menu) is also used to convert between mono and stereo formats. It allows independent level settings for each channel and can be used to inter-mix the left and right channels of a stereo file to create interesting pan effects. For simple conversions, use the included presets. See Channel Converter later in this manual in the Process menu section of the Reference chapter for more detailed information.

Converting 16-bit samples to 8-bit samples

To save storage space, 16-bit sound files are often converted to 8-bit files. However, when you represent a sound file with only 8-bits, an audible distortion referred to as quantization error becomes very prominent. To minimize quantization error, there are several things you can do to the 16-bit sound file before converting it to 8-bit. You can:

Apply a *Noise Gate* (Effects menu) to completely mute out the silent parts in a sound file. Often, a low level signal in a 16-bit sound file will become noise after the 16- to 8-bit conversion, so it's best to have complete silence between the sound parts.

Apply dynamics compression (select *Dynamics|Multi-Band* in the Effects menu) to the sound file. A small amount of compression (2:1 or less the presets are usually sufficient) will lower the dynamic range of a sound, making it easier to represent using 8-bit samples.

Normalize (Process menu) the sound to 0 dB (peak). This ensures that the entire dynamic range available in 8-bit samples is used and lowers the signal to noise ratio.

You can apply both dynamic compression and normalization at once by using the *Normalize* function with the *Average RMS power* option. Set *Apply dynamic compression* in the *If clipping occurs* drop-down list.

Once you have performed the above operations, you should use *Convert to 8-Bit* in the Process menu to do the 16 to 8-bit conversion. Sound Forge provides three different 16 to 8-bit conversion options; *Truncate*, *Round*, and *Dither*. *Truncate* simply removes the lower 8 bits and they are lost forever. *Round* is similar to *Truncate*, only some attempt is made to represent the highest of the low bits by rounding up if possible. *Dithering* is used to mask the quantization noise with less obtrusive noise. In the *Convert to 8-Bit* dialog, set the *Dither* bit depth to a low value (like 0.1) and increase the value slowly until you reach an acceptable sound.

Remember, an 8-bit sound file will always sound noisier than a 16-bit sound file, so whenever possible, stick with 16-bit. When trying to save space, it is possible to get better results from lowering the sample rate (see *Resample* under the Process menu covered in the Reference chapter) instead of using 8-bit samples.

Converting Files

When producing audio files you may find that you need to provide the file to a client in a variety of formats. This could be .WAV files with data formats such as 8-bit mono, 16-bit stereo, or even ADPCM. You may need the files converted to Raw data, .VOC, or some other PC format. You may even need to convert to a different platform like Macintosh .AIF files or Amiga .SVX.

We have seen in the previous sections that you can quickly convert between mono/stereo, 8/16-bit, and change the sample rate in Sound Forge. You can then use the *Save As* command to save the file to a new name, or you can save in a new format within the *Save As* dialog.

If you don't have TUTOR1.WAV open, open it up now. Select the *Save As* command from the File menu and the dialog for saving will be displayed.

When the *Save As* dialog is displayed, there are three controls that are used for file conversion; the *Save as type*, *Format*, and *Attributes*. The *Save as type* is the list of all available output file types. The *Format* field sets the format of the data which is saved in the file. The *Attributes* field controls sample size, sample rate, and the number of channels.

When you bring up the *Save As* dialog, the *Save as type* will be set to the type from which the file was opened. The default for new files is *Microsoft Wave (.WAV)*. You can change to the type of file you want to save by selecting the new type from the *Save as type* list. Notice that the extension of the file name will be changed to reflect the type of file you are saving.

The *Format* drop-down list shows the format of the data which will be saved in the file. Usually this will be *PCM* which is the standard for most file types. *PCM (Pulse Code Modulation)* is a standard format for normal uncompressed audio. If you are saving a file to a file type which supports other formats, they are displayed in the drop-down list. Other formats are typically used when saving audio data in a compressed form.

The *Attributes* drop-down list shows the sample size, channels, and sample rate which will be stored when saving this file. Examples include: 11,025 Hz, 8-bit Mono or 44,100 Hz 16-bit stereo. You can quickly change the attributes of the data by selecting one of the options available in the drop-down list. When you change from a mono file to a stereo file using the *Attributes* field, that data is copied to both channels. If you change a stereo file to a mono file, the data is mixed to one channel.

When you change the *Save as type* you may notice that the *Format* and *Attributes* fields will also change. This is because not all file types support the same types of sound data. For example the Dialogic .VOX format only supports 16-bit mono Dialogic ADPCM files. This means you can't store stereo 8-bit data in this file. Don't worry, in most cases Sound Forge automatically converts any file you save to an acceptable format.

Embedding Summary Information in WAV Files

Microsoft .WAV and .AVI files allow you to store over 30 text fields including: *Creation Date*, *Copyright*, *Keywords*, and a variety of other informational text data. Sound Forge supports viewing and editing any of these fields.

Summary Information

When editing .WAV and .AVI files, the *Summary* tab is available in the *Properties* section of the File menu. This allows you to view and/or change basic summary information currently stored in the file.

Extended Summary Information

The *Extended Summary* dialog consists of a list of available fields, each of which may or may not have attached text. The *Fields* list shows the abbreviation of the field type, a short description, and if the field is currently empty, the word (*Empty*) appears after the short description. Below the list of fields is the *Contents* which shows the contents of that field. At the bottom of the dialog is a longer description of the current field.

To the left of each field in the list is a check box which is used to enable or disable fields of this type when saving .WAV and .AVI files. The field is enabled or disabled by clicking on the check box. If a field is empty, i.e. has no text associated, enabling the check box has no effect on a saved file. Field information is only saved if text information exists for that field.

The *Save Summary Information in .WAV/.AVI files* check box in the *Save As* dialog is used to quickly enable/disable saving of all summary fields to a file during a *Save As* operation. This allows you to quickly strip all summary information from a file.

For a description of the fields available refer to the *Properties* within the File menu section in the Reference chapter later in the manual.

Default

If the *Default* button is selected, the text in the summary fields is saved with the defaults fields that are automatically filled when creating a new .WAV file. The *Creation date* field (ICRD) is always filled with the current date for new files. Saving a custom default setup is handy for saving copyright and engineer information for new files created at your site.

Open the file TUTOR1.WAV and select the *Summary* tab from *Properties* in the File menu. This will show you the *Summary* dialog. Press the *Extended* button and notice that the default for saving summary information is to have all fields enabled. The fields which have already been filled in for TUTOR1.WAV are *Comments*, *Copyright*, *Creation Date*, *Engineer*, *Keywords*, *Name/Title*, *Product*, *Subject*, *Software*, and *Source*. Scroll to and select each field. You will see the embedded text for each of these fields.

Additional Embedded Information

The Microsoft .WAV and .AVI file formats allows non-text data to be embedded in the sound file. If you open and edit a file which has additional information created by other software, Sound Forge will keep track of this data and place it back in the file when you save to the original format.

If you wish to add or remove additional data (such as embedded bitmaps, metafiles, etc.), you can use the *Save As* item from the File menu. When you save data using the *Save As* option, Sound Forge will ask you if you wish to copy the additional non-audio information to the new file. At this point you can answer either *Yes* to place the additional information in the new file or *No* to save only audio data and any additional data types you have selected.

Sound Forge will only prompt you to save if such data exists.

Recording

[Video tutorial](#)

Sound Forge has a variety of sound recording modes and options each of which come in handy for different situations. These include recording new data, recording over existing data, commonly known as Punch In mode, and Remote Recording.

[Recording basics](#)

[Recording modes](#)

[Record Meters](#)

[Sync While Recording](#)

[Remote recording](#)

Recording basics

[Video tutorial](#)

To start a record session you can either select *Transport* then *Record* from the Special menu or press the *Record* button on the *Transport* toolbar. The *Record* button is the first button on the *Transport* toolbar with the red circle on it.

If you experience problems recording, refer to the Troubleshooting chapter for information on common recording problems.

After pressing the *Record* button or selecting *Transport|Record* from the Special menu, you will be presented with the *Record* dialog. Notice that the window into which you will be recording has its title displayed in the dialog title.

Warning: The *Record* and *Record Remote* dialogs are always destructive and contain no *Undo* capabilities.

To prevent accidentally recording over sound data, record into a new or scratch data window and paste the takes you want to keep into the desired sound data window. For extra safety Sound Forge automatically defaults to record in a new window when the current sound file is opened in direct edit mode.

In the upper left corner of the dialog are the *Recording attributes*. These are the record sample rate, sample size, and number of channels which will be used when recording. These attributes are applied to the data window into which you will be recording. If you want to change these attributes, exit the *Record* dialog and change them in the data window, record to a new window, or record to another window.

Recording to a New Window

If you want to record to a new window rather than the currently selected record window, select the *New* button found at the upper right of the *Record* dialog. This brings up the *New Window* dialog where you can specify the *Sample rate*, *Sample size*, and *Channels* for the new data window. These attributes will be applied while recording.

Selecting an Alternate Record Window

If you wish to record to a window other than the one currently displayed in the *Record* dialog title, you can do so by selecting the *Window* button. Pick the window you want to record to from the drop-down list in the *Record Window* dialog and select *OK*. The title of the window you select will now appear in the *Record* dialog title.

Recording Modes

[Video tutorial](#)

Sound Forge has four different modes of recording. These are Automatic Retake, Multiple Takes with Regions, Multiple Takes (no Regions) and Punch In. Each mode is described below.

Automatic Retake

The Automatic Retake mode is the easiest method of recording. Recording starts at the position shown in the Start field when you select the Record button and continues until you select the Stop button. Any data which is currently after the position in the Start field will be replaced. When recording is stopped, the start position is reset to the beginning of the take allowing an immediate review and retake if desired.

Automatic Retake is the default mode when recording into an empty Data Window or when you select the record button with no data selected in the current Data Window.

Multiple Takes with Regions

The Multiple Takes with Regions mode allows multiple takes to be recorded and each take with automatically have a Region defined in the Regions List. Recording starts at the position shown in the Start field when you select the Record button and continues until you select the Stop button. Any data which is currently after the position in the Start field will be replaced. When recording is stopped, the start position remains at the end of the next take allowing the take to be recorded immediately.

Multiple Takes (no Regions)

The Multiple Takes (no Regions) mode allows multiple takes to be recorded, but no Regions are defined in the Regions List. Recording starts at the position shown in the Start field when you select the Record button and continues until you select the Stop button. Any data which is currently after the position in the Start field will be replaced. When recording is stopped, the start position remains at the end of the take allowing the next take to be recorded immediately.

Punch In

Punch In mode is used when you want to record over a region of data in an existing Data Window. Recording starts at the position shown in the Start field when you select the Record button and continues until you select the Stop button, or the length of the data recorded is equal to the length in the Length field. This makes it easy to record over a section of audio without effecting the rest of the file. You can use the Play button to hear the selected Punch In region at any time.

You may adjust the Punch In region by changing the values in the Start, End, or Length edit fields. You may also adjust the format of these fields to a variety of different display status formats by selecting a format from the Input format drop down list box.

Punch In mode is the default mode when you select the Record option while you have a region of data selected in the current Data Window.

Using Pre/Post-Roll with Punch In Mode

At the bottom of the Record dialog are two edit fields which contain the Pre-Roll and Post-Roll times. These can be used when listening to a region in Punch In mode. These times define the amount of audio you will hear prior to (Pre-Roll), and after (Post-Roll), the selected region when using the Play button. This allows you to hear the transitions between the Punch In region and sound before and after the region. If you wish to use the Pre/Post-Roll option you must check the Review check box at the bottom of the dialog. To disable Pre/Post-Roll uncheck this box.

Using the Prepare Button

The Prepare button is used when you need Sound Forge to begin recording as soon as possible after selecting the Record button. The Prepare button opens the wave device and loads all recording buffers in order to minimize the time between selecting the Record button and sound actually beginning to be recorded.

The Prepare button is optional. It is not necessary to select this button prior to recording, however it does allow for more accurate takes in the Punch In mode.

Recording Status

While you are recording, the amount of time recorded will increase and the Time left on drive will decrease. Make sure and keep an eye on your Time left on drive if your available record time is limited. It's never fun running out of recording time!

Available Recording Time

Near the bottom of the dialog you will see the *Time recorded* and *Time left on drive* fields. These two boxes show how much time you have recorded and how much time is available on your hard drive for additional recording. If your *Time left on drive* field is displaying a limited amount of available time you may want to free up some space on your hard drive or pick an alternate drive where Sound Forge stores its temporary files. You can get more information on temporary file usage by referring to the Reference chapter on *Temporary Storage*.

Finishing Recording

When you have finished recording select the Close button to exit the Record dialog and return to normal editing mode.

Previewing Recorded Sounds

After recording your material, you can listen to what you have recorded by selecting the Play button. You can also listen to the section over which you plan to record in Punch In mode. To stop playing select the Stop button at any time.

Record Meters

Like the Sound Forge *Play Meters*, the *Record Meters* can be scaled to view differing dynamic ranges by right-clicking on the meters and selecting the appropriate range from the shortcut menu. For most recording situations it is probably best to select -42 -to 0 dB, as this is the most practical range to view a good record signal. To view very low levels, select -90 to 0 dB. This is a good way to measure the amount of noise you have in your system due to noisy sound sources and/or poor equipment (such as an inexpensive sound card).

Also in this shortcut menu, you can set the meters to *Hold Peaks* and *Hold Valleys*. It is recommended that *Hold Peaks* be checked, as this is a good way to view the peak levels while setting the record level (see below). To reset the current peak, single-click on the meter's text output (just above the meters).

For Windows 95 and NT 4.0 keyboard shortcut lovers, the *Play Meter* shortcut menu can be reached by the keyboard shortcut *Alt+Spacebar*.

Checking Record Levels

Sound Forge allows you to check the level of your input source before recording begins. To view your levels engage the *Monitor* check box. The meters will light up in relation to the level of the incoming signal. For best results, the level should be somewhere in the yellow range with an occasional red. Once your levels are checked you can immediately begin recording by selecting the *Record* button. If you do not see the meters light up, you may have your mixer levels or input source set incorrectly. You can refer to the Troubleshooting chapter for more information on these problems.

Adjusting Levels Using the Peak

The peak values displayed above the level meters are useful for maximizing your input level without clipping. When recording you generally want your input signal to be as hot as possible without clipping. By this we mean you want your input levels to be as high as possible without exceeding the range of values which can be stored digitally when recording. When you clip, the peaks of your waveform become clipped off resulting in distortion. The peak values show you (in decibels) the highest peak that you have reached since hitting the *Monitor* button.

To adjust your levels, select the *Monitor* check box so that Sound Forge begins to listen to your recording device. This is just like recording except that Sound Forge doesn't store any of the data it receives. Apply an input signal by speaking into your microphone, playing your CD, or whatever it is you're trying to record. If the peak value stays at a low value, increase the levels of sound you are supplying so that the peak value is somewhere in the -6 dB range. If the peak reaches 0 dB then you have clipped and will see the word *Clip* above each meter. Once you lower your input levels, right-click on the meters and select *Reset Clip* to clear the current peak value. Sound Forge always keeps the maximum peak displayed above the meters or via *Hold Peaks*.

Once you have adjusted your levels you can immediately begin recording by selecting the *Record* button or end monitoring the levels by un-checking *Monitor* below the meters.

It is particularly important to record sounds with the hottest levels possible when you plan to later convert 16-bit data to 8-bit. This assures that you will use the greatest dynamic range possible in an 8-bit file (which uses fewer values to represent the waveform).

Sync While Recording

The Record Synchronization dialog is reached by pressing the Sync button. Use this dialog to set sync options for recording in conjunction with other devices or applications that respond to MIDI/SMPTE time code.

When *MTC/SMPTE Output Synchronization* is enabled, Sound Forge generates SMPTE/MTC to the output device selected in the *MIDI/Sync Preferences*. When you start recording, SMPTE output begins at the *Start* value. If *Pre-roll* is selected, SMPTE output will begin before recording by the specified amount. For example, if the *Start* value is 00:00:30:00 and the *Pre-roll* is 00:00:10:00, SMPTE time code will be generated starting at 00:00:20:00. After 10 seconds (at 00:00:30:00) have elapsed, recording will commence.

When *MTC/SMPTE Input Synchronization* is enabled, recording begins (is triggered) when the *Start* time is received from the current *MIDI/Sync Input* device (see the *MIDI/Sync* page in the *Preferences* folder). Recording will then stop when the end time is reached.

The *Prepare* button must be activated for record to be triggered.

Output Synchronization Start	Start time which will be generated when record begins
Output Synchronization Pre-Roll	Pre-roll time from the start time at which SMPTE generation will begin
Input Synchronization Start	Time at which recording will begin
Input Synchronization End	Time at which recording will end

Check to make sure that the input and output devices are specified correctly in *MIDI/Sync* under *Preferences* (Options menu). The SMPTE format here is determined by *Status Format* in the Options menu.

When recording tracks in from a tape deck that can perform a chase lock, the *Synchronization Pre-roll* can be set to compensate for the time it takes the tape deck to rewind and begin chasing prior to the synchronization start time.

Remote Recording

[Video tutorial](#)

By selecting the *Remote* button you enter into *Remote Recording* mode. *Remote Recording* mode hides the main Sound Forge window and puts up the small *Record Remote* dialog which always stays as the top-most window.

The *Remote* dialog is a fully functional but condensed version of the regular *Record* dialog. It's particularly useful when using applications that control your input sources, sound levels, CD audio, or MIDI sequencing. To return to normal record mode from the *Remote* dialog select the *Back* button.

You can access all of the same *Record* dialog features from the *Remote Record* dialog (right-click on the title bar).

Applying Sound Processing Functions

The Effects, Process, and Tools menu contains a large number of functions which are used to apply an effect on the sound data, display information, or even synthesize new sounds. Each function is documented in detail in the Reference section. In this section, we will run through a few of the functions to help you get a feel for using them.

[Applying sound processing effects](#)

[Applying an Effect to a Section of the Sound File](#)

[Canceling a Function in Progress](#)

[Applying Effects to Stereo Files](#)

[Getting Help on a function](#)

[Using Controls](#)

[Presets](#)

[Processing shortcuts](#)

[Changing the selection affected by the function](#)

[Previewing an Operation](#)

Applying Simple Processes and Effects

To apply an effect, first select a section of data on which you wish to operate. To select the entire file, double-click (triple-click if regions or markers are present) on the waveform display.

If you don't have a selection when you perform an effect which requires one, Sound Forge will apply the effect to the entire file.

To show how effects are used we will once again be using the TUTOR1.WAV file. If it's not currently open, please open the file now.

Select the *Reverse* item from the Process menu. The *Reverse* operation reverses the data selection, making it sound like it's playing backwards. Once the operation has finished, press the *Play* button to hear how TUTOR1 sounds when played backwards.

Applying an Effect to a Section of the Sound File

Select the *Undo Reverse* item from the Edit menu to put TUTOR1 back to its original state. Now select only the word "Wow" in the window and again select the *Reverse* item in the Process menu. Play the file again and notice how only the word "Wow" is played backwards. After you have listened to the effect, undo the reverse again so we can try another effect.

Again, let's select only the word "Wow" in the TUTOR1 window. This time after selecting the data, select the *Volume* item in the Process menu. Use the scale factor fader to set the scaling to -6 dB (50%). Clicking on the fader and using the up/down arrows will give you more precision. Now select *OK*. If you press *Play* you will notice that the volume is now at 1/2 of its original volume. You can undo the last operation by selecting *Undo* from the Edit menu.

The previous two examples show you how easy it is to apply effects to files. Sound Forge has a large number of different processes and effects, ranging from simple volume changes to complex multi-tap reverb/delays. To learn about a particular effect, please refer to the Effects, Process, or Tools menus which are covered in the Sound Forge Reference chapter.

Canceling a Function in Progress

While applying an effect, the progress meter at the lower left-hand side of the main screen shows what percentage of a selection has been processed while running a function. You can select the *Cancel* button to stop processing the file. When you cancel an operation in progress, the affected data remains in the sound file. Undoing the operation will return the file to its original state.

Applying Effects to Stereo Files

In a stereo file, only the selected region in the channel which you've selected is processed. Most functions can be applied to the left, right, or both channels. The only functions which cannot be applied to separate left and right channels are functions which affect the length of the data, since each track in a stereo file must be of the same length. These include *Insert Silence*, *Resample*, *Time Compress/Expand*, *Gapper/Snipper*, *Pitch Bend*, and *Pitch Shift (without preserving duration)*.

If you need to run a process like *Insert Silence* in only one channel, you should divide the stereo file into two mono files first, insert the silence, and then join them back together into a stereo file. Separating stereo files into two mono files is a good idea if you are going to be changing the duration of each track by cutting or inserting space to synchronize different events.

Applying a Function to a Single Channel in a Stereo File

Previously, we reversed TUTOR1 so that we could hear how the data sounded when played backwards. Now let's try this on a single channel of a stereo file. First convert TUTOR1 to stereo by right-clicking on the status field which has the word *Mono* and select *Stereo* from the shortcut menu as we did in a previous section. Make sure to select the *Both Channels* option in the dialog so we have data in both channels of the new stereo file.

Select the entire left channel by triple-clicking with the mouse in the upper 1/4 of the TUTOR1 waveform display. If you select both channels, just use the *Tab* key to toggle the selection to only one channel. Now select the *Reverse* item from the Process menu. You will notice that only the left channel data is reversed. Press the *Go to Start* button on the playbar to clear the selection and place the cursor at the start of the file. Press the *Play* button and you will hear "Wow, Sound editing is easy" backwards in the left channel and forward in the right channel. If you only hear one channel then you are in single channel preview mode. To hear both channels use the *Tab* key to toggle the cursor or selection to encompass both channels and again press the *Play* button.

<p>If you want more information on how to use a function, press the <i>F1</i> function key while selecting the function title in the menus. Also, you can press the <i>F1</i> function key or the <i>Help</i> button in all function dialog.</p>
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Getting Help on a function

If you want more information on how to use a function, press the F1 function key while selecting the function title in the menus. Also, you can press the F1 function key or the Help button while in the function dialog of those functions with dialogs.

Using Controls

Before going any further, let's become familiar with all of Sound Forge's different controls used to enter function parameters. To do this, open a new file and select the *Synthesis|FM* item from the Tools menu. Don't worry about understanding what all of the parameters mean. Let's just play with the knobs. Here's a list of controls used in Sound Forge:

[Vertical Fader and Horizontal Trackbars](#)

[Edit Box Spinner Control](#)

[Drop-down List](#)

[Push Button](#)

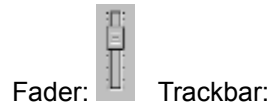
[Radio Button](#)

[Check Box](#)

[Envelope Graph](#)

Vertical Fader and Horizontal Trackbars

In the *FM Synthesis* dialog, the *Amplitude* is controlled by a fader and the *Feedback* is controlled by a trackbar.



There is nothing complicated about these controls. To change the parameter values, just left-click and hold on top of the trackbar thumb and drag left and right or up and down.

You can fine tune a control's value by holding down both the left and right mouse buttons at the same time (or holding down *Control* on the keyboard)

There are many keyboard shortcuts when using faders and trackbars. 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.

If you double-click on a fader or trackbar thumb, it returns to its reset value (usually 0%, 50%, or 100%). Left-clicking on the hash marks in a fader also changes the value by very small increments.

Edit Box Spinner Control



In the *FM Synthesis* dialog the *Total output waveform length* and *Frequency* parameters are modified with edit box spinner controls. The edit box is the box containing text, while the spinner is the small control to the right. With this type of control, you have several options for changing a parameter value. You can:

Type in the number by left-clicking on the edit box and then typing in the value.

Increment the value by clicking on the two small up/down buttons.

Use the spinner to change the value. This is done by left-clicking on the center button (between the two up/down buttons) and dragging the mouse up or down. Again, holding down both the left and right mouse buttons at the same time will cause the values to increment in finer steps. You can also use the *Up/Down* arrows and *Page Up/Page Down* keys to alter the value.

Adjust the spinner while holding down both mouse buttons to see the finer increment resolution (or hold down the *Control* key).

Drop-down List



To select an item from a drop-down list, click on the drop-down list and select the item. If you have to scroll through a large list, click on the scroll buttons or use the arrow keys. The *FM Synthesis* dialog uses a drop-down list to change presets, select waveforms, or change operators.

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.

Push Button



To use a push button (get ready for this...) left-click on it or press the *Spacebar* while it is selected. In the *FM Synthesis* dialog, the *OK*, *Cancel*, *Help*, *Save As*, *Delete*, *Selection*, *Preview*, and *Reset* controls are push buttons.

Radio Button



Radio buttons always come in groups of two or more. They are used like the radio station selector in a car radio; turn one on and the rest turn off. For example, the *Current operator* control in the *FM Synthesis* dialog is a group of radio buttons.

Check Box



A check box, as its name implies, is a square box you can check or leave blank. They are used to turn a feature on or off. *Create Undo* in the *FM Synthesis* dialog is a check box.

Envelope Graph

Envelope graphs are used to draw the shape of a frequency or amplitude envelope that will be applied. The horizontal axis represents time, with the left-most point specifying the start of the selection and the right-most point specifying the end of the selection. The vertical axis represents either amplitude or frequency.

- To create a new knob on the graph, left-click anywhere on the line connecting the envelope points and drag the knob in any direction, sort of like pulling a rubber band.
- You can delete a knob by double-clicking or right-clicking within it.
- To reset the graph (delete all knobs) click on the *Reset* button.

Moving Multiple Envelope Points

It's possible to select and move more than one envelope point at once. Just try:

1. Clicking on an unused section of the *Envelope Graph*.
2. While holding the mouse button down, drag the mouse. A dotted-line selection square will appear.
3. Move the mouse to surround the needed points with the selection square.
4. Release the mouse button. The selected points will now be colored red.
5. To move all of the selected points, drag any one of the points.

You can also select all points by pressing *Control+A* when the *Envelope Graph* is active.

Displaying the Waveform on an Envelope Graph

Some envelope graphs (such as in the *Fade|Graphic* or the *Pitch|Bend* dialogs) allow you to view the waveform of the selected data region on the graph. If the selection is small, the waveform is automatically displayed. Otherwise, you must press the *Show wave* check box. If you have a stereo file, you can choose between displaying the *Left channel only*, *Right channel only*, or *Mix channels* in the drop-down list.

Presets

Most processing functions allow you to store the current settings for later use. There are also system presets for each effect which can be used without having to learn how to use the knobs and tweakers. These presets are a good place to start when you're creating your own effects.

Open the file TUTOR1.WAV again. Under the Effects menu, select *Noise Gate*. The *Noise Gate* function mutes any sounds below a threshold level. Using the drop-down list labeled *Name*, scroll down the presets list. Notice that the all of the controls change automatically. Now select one of the presets and select *OK*.

If you didn't have a selection when you ran the effect, the entire file was processed.

Creating and Storing Your Own Preset

Now open the file TUTOR2.WAV. Select a large portion of the drum hit and then select the item *Pitch Bend* from the Effects menu.

The *Pitch Bend* function changes the pitch of the wave variably over time by using a pitch bend envelope. Use the following steps to create your own pitch bend effect:

1. Press the *Reset Envelope* button. This restores the pitch bend envelope to its reset state.
2. Now raise the *Range* trackbar to 24 *Semitones*. This affects the range of the pitch change. In this setting, the maximum pitch bend is 24 semitones (two octaves).
3. To change the pitch bend envelope, left-click on a point on the envelope. You can move a point around by dragging it or you can remove it by double-clicking on it. Create an envelope similar to the one shown here.
4. Now, select *Save As* to store this preset. You can give this preset whatever name you'd like and then select *OK*.
5. Press *OK* again to run the function. You should be able to hear the drum hit change in pitch relative to the envelope you drew.
6. From now on, you have the option of using the effect you just created by selecting it from the drop-down list. If you decide you don't have use for this effect in the future, you can always delete the preset by selecting it in the drop-down list and selecting the *Delete* button.

To reset all parameters in a dialog to their default setting, select *Reset All* from the shortcut menu which comes up when you right-click on any unused part of the dialog.

Previewing an Operation

You can preview how a function will affect a sound file by using the *Preview* button found in most sound processing dialogs. This is useful for fine-tuning parameters without having to leave the dialog.

To A/B test the sound (with and without processing), use the *Bypass* check box. When the *Bypass* box is checked, selecting the *Preview* button will play the unprocessed data. Otherwise, you will hear the processed sound.

You can change the state of the *Bypass* check box while the preview is playing to quickly switch between A and B.

Preview Configurations

You can temporarily select the length of data you wish to preview by right-clicking anywhere in the dialog and selecting *Configuration* from the shortcut menu. In the *Preview Configuration* dialog, enable the *Limit previews to* option and specify the number of seconds from 1 to 30.

You can set a permanent preview configuration by using the *Preview* folder in the Sound Forge *Preferences* section (or by selecting the *Default all previews to the current settings* in the *Preview Configuration* dialog).

You'll also see the *Reactive previewing* check box in the *Preview Configuration* dialog. When activated, this item automatically generates a preview which allows you to hear any change in dialog parameters. This function becomes very handy when tweaking the center frequency in the *Paragraphic EQ*, for instance.

When *Reactive Preview* is active, holding down the *Shift* key while changing parameters will suspend the automatic preview generation until the *Shift* key is released.

Pre-roll and *Post-roll* times can be set in the *Preview Configuration* dialog. These determine the amount of material that will play before and/or after the selected area in a preview. The *Pre-roll* and *Post-roll* can be quickly turned on and off by right-clicking anywhere within a dialog and selecting from the shortcut dialog.

Previewing a Cut

To preview a *Cut* operation, make a selection you wish to cut and select *Preview Cut/Cursor* from the Edit menu. Sound Forge will pre-roll and post-roll audio as specified in the *Other* page of the *Preferences* folder, skipping the selection.

If no selection is made, you can use this feature to listen to sound immediately surrounding the cursor.

Processing shortcuts

To execute a function with the same parameters as the last time the function was used, hold the *Shift* key down while selecting the function. This is known as shift-clicking and works for most operations in Sound Forge, not just the effects. Shift-clicking works for toolbar buttons, as well as menu items.

You can also use the *Repeat* item (*Control+Y*) under the Edit menu to quickly repeat the last operation used. This works for all Process, Effects, and Tools functions in Sound Forge. Holding down the *Shift* key while selecting *Repeat* takes you to the last function's dialog.

Toolbars

If you like toolbars, there are toolbars for functions under the Process, Effects, and Tools menus. To use these toolbars, you must go in the *Preferences* folder (Options menu) to the *Toolbars* tab and turn on those you wish to appear on your screen.

Changing the selection affected by the function

To apply a function to a region other than the current selection, choose the *Selection* button in the function's dialog (you cannot do this with functions that do not have a dialog, such as the reverse function). This causes the *Set Selection* dialog to appear.

In the *Set Selection* dialog, you can change the *Start*, *End*, *Length*, and *Channels* selections which will be affected by the function. You also have a choice of the format in which you are going to enter these values or you can even use a preset selection.

Sound Processing Techniques

In this section, we will cover some basic concepts relating to some of Sound Forge's processing functions used for modifying (or creating) sound. Detailed information about each effect is also available in the Reference Section.

[Delay](#)

[Multi-Tap Delay](#)

[Reverb](#)

[Chorus](#)

[Flange](#)

[Noise Gate](#)

[Compression and Limiting](#)

[Expansion](#)

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[Simple and FM Synthesis](#)

[Graphic and Parametric EQ](#)

Simple Delay

[Video tutorial](#)

Simple Delay/Echo creates copies of the original sound which are then mixed with the sound file to create simple echo effects.

Open TUTOR1.WAV and select *Simple* from *Delay/Echo* in the Effects menu. Choose the *Slap-back echo* preset and select *Preview*. A single copy of the sound is heard 0.4 seconds after the original. Changing the *Delay time* determines the time between the original and echoed sound.

Now, select the *Grand Canyon* preset. Notice that the *Multiple delays/echoes* check box is marked. This means that instead of just one echo, you will hear multiple decaying echoes of the original sound. The *Decay time* determines how long it takes for these echoes to fade out.

The *Pre-delay/echo* function creates echoes heard before the original sound. Use this effect while the spaceship in your next sci-fi flick travels back in time by means of a wormhole.

Multi-Tap Delay

[Video tutorial](#)

The *Multi-Tap Delay* function allows you to specify up to eight delay-taps spaced anywhere within 2.5 seconds (0.5 seconds in 16-bit Sound Forge) of the original sound.

Open the file TUTOR1.WAV and select *Multi-Tap* from *Delay/Echo* in the Effects menu.

Now, select *Tapped delay, stereo pan* from the end of the preset list. To hear the effect, select *Preview*. Selecting *Bypass* allows you to listen to the unprocessed file. Listen to how echoes bounce between left and right channels.

This preset uses all eight delay taps. To switch between taps, use the *Current Tap* function. There is a delay time and an amplitude associated with each delay tap which determine when and how loud the delay tap is heard after the original sound. In stereo files, each tap can also be panned left or right, to simulate sound coming from all directions.

The selected tap is represented as a red vertical line on the echogram. The echogram represents an approximation of the impulse response of the current settings and can be used to estimate the decay time. Blue lines correspond to other active taps and black lines are feedback echoes.

The horizontal axis of the echogram is time, with the right-most point equal to the *Graph resolution* scale setting. The vertical axis represents the amplitude of the echo. You can change the scale factor between 500, 1,000, or 5,000 milliseconds for viewing the impulse response over different time ranges.

By changing the *Feedback*, you can change the decay time of the delay.

If you set the *Feedback* too high, the echoes keep adding up until they are clipped (use the 5000 ms *Graph resolution* setting). Too much feedback also creates unnatural ringing of some frequencies.

Select the *Default all parameters* preset. Here, there is only one tap active, with no feedback. This corresponds to a simple delay. Set the *Dry out* to zero and turn on the *Low-pass start frequency*. Low-pass filtering removes high frequencies, in the same way that many building materials absorb high-frequency sounds. Adding modulation creates slight pitch changes, which can make a delay sound fuller.

Now, switch to the *Chorus-reverse verb 1*. Here, the delay taps increase in magnitude over time, thereby creating a swelling effect. Many other sound effects are possible if you experiment a little with tap arrangements.

Reverberation

[Video tutorial](#)

The *Reverb* function is used to simulate the acoustics of different environments. When a sound is generated inside a room, you first hear the original sound, followed by a quick succession of echoes. These echoes are the early reflections. Then, as the echoes accumulate, you hear of thousands of echoes combined into a smoothly decaying reverberation.

Open TUTOR1.WAV and select *Reverb* from the Effects menu. Select the *Bright hall* preset and hit *Preview*. Notice how the imagined space where the recording took place changes from a small studio to a large, rich hall. You can adjust the loudness of the reverberation and early reflection by using the *Reverb out* and *Early out* faders.

To increase the reverberation length, try changing the *Decay time* from 2.5 to 5.0 seconds. The *Pre-delay* parameter specifies the time between the original sound and the start of the reverberation.

Now, change to the *Sewer* preset. This preset uses a different *Reverberation mode*. Notice how its tone quality is very different – very metallic. The reverberation modes provided give you many choices.

Chorus

[Video tutorial](#)

The *Chorus* function is used to simulate multiple sound sources from a single sound. This is achieved by mixing a delayed, pitch-modulated copy of itself to the sound source.

With TUTOR1.WAV open, select the *Chorus* item from the Effects menu.

Preview the *Chorus 2* effect to hear an example of chorusing. Increase the depth to 50% to have a slowly de-tuning effect. The rate changes how fast the de-tuning occurs.

Changing the *Chorus out delay* time affects how much later the modulated sound is heard. Very small values usually create interesting filtering effects, such as flanging. With larger values, chorused echoes are created.

The *Feedback* and *Chorus size* parameters change the strength of the chorus. Large *Feedback* or *Chorus size* values intensify the chorus. You can use the Low-pass filter to remove unwanted high frequencies.

Flange

[Video tutorial](#)

The *Flange/Wah-Wah* function is used to create the sweeping effects often heard in 60's guitar recordings and techno-sounds of today. With this function, you can recreate the sounds that flanger, wah-wah, and phaser pedals are famous for.

To hear what flanging sounds like, open TUTOR1.WAV and select *Flange/Wah-Wah* from the Effects menu.

Preview the *Slow flange 2* preset. The *Rate* and *Depth* parameters are similar to the ones in the *Chorus* effect and can be used to create faster, slower, or stronger flanging.

Noise Gate

[Video tutorial](#)

When recording a sound, there is often an audible noise floor during silent breaks. Noise is generated by many different things, including electrical equipment, machinery, and traffic outside your window. When your sound source is much louder than this background noise, it is simple to remove the noise during silent breaks, where the noise is most noticeable, with a noise gate.

For example, open TUTOR1.WAV. Move the cursor to a location between Wow and Sound and select *Insert Silence* (Process menu). Specify 1 second of silence to be inserted at the cursor. Next, create a new file. In the new file, use *Synthesis|Simple* (Tools menu) to generate four seconds of noise at -60 dB amplitude (keep the frequency around 2,000 Hz). Next, copy all of the noise on to the clipboard and mix it to the beginning of TUTOR1. Use 100% for the noise (*Source*) and 100% for the voice (*Destination*). You should now have a file with voice and a relatively low hissing sound in the background.

To remove the noise during silent breaks, first you should analyze the amplitude of the noise. In TUTOR1, select a region with only noise (no voice), and run the *Statistics* function under the Tools menu. Take note of the *Maximum sample value* percentage. If you followed the preceding steps, it should be close to -60 dB.

Next, select *Noise Gate* from the Effects menu.

In the *Noise Gate* dialog, select the *Noise gate 1* preset. Slide the threshold level to a bit over the *Maximum sample value* (in percentage). A value of -50 dB should be high enough to differentiate the noise from the vocals. Press *OK*. Notice that the noise during the silent regions should have disappeared (total attenuation). If it didn't, try raising the *Threshold level* until it does.

The other two parameters in the *Noise Gate* determine how fast the gate will open and close. This will affect how the beginning and end of the sound will fade in and out. If the *Attack time* is too slow, the beginning of the vocals might get cut off. Likewise, if the *Release time* is too fast, the end of the vocals will be cut. Experiment with these times until you can remove as much noise as possible without removing any of the voice.

Compression and Limiting

[Video tutorial](#)

Compressing and limiting are terms used to indicate effects that lower the dynamic range of a sound. When you compress a sound, you lower the volume of loud sections and then raise the overall volume to compensate. This is done to keep the volume level from fluctuating too much over time. Limiting works exactly like compression, but to a higher degree.

Compression and Normalization can be used together to increase the apparent loudness of a file. This is currently a popular mastering technique.

Open the file TUTMUSIC.WAV. Make a copy of it by selecting the whole region and dragging it to an open space in the Sound Forge workspace. Now, with the new file active, select *Dynamics* from the Effects menu and choose *Graphic*.

The *Graphic Dynamics* dialog allows you to precisely tailor the gain at all input levels of a signal. The *Input vs. Output level* graph shows you the gain that will be applied to the input signal, depending on its level over time. The diagonal line, referred to as the *No Gain* line, indicates where the input and output levels are equal. When an envelope point is below the line, gain reduction occurs.

If you understand the principle of the graph, you can edit its points directly to create your own compression, expansion, limiting, or noise gating curves (see the Reference chapter for more details). However, you don't need to use the graph at all. Using presets and the threshold and ratio trackbars is sufficient for most cases.

Choose the *2:1 Compression starting at -18 dB* preset, turn *Auto gain compensate* off, and select *OK*. If you can see both files at once and are zoomed out, you'll be able to notice how the levels of the new file are more constant than the original. Listening should reveal the same thing.

In this preset, sounds above the *Threshold* (-18 dB) are attenuated at an input to output ratio of 2:1. Sounds below the *Threshold* are not affected by the compressor. The *Auto Gain Compensate* check box raises the total volume of the file after processing to bring the overall output level back to an acceptable level. Turn it on once and you'll see how the overall level is much higher now. This can also be done manually with the *Output gain* fader.

Undo the previous operation and select *Dynamics* again. Now, increase the *Ratio* to 7:1 and select *OK*. The sound will be even more compressed, meaning that more attenuation and boosting will be applied to compressed signals.

Too much compression starting at too low of a threshold level will usually lead to a distorted sound.

Again, undo the operation and select the *Limit levels to -6 dB (hard limiter)* preset. Now, signals above the *Threshold* (-6 dB) will still be attenuated, and with a huge ratio of Infinity:1. Essentially, after processing, there will be no levels above -6 dB included.

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 done on peaks; if the *Threshold* level is set too low, heavy distortions will occur.

Expansion

Dynamic expansion is the opposite of compression and limiting. Any signal below the threshold will be attenuated by a certain ratio. The most common effect of expansion is to attenuate low-level noise, such as a noise gate. However, you can also use expansion to add more dynamic range to a sound.

With the TUTOR1.WAV file open, select *Dynamics|Graphic* from the Effects menu. Choose the *Soft noise gate below -36 dB* preset and select *OK*. Notice how very low-level sounds have been attenuated. This is useful for removing noise from silent regions in a similar fashion as a noise gate.

Multi-Band Dynamics

[Video tutorial](#)

The other type of dynamics processing that Sound Forge allows for is *Multi-Band Dynamics*. Essentially this is a compressor that only affects certain frequency bands. The possibilities with this type of compressor are endless, but the most common use is for de-essing and de-popping.

You can also use this function as an advanced *EQ* for such things as tightening up a bass line or drums.

Open the file SIBIPLOS.WAV. This file contains both loud sibilants (S's) and plosives (P's). Let's first remove the sibilants. Double-click on the region named *...and sibilants* to select it. Next, select *Dynamics|Multi-Band* from the Effects menu and choose the *Reduce loud sibilants (de-esser)* preset. This preset acts upon frequencies around 5,000 Hz. Previewing the effect should make the S sounds less painful.

Next, make a selection on the region named plosives. This can be done by pressing the *Selection* button and choosing *Region: ...and sibilants*. In the *Multi-Band Dynamics* dialog, choose the *Reduce loud plosives* and hit *Preview*. The loud P becomes less obtrusive and unlike *EQ*, this reduction doesn't affect the entire sound file. It only acts upon the plosives.

There is also a *Multi-Band* preset called *Reduce plosives and sibilants* which you can use to remove both S's and P's at once. This is basically the same effect as running one first and then the other.

When applying these presets to your own files, you will most likely need to change the *Threshold* setting until only the unwanted sounds are affected. If the *Threshold* is too low, too much filtering will occur. Setting the filter to *-Inf.* forces filtering of all the data. Also, you can also boost a band after compression by using the *Gain* control to compensate for any loss during processing.

Changing Pitch and Time Duration

You are probably familiar with how changing the playback rate of a recording affects its pitch. For example, playing a 33 1/3 RPM vinyl record at 78 RPM makes the Beatles sound like the Chipmunks. Likewise, playing a 78 RPM record at 33 1/3 makes a trumpet sound like a tuba. This concept is used by most sampler units to make a single sound file achieve different pitches.

Open the file SAXRIFF.WAV. Under the Effects menu, select *Pitch|Shift*.

Next, select the *Fifth up* preset and select *OK*. Notice that the length of the file is less than its original size. Listen to the sound to hear both a higher pitch and a faster rate.

Undo the last operation and return to the *Pitch Shift* dialog. Now, select *Fifth down* and select *OK*. The file is now larger, slower, and lower in pitch.

Altering Pitch Without Changing the Duration of the Sound

Undo the last operation and return to the *Pitch Shift* dialog. In the last two examples, the *Preserve duration* check box was left unchecked. When you check this box, a completely different algorithm is used to change the pitch of the sound.

Select the *Third down* preset and check the *Preserve duration* check box. Next, you must select a mode. Different algorithms are included for different kinds of sounds. These modes are only suggestions, since many will work with different types of sound. It's best to just try a few different ones and choose the one that sounds best to you. For this saxophone riff, the mode named *A13. Solo instruments 4* does an excellent job.

Now, preview the sound. The sound file will now have a higher pitch, but it will not sound as if it's being played back faster.

Changing Time Duration Without Changing the Pitch of a Sound

Another Sound Forge process unique to digital processing is the ability to stretch or compress the time duration of a sound without altering the pitch. This is useful for lengthening or shortening sounds to meet a specific time length.

Open SAXRIFF.WAV and select *Time Compress/Expand* from the Process menu.

This process is very simple to use. In the dialog, you have the ability to specify the final duration of a sound, from 50% to 500% of the original length. Like in the *Pitch Shift* dialog, you must choose a mode from a list depending on the type of material you're processing. The mode named *A13. Solo instruments 4 (less echo)* does a good job with this sound file. Change the *Final length* from 2.5 to 2.0 seconds (80% of original) and select *OK*. Press the *Play* button and you should hear the riff played at a faster tempo.

The *A* and *B* modes used in both *Time Compression/Expand* and *Pitch Shift* without change of duration work in a similar fashion. Modes labeled *C* are from Sound Forge 3.0. Basically, the reason why there are so many options is that compromises must always be made between having excellent time localization (no echoes) and accurate frequency reproduction (no low-frequency distortion or flanging).

If you have sounds that have few loud, sustaining low-bass frequencies, you can use modes that will result in no echoes. These include the *Drum* modes and some of the *Solo Instrument* modes. For complex music, you must sacrifice some time localization for the ability to reproduce long bass notes.

Since classifying sounds into strict categories is not possible, different settings are included for each sound type. For example, if you have a drum recording with bass drum, snare, and cymbals, you find the *A17. Drums, unpitched (minimum echo)* does an impeccable job. However, once you start adding ringing low toms and congas, the *A18. Drums (better for toms)* preset will work better. Again, experimentation is the key.

Small and Accurate Changes in Time Duration

If you have a sound file that is one minute and three seconds long, but you need it to be exactly one minute, there are two ways to go about it. If a very small change in pitch is acceptable, you might try using the *Pitch Shift* with no *Preserve duration*. Keep the *Semitones* at 0 and adjust the *Cents* until the required *Final length* is reached. It is sometimes possible for this method to give better sounding results than when using the *Time Compression* algorithm.

Synthesis

Simple Synthesis

The *Simple Synthesis* tool can be used to generate a simple waveform of a given shape, pitch, and length. More complex waveforms can be generated with the *FM Synthesis* tool.

Create a new window by selecting the *New* item from the File menu and set the data format for the window to be *16-bit, 22,050 Hz, Mono*. Next select *Synthesis|Simple* from the Tools menu.

Select the *Middle C Reference (3 seconds)* preset and press *OK* to generate a sine wave in the window that is 3 seconds long and has a pitch of 261.52 Hz (Middle C). Press the *Play* button to hear your reference tone.

FM Synthesis

Sound Forge's *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 de-tuned over time. However, if the frequency of the modulator is high, the carrier will be modulated so fast that many additional frequencies, or sidebands, are created.

Sound Forge allows up to four waveforms (operators) to be generated in a variety of configurations. Depending on the configuration, an operator can be a carrier, a modulator, or a simple unmodulated waveform. Waveforms can also be added together (additive synthesis) to add more complexity to the sound.

The best way to become familiar with this module is to start with a single operator configuration and try out all the different waveforms by themselves. Experiment using *Feedback* on each waveform to hear the different effects of self-modulation. *Feedback* adds overtones to the sound by modulating the sound with itself. Zero feedback indicates simple synthesis.

In the second configuration, two unmodulated operators are mixed together (horizontal connection) and heard simultaneously (additive synthesis).

Now change the *Configuration* to two stacked operators (vertical connection) and again experiment using different waveforms and frequencies on the carrier (bottom operator) and the modulator (top operator). With the modulator *Frequency* set low (1-5 Hz), lower the modulator *Amplitude* to create slight de-tuning. Raising it creates big pitch bends. If the modulator *Frequency* is set high, many unusual FM sounds can be achieved. Make sure that the modulator *Amplitude* is not too high, otherwise the result will be harsh, noise-like sounds.

Setting the modulator *Frequency* to 0.00 allows absolute control of the carrier's pitch by using the modulator's envelope, which you can use for de-tuning over time.

Adding more operators increases the complexity of the waveform. When both FM and additive synthesis are combined, an almost endless variety of sounds can be generated.

DTMF Synthesis

The *DTMF Synthesis* tool is used to generate the dial tones used by telephone companies. These dial tones correspond to the numbers in a telephone unit, along with other special codes.

For example, select the *Sonic Foundry, Inc.* preset. The dial string shows Sonic Foundry's phone number. The *Amplitude* control determines how loud the tones will be. Press *OK* to generate the tones. Now when you play back the sound file, you'll hear the tones which when played over a phone line would trigger a call to Sonic Foundry.

Filtering

[Filtering Intro](#)

[Paragraphic EQ](#)

[Graphic EQ](#)

[Parametric EQ](#)

Filtering Intro

What can you do when you have a sound that is not quite perfect, but not bad enough to throw away? Sometimes, judicious use of filtering can keep a favorite sound bite from losing favor. Filtering is not a cure-all for bad sound, but slight alterations of a sound can bring it back to life.

Sounds are composed of varying amounts of one or more frequencies. For example, a sound with a rich timbre will contain many different frequencies. A sine wave sound has only one frequency in it, like 60 or 440 Hz.

Filters allow the tailoring of the frequency spectrum of a sound. A spectrum is nothing more than a representation of how much of each frequency component (from 20 Hz up to one-half the sample rate) is present in a signal. Less of any given component means that the given frequency component is not as prevalent aurally as another component. Filters pass or reduce (attenuate) frequency components.

Many factors such as the frequency response characteristic of your speakers and sound card can also affect which frequencies are more prominent during the playback of a sound. In general, the goal is to make a recording sound as close to the original as possible. However, filtering is also commonly used to remove unwanted sounds such as noise, or make individual instruments in a recording sound louder.

Paragraphic EQ

[Video tutorial](#)

The *Paragraphic EQ* is a set of six very flexible parametric filters. Four independent band filters allow you to boost or attenuate specific frequency ranges. Also, two shelving filters let you control the amount of low and high frequencies in your recordings. A *Gain vs. Frequency* graph shows the overall effect of the combined filters, making it easier to visualize the final sound.

Open TUTMUSIC.WAV and select *EQ|Paragraphic* from the Process menu. Next, select the *15 dB notch centered at 1000 Hz* preset. Notice how the graph goes below the centerline at 1000 Hz.

Hit *Preview* and listen to the very strange, hollow sound caused by the missing mid-range frequencies. Moving the *Gain* fader of the third filter further down from -15 dB to -25 dB will increase the effect. Next, move the fader - 25 dB to +9 dB and listen to the drastic tone difference.

Double-clicking on a band's *Gain* fader resets its value to 0 dB, which disables the band. To disable all bands, right-click on an unused portion of the dialog and select *Reset All* from the shortcut menu.

Each of the four bands has a *Center frequency*, *Gain*, and *Width* control. The *Width* control determines how many octaves above and below the center frequency will be affected by the filtering. For the most part, try to keep the width around one octave. If the width is small, fewer frequencies will be affected.

Select the *Boost bass frequencies below 250 Hz by 4 dB* preset and hit *Preview*. Here, the low-shelf filter is used to make low frequencies louder. Change the gain from +4 to -15 dB and check out how most of the bass has been drastically reduced, as if being played through a small radio.

Open the file CREEK.WAV. This file contains a recording of a small stream. If you look at the waveform, you'll notice that there are some very high level, low frequency waves in this recording, probably caused by the wind. They are inaudible, yet if you look at your speaker's woofers during playback, you'll see them bouncing up and down. This large excursion can damage some speakers and forces both the amplifier and woofer to work overtime. Woof!

To remove these frequencies, go to the *EQ|Paragraphic* dialog and select the preset titled *Remove very low and inaudible frequencies below 20 Hz*. Now hit *Preview*. Turning the *Bypass* check box on and off, you should notice no audible difference. Hit *OK*. Notice how the large waves are gone and your woofers are no longer crying for help.

Removing very low and inaudible frequencies after recording is a very good idea. For one, your waveform will look more natural and the level meters more accurate. Also, edits and effects will behave more predictably, giving you extra headroom.

Graphic EQ

[Video tutorial](#)

The *Graphic EQ* divides all the possible frequencies into ten bands which you can boost or attenuate (cut). Each band has a *Center* frequency related to it. The *125 Hz band*, for example, affects frequencies between 90 and 190 Hz. The *Gain* value above each band indicates the amount of cut or boost applied to the band. When a band is set to zero, it means that the frequencies in the band will not be modified. Positive gain values indicate a boost and negative values indicate a cut in amplitude.

Open TUTMUSIC.WAV and select *EQ|Graphic* from the Process menu.

Select the *Cut high frequencies above 4 kHz by 12 dB* preset and press *OK* to remove the high frequencies from the music bed file. Press the *Play* button to hear how the music is much duller.

Parametric EQ

[Video tutorial](#)

Sound Forge has four basic types of parametric *Filter styles*: *High-pass*, *Low-pass*, *Band-pass (peak)*, and *Band-reject (notch)*. A low-pass filter alters frequencies above a certain value, and a high-pass filter alters frequencies below a certain value. A band-pass filter alters frequencies outside of a specified range. A band-reject (or notch filter) alters frequencies inside a specified range. Generally, each filter attenuates frequencies in its stopband range(s) and leaves frequencies in its passband range alone.

The rate at which the passband transitions into the stopband is the transition band (or rolloff). When you specify the characteristics of a filter in Sound Forge, you will no doubt be tempted to specify that a low-pass filter will start to transition at, say, 1,000 Hz and finish transitioning by 1,001 Hz. The problem with doing so is that there are mathematical limitations to creating such a filter. When you specify a sharp transition band, the software will try to design a filter that will take a long time to apply and will unnecessarily filter the signal to overly exacting requirements. Razor-sharp filtering can often create unwanted side-effects. The more subtly you filter a signal, the more natural it sounds.

When specifying the attenuation amount for your filter, you are selecting how much to de-emphasize the frequency components that you do not want to hear. Numbers that are more negative suppress the frequency components more heavily. In other words, -6 dB of stopband rejection is going to let more unwanted frequencies through than -15 dB will.

So, let's say you just recorded a great hamster sneeze, but there is something not quite right about it when you play it back later. What can you do to restore the hamster sneeze to its rightful place as your favorite sound in your Windows environment?

Maybe you used a microphone preamp with bad common-mode rejection characteristics, or you set the hamster right next to a lamp when you recorded the sneeze? Well, don't panic, because Sound Forge can fix that sound. Try applying a 60 Hz notch filter to the sound. A notch filter will notch-out the frequencies in a range about 60 Hz so that they are not as readily heard relative to other frequencies.

Maybe the hamster sneeze has too much hiss in it? Try applying a low-pass filter and see if that makes it better. Remember, a small amount of filtering goes a long way.

Open TUTMUSIC.WAV and select *EQ|Parametric* from the Process menu.

Select the *Phone-line effect* preset and press *OK* to run the filter. Press the *Play* button to hear how the music is much thinner sounding, quite like hearing it over the phone. The *Band-pass* filter has attenuated frequencies below 250 and above 7000 Hz.

The Playlist and Regions List [Video tutorial](#)

In this section, we will go over Sound Forge's ability to create regions and markers in a Data Window and how to use the Playlist to arrange them. Each Data Window has its own Regions List and Playlist stored along with the sound data when using the .WAV file format.

[Uses for the Regions List and Playlist](#)

[Creating Regions](#)

[Editing Regions](#)

[Arranging Regions in the Playlist](#)

[Creating a new file from the Playlist](#)

[Using Markers](#)

Why use the Regions List and Playlist?

Fast Navigation

The most basic use of the *Regions List* is for dividing a sound file into separate named regions. These regions can then be quickly played or highlighted in the data window. You can also mark important time positions with markers so that you can get around large files faster.

Destructive vs. Non-Destructive Editing

When you perform edits on sound files using cut-and-paste operations, Sound Forge must shift the sound data on your hard drive. When working with large files, this process can take longer than your patience may be able to handle. However, when you create regions in the *Regions List* and then rearrange them in the *Playlist*, only the playback (as opposed to the hard-disk storage) order changes. This is called non-destructive editing since no large chunks of data are moved or deleted in your original file.

With non-destructive edits, as opposed to hard-disk intensive destructive edits, you can listen to your changes right away. Once you are done arranging the regions, it is also possible with Sound Forge to create a new file containing your final arrangement.

MIDI Synchronization and Triggering

The regions you create can also be triggered using MIDI or SMPTE time code. This is used for synchronizing sound files to your sequencer, MIDI controller, or any other time-based media. For example, you can assign a MIDI trigger to a voice sound file in Sound Forge and then have your sequencer trigger it along with other MIDI instruments. Another use is to assign SMPTE times to special effects sound files to match the visual action on a soundtrack. Refer to the MIDI/SMPTE chapter for more information on synchronization.

Creating Regions

Creating a Region Using Menu Commands

As with most features in Sound Forge, you can create a region in your sound file in a number of different ways. One of the simplest ways is to drag a selection from a data window into the *Regions List*.

Let's create a region in the file TUTFILL.WAV. Open the file as explained earlier in the manual and then open the *Regions List* window. This window is opened by selecting *Regions List* from the View menu or by pressing the *Regions List* button on the *Regions/Playlist* toolbar. You can also open this window by pressing *Alt+1*.

To create a region, listen to the file by pressing *Play* and then select the cymbal crash at the end of the file. Next, select the *Add* command from the *Regions List* shortcut menu in the Special menu. The *Add Region/Marker* dialog will appear.

Name the region *Cymbal Crash* and then select *OK*. A region will be created and placed in the *Regions List*. Each region in the *Regions List* has a *Play* button which you can use to play that particular region. In addition, *Region Tags* that show the location of the region are created in the *Time Ruler*.

Other Ways of Creating Regions

There is also a shortcut menu to let you quickly create regions. Select the beginning part of the sound file with the drum roll and then right-click on the ruler. From the *Ruler* shortcut menu, select *Add Region/Marker* and the *Add Region/Marker* dialog will appear. Name this region *Drum Roll*. Notice that if you click on *Drum Roll* or *Cymbal Crash*, in the *Regions List* window, the region is selected in the data window.

The easiest way of creating regions is to drag them from a data window directly into the *Regions List* window. To do this, select the portion of data between the *Drum Roll* and *Cymbal Crash* regions and then drag it to the *Regions List* window and release the mouse button. The *Add Marker/Region* dialog will immediately appear upon releasing the mouse button. Name this region *Toms* and press *OK*.

Pressing *R* after a selection is made will create a region as well. This is a very fast method of creating regions, however, the *Add Regions* dialog won't pop up. This means the region will be named automatically by its time boundaries. You can change the region name later (see below).

Automatic Creation of Regions

Sound Forge can create regions automatically for you in three different ways. One is to select the *Multiple takes creating Regions* mode while recording. This will create a regions every time you hit *Record* and *Stop*. Another way is to use the *Auto Region* function (Tools menu). This function analyzes the audio data and creates regions based on peaks, or it can create regions based on your *Beats per minute* settings. The other way of automatically creating regions is to use the *Regions List|Markers to Regions* function in the Special Menu, which converts all adjacent markers into regions. See the Reference chapter for more information on these functions.

Editing a Region

Changing the end points of a region is as simple as sliding the region tags on the data window ruler (above the waveform display). When you left-click on a region tag, the corresponding region end point tag is also highlighted. Also, the name of the region is displayed in the left side of the status bar. Double-clicking on a region tag changes the selection points to the region area.

Region edit operations can be performed by right-clicking on the region tags. This brings up the *Marker/Region* shortcut menu. You can then choose from the following options:

Select	Modifies the current selection points to be equal to the region end points
Delete	Deletes the region, not the sound data, from the Regions List
Split	With the cursor between the region boundaries, this command splits the region in two using the cursor as the new region boundary
Edit	Takes you to the Edit Marker/Region dialog for manually editing region attributes
Update	Modifies the region end points to be equal to the current selection

Editing Regions in the Regions List

Another way to edit region points is to double-click on a region name in the *Regions List*. This brings up the *Edit Marker/Region* dialog, which allows you to type in region parameters or select triggers (see the MIDI/SMPTE chapter). Also, right-clicking on the *Regions List* allows you to choose from various commands in the *Regions List* shortcut menu.

Adding Regions to the Playlist

Let's now arrange the regions you've just created in the Playlist. To add regions to the Playlist you can either use the menu or you can drag regions from the Regions List into the Playlist.

Since we already have some regions defined, we can just drag the regions from the Regions List into the Playlist. First, open the Playlist window by selecting the *Playlist* item from the *View* menu or pressing Alt+2. Next, drag the Drum Roll region into the Playlist. Repeat this procedure with the other two regions. Selecting the play button in front of any region in the Playlist plays the regions sequentially starting at the selected region.

The Playlist shows the sequential order that regions will be played back. In front of the name of the region, there is a number which indicates the number of times the region will be played before the next region is played. After the name is the start time of the region. Several display options are available in the Regions List and Playlist folder in the Preferences dialog.

Arranging the Playlist regions

Let's assume that you want the *Toms* region before the *Drum Roll* region. Select *Drum Roll* from the *Playlist* and drag and drop it to below *Toms*. When you start playback from the beginning region (press the *Play* button in front of the top region) you can hear the new arrangement. To place a region between two other regions, drag and drop the region between the two other regions.

Changing the Region Playback Count

Each region in the *Playlist* can be played (looped) a number of times. To do this, double-click on a region in the *Playlist* and in the *Edit Playlist* dialog set the *Play count* to any number between 0 and 999. You can quickly change the count for any selected region by using the + or - keys rather than bringing up the dialog. If you set the *Play count* for the *Drum Roll* region to 3 you can hear the drum roll played 3 times in a row.

Playing a Region at Different Points in the Playlist

One other advantage of using a playlist is the ability to repeat the same region at different points in time without having to copy the sound data in the file. If you want to create another region with the same data, select a region and then select *Replicate* from *Playlist* in the Special menu or from the *Playlist* shortcut menu. You can also replicate a region by dragging the region and, while holding the *Control* key, dropping it at another position in the list.

Removing a Region from the Playlist

To remove a region from the *Playlist*, select the region and select *Delete* from *Playlist* in the Special menu or from the *Playlist* shortcut menu.. You can also select a region and press the *Delete* key. This operation does not affect the sound file data.

Creating a new file from the Playlist

Once you've arranged the *Playlist* regions, Sound Forge allows you to create a completely new file containing the sound data in the *Playlist* regions. For example, say you've created regions in a file and then re-ordered them in the *Playlist*. To make a new file containing the re-ordered regions, select *Playlist* | *Convert to New* from the Special menu or the *Playlist* shortcut menu.

Using Markers

While regions are used to specify a section in a sound file, a marker is a location in the sound file you wish to use as a reference point. Markers make it easier to get around the sound file, as they are often used to tag important locations.

Creating a Marker with the Ruler Shortcut Menu

A marker can be created in the same ways discussed earlier for making regions. The quickest way is to place the cursor where you want the marker and then right-click anywhere on the ruler. This will bring up the *Ruler* shortcut menu, where you then select the *Marker/Region* option. In the *Add Region/Marker* dialog, type in a name for the marker and select *OK*. If you had no selection in the data window, Sound Forge will automatically create a marker at the cursor position. If you have a selection, Sound Forge defaults the *Type* to *Region* in the *Add Marker/Region* dialog.

Editing Markers

Markers can be edited in the same way as regions. The marker tags in the ruler can be moved back and forth, and shortcut menus appear when you right-click on the tag.

Starting Playback from a Marker Location

To play the sound file beginning at a marker location, press the small *Play* button to the left of the marker name in the *Regions List* window.

Dropping Markers While Playing or Recording

Another useful feature in Sound Forge is the ability to create markers as you listen to the sound file. To do so, first start playback on any sound file. A marker will be created each time you press the M key or the *Mark-In* toolbar button. When finished, you can then rename each marker to whatever you wish. The most accurate method for creating markers during playback is to use the M key.

While recording, you can press the *Drop Marker* button (to the right of the *Play* button) in the *Record* dialog. Once focus is on this button, pressing the *Spacebar* will create other markers.

Sampler Tools

Sound Forge has been designed to allow for integration with samplers, MIDI, and SMPTE devices in your studio. In this section, we will cover how you can use Sound Forge along with other music hardware or software.

[The MIDI Keyboard](#)

[Creating and editing loops](#)

[Editing a loop without the Loop tuner](#)

[Looping Techniques](#)

[The Loop Tuner](#)

[Using the Sampler Tool to download and upload samples](#)

The Sound Forge MIDI Keyboard

With the Sound Forge MIDI keyboard, you can control internal or external synthesizers and samplers from Sound Forge. For example, you might want to test a sound in a sampler after sending (downloading) it from Sound Forge. You can also use the MIDI keyboard to listen to the different sounds on a synthesizer, in the synthesis section of your sound card, or to practice your mouse-keyboard chops.

Opening the MIDI Keyboard

First, open the MIDI Keyboard window by selecting *Keyboard* from the *View* menu. Like any other toolbar or pop-up window in Sound Forge, you can resize it, move it around, or dock it to any side of the Sound Forge window. Depending on how you resize the keyboard, the keys will be arranged horizontally or vertically.

When you left-click on any of its keys, the keyboard automatically turns on if it is not already on. If you don't hear anything, it's probably because its output has not been connected to a MIDI output device.

Configuring the MIDI keyboard output port and channel

To select a MIDI device, left-click on the MIDI OUT button. A Shortcut Menu will appear listing all the devices available for output, like your sound card or other MIDI port. Select the desired device. At the bottom of the list is an option to have the MIDI Keyboard send program changes or not. Check it if you want to be able to select instruments in the device from the keyboard.

Next, you must select the MIDI channel to be used on the chosen device. Most MIDI devices are configurable to accept MIDI commands on any channel.

Still don't hear anything?

If after selecting the correct device, you still can't hear anything when you play on the keyboard, check the following things:

- Set the MIDI Output Velocity of the keyboard to a high value (above 100).

- Set MIDI input channel in sound module to the same channel as the 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.

Selecting Instruments

To change to a different voice in the synthesizer, select the instrument name from the Instrument List. Patch names are arranged as specified in the General MIDI standard (with synthesizers not using the General MIDI convention, use the patch number instead of the instrument name). Make sure that you have the *Send Program Changes* option under the MIDI Out Shortcut Menu checked, otherwise you won't be able to switch between patches.

You can also have Sound Forge generate chords instead of single notes by selecting a chord structure from the Chord Structure drop-down list. A chord is then generated using the pressed note as the root of the selected chord or interval.

A loop is a region in your sound file which is repeated on playback. In most samplers, since samples are not infinite in length, loops are used to create sounds which sustain for a long time. For example, when a sampler is told to sustain a flute sound for a long period of time, it loops a short section of the sample over and over. Loops are also used to repeat sections of music, although for these purposes it is recommended that you use the *Playlist*.

Creating and editing loops

A loop is a region in your sound file which is repeated on playback. Since samples are not infinite in length, in most samplers loops are used to create sounds which sustain for a long time. For example, when a sampler is told to sustain a flute sound for a long period of time, it loops a short section of the sample over and over. Loops are also used to repeat sections of music, although for these purposes it is recommended that you use the *Playlist*.

Ways to Create Loops

To create a loop, first select a section of the sound file. You can hear how this section will sound in looped mode by selecting the *Play Looped* button on the playbar. Next, select the *Edit Sample* item from the Special menu. In the *Edit Sample* dialog, set the *Sample type* to *Sustaining* and switch the *Loop count* from *Infinite Loop* to three and select *OK*. You could have also created the loop by right-clicking on the ruler and selecting *Edit Sample* or hitting L on the keyboard.

The data window now has tags in the ruler specifying the loop end points. You can play the sound file with the loop counts by selecting the *Play as Sample* button on the playbar. This plays the entire file, with the looped section repeated three times.

Editing a loop without the Loop Tuner

Once you've created a loop, you can change its length, start, or end-points by sliding the loop markers in the Ruler. Left-click on the loop-start marker. While holding the left-button down, slide the marker to a new location and then release the button. It's as simple as that. You can accomplish the same thing by right-clicking on the marker and selecting *Edit* from the Shortcut Menu. In the *Edit Sample* dialog, you can then enter manually start and end-points for the sample.

Other options available when right-clicking on a loop point marker are:

Select: Modifies the current selection to be equal to the loop end-points.

Delete: Deletes the loop markers.

Update: Modifies the loop end-points to be equal to the current selection.

Looping Techniques

Have you ever tried to create loops on an external sampler using the sampler's microscopic LCD display? Only then will you fully appreciate the ease of use of Sound Forge's Loop Tuner. With it, you can create loop-points visually and avoid obvious sound wave glitches which result in annoying pops.

Creating a natural sounding loop is not something that comes naturally. There are many factors involved that can cause annoying pops and glitches. While experimentation is always required, here are a few basic guidelines to follow:

Match end-point amplitudes. It's most common to select the end-points with amplitude of zero (Zero crossings).

Match end-point waveform slope. If the waveform changes slope drastically, a pop will be heard.

Match end-point sound level. What this means is that the overall loudness near (~ 100 ms) the end-points should be equal. It is often very difficult to avoid this with quickly decaying sounds, since the start point will always be greater than the end-point sound level.

Avoid very short loops. If the loop is smaller than ~50 ms (1/20 Hz), you will most likely create a loop with a pitch not equal to the sample pitch. You can experiment with pitch-tuning a loop by creating short loops with a length equal to 1/Frequency. For example, if you had a sample of pitch 440 Hz which corresponds to A5 on the keyboard, you can try making the loop 2.27 ms. Pitched loops, however, tend to not sound like the original sample.

Using the Loop Tuner

Before opening the Loop Tuner window, you must have created a loop in the sound file. For this demonstration, open the file TUTORSMP.WAV. This file has a sustain loop already specified in it.

Under the View Menu, select Loop Tuner.

The Loop Tuner allows you to view both loop end-points at once. The left-hand side of the Loop Tuner window shows the end of the loop while the right hand side shows the start of the loop. A seamless loop will not have any discontinuities; in other words, the waveform should look smooth as it goes from the end of the loop back to the start.

Playing from the Loop Tuner

There are three playback buttons in the Loop Tuner, the Play Before Loop , Play Loop, and Play After Loop. You can also still use the Play Sample button in the Data Window to hear the entire sample with repeated loops.

Switching between Sustain and Release Loops

To switch between sustain and release loops, press the left-most button in the Loop Tuner. An S indicates Sustain, while an R indicates Release. Keep this on Sustain for now (the S should be showing).

Fine-tuning loop end-points with the Loop Tuner

If you want to change the loop end-points by a small amount, you can use the Start Point and End-point Edit Box spinners. To increment the loop start point by one sample, click on the Start Point Edit Box spinner up arrow.

If you left-click on the center button of the spinner, you can increment or decrement the start point by larger amounts by sliding the mouse up and down while holding the left-button down. Notice how the start point waveform (right side of the Loop Tuner display) and the Loop Marker in the Data Window slide along horizontally in time.

On the lower right hand side of the Loop Window, the sample amplitude at the loop end-points is displayed. Although it is highly dependent on the waveform, a good rule of thumb is that the closer these two values are to each other the better.

Zero Crossing Finder

As we mentioned earlier, loop end-points should match amplitude and slope. The Zero Crossing Finder accomplishes this by finding the next zero crossing of the waveform with a positive slope (you can select the slope in the *Editing* folder in the *Preferences* dialog).

There are two Zero Crossing Finders for each loop end. One for finding the next zero crossing and one for finding the previous zero crossing. To use the finders, just left-click on one of the buttons. By trying out different locations, moving the start and end-points forward and backward you should be able to quickly find loop ends that meet smoothly.

Press on the Play Loop button in the Loop Tuner to hear the current loop. If the loop still clicks when played, try moving to different zero-crossings. Remember, you can also use the Loop Point spinners to fine tune the end-point location.

Lock Loop Length

Let's assume that you want to shift a loop's position without changing its length. Pressing the *Lock Loop Length* button (*Options* menu) allows you to do so. When on, any operation you use to change a loop point affects both the start and end of the loop, therefore keeping the loop length constant.

For example, with the *Lock Loop Length* button down, drag the Loop Start Marker in the Data Window Ruler to the left. The start and end markers will slide together. Turning the *Lock Loop Length* off allows

you to shift each end-point individually. The Lock Loop Length button works the same as the Lock Loop/Region Length item in the View menu.

Real Time Loop Tuning

The tools mentioned above for changing the loop end-points can be used while the loop is being played. This allows for instant feedback on how the loop sounds. To do so, start playback of the loop by selecting the Play Loop button and proceed to use any method to change the loop length. Sound Forge will play the modified loop points.

Saving Loop Points

Loop points are automatically saved when you save your sound file in WAV, VOC, SDS, SMP, DIG, PAT, and AIF file formats. You are allowed to disable the saving of loop points when saving to the WAV format.

MIDI and SMPTE Synchronization

In this section, we will cover how you can synchronize Sound Forge with devices generating SMPTE Time Code or use a MIDI sequencer or MIDI controller to trigger audio playback.

[What is MIDI?](#)

[The Sonic Foundry Virtual Router](#)

[Using MIDI Triggers](#)

[MIDI/MTC synchronization](#)

What is MIDI?

The Musical Instrument Digital Interface (MIDI) is a set of commands that many pieces of music software and hardware use to speak to each other. MIDI is most often used for sending commands such as Play Middle C Now but can also be used to send information like Current Time is: 00:00:01:23 SMPTE or even digital sound data (see Sampler Tools section).

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 boxes, you would run a MIDI cable from the sequencer's MIDI OUT port to the synthesizer's MIDI IN port.

Sonic Foundry Virtual MIDI Router

NOTE: If you have not installed the Sonic Foundry Virtual MIDI Router (VMR) driver yet, we recommend that you do so at this point. Refer to the Appendix labeled [Sonic Foundry Virtual MIDI Router](#) for installation instructions.

The VMR allows you to route MIDI signals between software programs such as a MIDI sequencer and Sound Forge in much the same way that you would run MIDI cables between MIDI hardware devices. The only difference is that there's no need for patch cords or extra hardware.

Sound Forge and MIDI Triggers

[Video tutorial](#)

There are many different ways to generate MIDI commands, including MIDI sequencers, MIDI keyboards, MIDI guitars, and there's probably MIDI maracas out there somewhere. Whatever their source may be, MIDI commands can be used to trigger audio playback in Sound Forge.

Triggering file playback from the Sonic Foundry MIDI keyboard

To demonstrate how to use MIDI Triggers, we will first show you how to control Sound Forge using the internal MIDI keyboard. This might not be the most useful way of using MIDI triggers, but since everybody using Sound Forge will have access to the Sonic Foundry MIDI keyboard, this will be a good demonstration of using triggers and the VMR. We are assuming that you have the driver installed.

First, open the Keyboard Window by selecting *Keyboard* under the *View* menu or by pressing Alt+3. Even though the Keyboard Window is intended for controlling MIDI devices like sound cards or samplers, it can also be used to trigger Sound Forge itself. Next, open a sound file that you wish to trigger.

The first step in setting up MIDI triggering is to make sure that the output of the Master goes to the input of the Slave. In this case, the Sound Forge MIDI Keyboard is the master and Sound Forge is the slave. Left-clicking on the MIDI Out button in the Keyboard allows you to set where the Keyboard will send its MIDI commands. For this example, set it to *1 Sonic Foundry MIDI Router*. Also, make sure that the output channel is set to Channel 1. You won't be able to hear anything at this point when you press on the keyboard because nothing is connected to the other end of the VMR.

Now, go into the *MIDI/Sync* folder under *Preferences* in the *Options* menu. For Sound Forge's MIDI In, also select *1 Sonic Foundry MIDI Router*. At this point, we've connected the two devices, much like running a MIDI cable between them.

Next, you must select *MIDI Triggers* under the *Options* menu to determine what you want to trigger in Sound Forge. For this example, select *Play All* from the *Event* list box. In the Trigger section, select *Note*. Type in the note *C4* and set the channel to 1. What this does is tells Sound Forge to play the current sound file when it receives a C4 Note-On command on MIDI channel 1.

Finally, you must tell Sound Forge to begin waiting for MIDI commands. In the *Regions/Playlist* Toolbar, select the *MIDI Input Sync/Trigger* button to enable synchronization to MIDI input. (If the toolbar is not open, go to the *Toolbars* folder under *Preferences* in the *Options* menu and check the *Regions/Playlist* option). The *MIDI Input Sync/Trigger* button will remain pressed down and *Waiting...* will appear in the left-hand status box. Sound Forge is now ready to accept MIDI commands.

To test triggering, press the C4 key in the MIDI keyboard (the key with the number 4 on it). The sound file should start playback and play all the way through. Notice that when you press C4, the status box displays that it has received a C4 Note On message on channel 1.

It's important to understand the difference between triggering playback with the MIDI keyboard and selecting the play button in Sound Forge. When you press a note in the MIDI keyboard, MIDI commands are sent from Sound Forge to an external device (in this case the VMR) and then back to Sound Forge's MIDI input port. The MIDI commands which arrive at the Sound Forge MIDI input port could be coming from any software or hardware device that generates MIDI.

There are a few other commands in the *MIDI Triggers* dialog you can experiment with. They are explained in more detail in the Sound Forge Reference Section.

Triggering region playback from the Sonic Foundry MIDI keyboard

Let's now trigger regions from the Regions List. First, go to the *MIDI Triggers* dialog (*Options* menu) and set all the triggers to *NONE* by selecting the *All triggers off Preset*. If you have triggers in both the *MIDI Triggers* dialog and the *Regions List*, make sure that they correspond to different MIDI notes if you don't want unexpected results to occur. Next, open a file which has regions specified in the *Regions List*.

Double-click on the first region in the Regions List and select MIDI Note-On Play. Also, set the note to C4.

Make sure that the output of the MIDI keyboard and the MIDI Input of Sound Forge are set to Virtual MIDI 1 and all channels set to 1 as in the previous section. Press down the MIDI Input Sync/Trigger toolbar button to Synchronize MIDI Input (you can also do this in the *Options* Menu).

If everything has been done correctly, when you press C4 in the keyboard the first region will be played.

By repeating the previous steps, you can give each region a MIDI trigger. Set the MIDI notes to different values, like C3 and C5 so that notes on the keyboard correspond to different regions. **Note:** Sound Forge can play only one region at a time. Overlapping causes the earlier region to cut-off and the new region to be played back.

Triggering from internal sequencers

The basic concepts of MIDI In and MIDI Out routing used in triggering from the internal MIDI keyboard apply to any hardware or software device which can generate MIDI commands. You should be able to follow the instructions for triggering with the MIDI keyboard and substitute your device's output port and channel number.

Suppose you have a MIDI sequence in you sequencer to which you wish to add a vocal track starting at a set time. You could accomplish this most easily by attaching a MIDI Trigger note to the sound file in Sound Forge and having the sequencer play the trigger note at the correct starting time. You can also accomplish this using SMPTE times as shown in the next section.

Triggering from external devices

The same procedure applies for triggering with external devices. You will have to set Sound Forge's MIDI input port (under *MIDI/Sync Preferences*) to the external device output port.

Sound Forge and MIDI Time Code synchronization [Video tutorial](#)

MIDI Time Code (MTC) is a way to transmit SMPTE timing signals between devices for synchronization. For an overview of SMPTE , check out Appendix E which explains the differences in 29.97 FPS, 30 FPS , Drop and Non-Drop, and other confusions. Although MIDI Time Code is most often used to sync audio tracks to visual action on video, it can also be used to synchronize other playback devices, like a MIDI sequencer, to Sound Forge.

Sound Forge has the ability to synchronize to external MTC or generate MTC for other devices to follow.

Playing regions using MIDI Time Code from a software MIDI sequencer

It is possible to specify start times for each region in your Regions List for synchronizing digital sounds with other timed events. For example, if your MIDI sequencer or other device generates SMPTE Time Code, Sound Forge can make its SMPTE time accurately follow the device's SMPTE time and play the regions at the specified times.

Synchronizing to MTC is very similar to synchronizing to other MIDI events. However, if you just want to trigger a few sounds in Sound Forge from your sequencer, it might be easier to use Note-On MIDI Triggering and not worry about MTC or SMPTE synchronization (refer to the section on Triggering from internal sequencers). MTC synchronization uses up more of your computer's processing power, yet its the only way to go if you don't want to (or can't) generate MIDI triggers. Also, MTC synchronization allows you to specify accurate SMPTE start times in Sound Forge instead of in other software or hardware devices.

Here are the basic steps to follow when triggering a sound region in Sound Forge from a sequencer using MTC:

1. In Sound Forge, open a file and create a region in both the Regions List and the Playlist.
2. Double-click on the region in the Playlist. Select Play at SMPTE Time from the Trigger drop down list.
3. Set the SMPTE start time for the Region. Sound Forge displays SMPTE time as Hours:Minutes:Seconds:Frames. Make sure that you are using the same SMPTE Time format and frames per second in both Sound Forge and the sequencer. Refer to Status Format and Status Preferences
4. Now, in the *MIDI/Sync* folder under *Preferences*, make sure that the Sound Forge MIDI input port is set to the same port as the output of the sequencer. For example, set both ports to *1 Sonic Foundry MIDI Router*.
5. Verify that in the *MIDI Triggers* dialog (*Options* Menu) all the triggers are set to None. This prevents other MIDI commands from also creating triggers.
6. Enable *MIDI Input Sync/Trigger* mode by either pressing the MIDI Input Sync/Trigger toolbar button or selecting *MIDI Input Sync/Trigger* from the *Options* menu. Now, Sound Forge is ready and waiting for synchronization to MTC.
7. Switch to your sequencer. Set the MIDI Out port to be the same as Sound Forge's MIDI Input port. Refer to the sequencer's manual for information on setting MIDI ports.
8. Enable MTC output in your sequencer. (Refer to the sequencer's manual for information on how to do this.) If your sequencer supports generating MTC you should now be ready to sync.
9. Press play in the sequencer. In Sound Forge, you should see the MIDI In status box display the same SMPTE Time as the sequencer's time. At the set SMPTE time, Sound Forge should play the region without interrupting your sequencer's output to any other devices.

Playing regions using MIDI Time Code from an external device

If you have a hardware device (i.e. SMPTE Time Code to MTC card) that generates MTC, the procedure is basically the same as outlined above. In Sound Forge's *MIDI/Sync Preferences* folder, you should be able to directly select the device's MTC output driver as Sound Forge's MIDI input port.

Using Sound Forge to generate MTC for a MIDI sequencer

Follow these step if you want Sound Forge to generate MTC for other devices to follow.

1. Open a large sound file. Sound Forge only generates MTC while playing a sound file or the Playlist.
2. Now, in the *MIDI/Sync* folder under *Preferences*, make sure that the Sound Forge MIDI output port is set to the same port as the output of the sequencer. For example, set both ports to *1 Sonic Foundry MIDI Router*.
3. Verify that in the *MIDI Triggers* dialog (*Options* menu) all triggers are set to None. This prevents other MIDI commands from also creating triggers, which might create unexpected results.
4. Enable MIDI Output Sync mode by either selecting the MIDI Output Sync toolbar button or selecting *MIDI Output Sync* from the *Options* menu. Now, Sound Forge will output MTC whenever you start playback.
5. Switch to your sequencer. Set the sequencer's MIDI In port to be the same as Sound Forge's MIDI Output port. (Refer to the sequencer's manual.)
6. Set the sequencer's SMPTE Offset Time as required. Sound Forge always uses 00:00:00:00 as its output start point. For some sequencers, it is suggested that the SMPTE Offset Time be set to at least four seconds to ensure correct synchronization.
7. Enable MTC input (again, check the sequencer's manual). This usually entails switching the timing generator in the sequencer from Internal to MTC. You should now be ready to sync.
8. Press play in the sequencer (some sequencers don't require you to do this). The sequencer should not start playback. Instead it should switch to a Waiting for MTC mode.
9. In Sound Forge, start playback in the Data Window or Playlist. The sequencer should start following Sound Forge's SMPTE Time when the SMPTE Offset Time is reached.

Using Sound Forge to generate MTC for an external device

If you want Sound Forge to send MTC to an external device, follow the previous instructions for sending MTC to a sequencer with a few changes. Instead of sending to the VMR, have Sound Forge's MIDI output port send directly to the device's MIDI driver. Check the device's manual for more detailed information on using MTC.

Optimizing Sound Forge

This chapter contains information on how to configure your system to maximize Sound Forge's performance.

[Hard-Drive Use](#)

[Editing in Direct Mode](#)

[Fast File Saving](#)

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Hard Drive Use

Sound Forge is a disk-based sound editor which means that all editing operations are performed on your hard drive rather than in your computer's memory. This allows Sound Forge to edit sound files of almost unlimited size. Sound Forge also stores undo/redo information and clipboard data on the hard drive.

The drive you specify for *Temporary storage* (on the *Performance* page of the *Preference* folder) must have enough free space to store the data for each opened sound file, as well as undo/redo information, and the clipboard. Although this can use a lot of disk space, it allows you to edit long sound files that could never fit into your computer's memory (RAM). And hard drives are typically much cheaper than memory.

Since Sound Forge is hard-drive intensive, the faster your disk access, the faster Sound Forge will be. The easiest thing to do to improve your performance without buying new hardware is to defragment your hard drive. Over time, and especially with heavy use, a hard drive can become fragmented. This means the data stored on the hard drive gets scattered across the surface of the disk making your files discontinuous. When files are discontinuous, the hard drive must jump around to retrieve the data. This jumping around can slow down access to your files and can be very noticeable on older hard drives. DOS 6.0 and above come with a program called *Defrag* that will defragment your hard drive. Windows 95 comes with a *Disk Defragmenter* utility which can be found in the *System Tools* folder in your *Accessories* folder.

The following guidelines will help you optimize your hard drive depending on which operating system you are running.

Windows 95

Windows 95 Default Configuration

Choosing system configuration settings other than the default Windows 95 settings can result in reduced performance from Sound Forge, as well as Windows 95. To check these settings go to *System* under *Control Panel* and click the *Performance* tab. *File System and Virtual Memory* should both be set to 32-bit. In the advanced settings area of the window, click on *File System*. In the *File System Properties* dialog, the *Hard Disk* tab setting for *Read-ahead optimization* should be set to *Full* and the *Troubleshooting* tab should have all of the options unchecked.

Now, click on *Graphics* in the *Advanced Graphics Settings* dialog. In most cases this setting should be set to *Full*. If you are experiencing unexplained glitching in your audio playback, try different settings. However, if changing these settings doesn't clear up the problem, change them back. Finally, click on the *Virtual Memory* button under the *Advanced* settings of the *Performance* page. In most cases, you should choose to let Windows 95 handle your virtual memory.

Windows NT

Windows NT Default Configuration

There are no special considerations for Windows NT.

Windows 3.x

Use 32-Bit Disk Access

When running Windows in *Enhanced* mode, most PCs can be configured to use 32-bit disk access for virtual memory in Windows. Enabling this option can dramatically improve overall system performance especially if you have less than 8 megabytes of RAM. This option can be found in the *Enhanced* option of the Windows *Control Panel*. Select the *Virtual Memory* button and enable *Use 32-bit*

Disk Access (if it is not already enabled).

Note: You will need to select the *Change* button before you can see this option. Windows will inform you if 32-bit disk access cannot be used on your computer.

Windows 3.x for Workgroups Use 32-Bit File Access

Windows for Workgroups 3.x provides greatly enhanced file access performance. Enabling this option can dramatically decrease the time it takes to access data in a file. This option can be found in the *Enhanced* option of the Windows *Control Panel*. Select the *Virtual Memory* button and enable *Use 32-Bit File Access* (if it is not already enabled).

Note: You will need to select the *Change* button before you can see this option. The default *Cache Size* should be sufficient. Refer to your Windows for Workgroups 3.x documentation for more information on selecting a cache size.

Editing in Direct Mode

Whenever you open a file in Sound Forge, a backup copy is created so that the original file is not affected until you select to save your work. However, when you open a file as read-only or direct (specified in the Open dialog), Sound Forge will not create a backup copy of the file, which makes opening files quicker and uses less disk space. In fact, editing in direct mode can make opening a file almost instantaneous. Direct mode can only be used on files which are 16-bit and stored in the Microsoft Wave format.

The first time you open a file in direct mode Sound Forge will scan the file to create a peak file. This allows Sound Forge to display the data in the file extremely fast as well. This peak file is stored in the same location as the wave file with the same name as the wave file but with the extension *.sfk*. After a peak file is created Sound Forge automatically keeps it updated when you edit the file. The only time the peak file may need to be regenerated is if you modify the file with an application other than Sound Forge. This allows Sound Forge to instantaneously load the file next time you open it.

The only drawback to editing a file in direct mode is that you no longer have the security of a backup file. In most cases this is not a problem, since all changes can be undone (with ample disk space). But if, for some reason, Sound Forge terminates improperly, the file would remain in its edited state.

Fast File Saving

When creating a new file Sound Forge creates a temporary file that contains all of your new sound data. When you are ready to save the file, Sound Forge must copy this data into the file you specify in the *Save As* dialog. If you are saving the sound data as a 16-bit Microsoft Wave file, Sound Forge can make this process almost instantaneous by quickly moving the file rather than copying the data to a new file. In order for this to work, the new file into which you are saving your data must be on the same drive as your temporary storage folder. Thus it is recommended that you keep your temporary storage folder on the same drive where you keep most of your sound files. You can set your *Temporary storage folder* on the *Performance* page in the *Preferences* folder.

Playback Cursor and Record Counter

The playback cursor and record counter options in the *Preferences|Performance* folder allow you to configure whether or not the numeric counters (position information displayed in status fields) will be updated during playing and recording. If you are recording or playing at high sample rates (44,100 or 48,000 Hz) and are experiencing gapping or skipping problems in the data, try disabling the playback cursor and/or record counters to decrease overhead.

Meters

The *Play* and *Record Meters* in Sound Forge take a small amount of processing overhead during playback and record. If you are experiencing gapping during playback and you have the *Play Meters* showing, try turning off the *Play Meters* by selecting the *Play Meters* option in the View menu. If you are experiencing gapping during record and you have the *Monitor* option enabled in the *Record* dialog, try turning off the *Record Meters* by un-checking the *Monitor* option.

Passive Updating for Video and Time Displays

Selecting *Passive Update* for *Video* and *Time Display* from the Options menu, will lower the priority of the repainting for these displays during playback. If your computer is relatively slow and you are experiencing gapping during playback, or the your machine just seems to be bogging down during playback, try turning on *Passive Update*. This will force the repaints for the *Video* and *Time Display* to only update if there is time to do so. In most cases you won't even be able to tell if it is missing some updates, so turning these options on can decrease the amount of overhead during playback with little inconvenience.

Audio and Video Synchronization

If you are having trouble synchronizing the audio with your video, there are a few points to keep in mind. If your video has been opened from a slow device, such as a CD-ROM or network drive, then Sound Forge may have trouble accurately playing back the audio and video in sync. This happens because video information is streamed at all times from the original .AVI file. You should always copy the videos you are working with to a fast hard drive.

After you open an .AVI file you may be notified that your video has very few key frames. This is generally representative of tutorial videos captured with a program that saves the video as a *Run Length Encoded (RLE)* stream with a single key frame. Due to non-linear access requirements of Sound Forge this may result in difficulty with audio synchronization. It is recommended, that you save your file with more key frames and re-open it for better performance during digital editing.

Here are a few other 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 provided the *Video Preview* window is open or the *Animate Video Strip* option is enabled. After primary locations have been located, double-click and drag your audio to these markers bringing up corresponding dialogs for mixes, pastes, and crossfades.

Features like *Insert Silence*, *Clear/Delete*, and *Time Compress/Expand* are commonly used to tighten synchronization. Another useful trick is to create a region representing the delta between a video frame and audio event. Then you can enable the *Lock Loop/Region Length* and drag the delta 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.

When cropping beginning and end silence for your .AVI file, drag your finalized selection to the *Playlist*. Choosing *Convert to New* will generate a new file depicting your selected region.

Setting up Wave Devices

The Sound Mapper/Wave Mapper

Sound Forge allows you to choose any installed Windows compatible sound card for playback and recording. It also provides one more option for playback and record called the *Sound Mapper* (depending on your configuration, this may be called the *Wave Mapper*). The *Sound Mapper* is a special device which 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.

If you are having problems with breakups this may be caused by the overhead introduced when the *Sound Mapper* does format conversions. To check this, make sure that your sound card supports the data format you are playing or recording. The easiest way to do this is to choose a playback and record device that is not the *Sound Mapper*. If Sound Forge is able to play and record the sound using the actual sound card's Wave driver, then the *Sound Mapper* is not causing the breakups.

However, if the *Sound Mapper* must be selected to play or record the format that you are having trouble with, then you should convert your sound data to a format that is directly supported by your sound card. This will remove all overhead required to translate the sound data for your sound card.

There is nothing wrong with using the *Sound Mapper* when the sound data format is directly supported by your sound card. The *Sound Mapper* will simply pass the data through to your sound card which requires negligible overhead. For more information on the *Sound Mapper*, refer to the Appendix called Sound Forge and The Microsoft Audio Compression Manager. The *Sound Mapper* is a component of the Audio Compression Manager (ACM).

Total Buffer Size

The *Total buffer size* option, found in the *Wave* folder of the *Preferences* dialog, specifies the total amount of RAM that should be used for buffering when recording to or playing from the hard drive. For most systems 512 kb is a good number. If you are still experiencing gaps during playback you may want to use more buffering or record at a lower sample rate.

There are some drawbacks to increasing the *Total buffer size*. It requires more of your computer's memory and, if you have set a large *Preload size*, there can be small delays when starting and stopping playback.

Preload Size

The *Preload size* option in the *Preferences|Wave* folder tells Sound Forge how much sound data it should prepare and load into your sound card driver before starting playback. This preloading is performed between the time you press the *Play* button and the time you hear the sound play.

Turning this option on may help skipping (audible gaps) on initial playback of a sound. These gaps are normally observed by people with slow or fragmented hard drives. The trade-off for enabling this option is a slight delay before play begins; the delay time is greater for larger preload sizes. This option does not affect operations other than playing.

Some Windows sound drivers do not support this option correctly. If you are experiencing noise or dropouts at the very start of playback, try disabling this option.

Background Processing

When Sound Forge is performing operations that take a long time, such as opening a file or performing audio processing, a *Progress Meter* is displayed on the status bar. While this Progress Meter is visible, Sound Forge is doing its work. You can cancel the operation at any time by pressing the *Cancel* button to the left of the *Progress Meter*, or you can press the *Escape* key.

All processing performed by Sound Forge is done in the background. What this means is you do not have to wait for Sound Forge to finish its work before you can use other applications. If an operation is taking a long time in Sound Forge, simply switch to another application and work on something else. Sound Forge will continue processing while you work in another application.

If you minimize Sound Forge while it is processing, the percentage processed will update in the title of the Sound Forge icon. This allows you to keep an eye on Sound Forge's progress while using other applications.

You may have noticed that Sound Forge displays the amount of processing time an operation required in the status bar when it is completed. This processing time is intended for entertainment purposes only; this value is not intended for accurate profiling. The accuracy of this processing time value is heavily dependent on what other activity is being performed by your computer while Sound Forge is working. Using other applications while Sound Forge is working can cause the processing time to increase substantially.

Crash Recovery

If for some reason Sound Forge is terminated improperly when Sound Forge or, most likely, another program crashes all of the opened and unsaved sound files can be recovered excluding those opened in direct and read-only modes. Unless a file is opened in direct or read-only modes, a temporary file is created and any edits made are stored in this file. When an improper termination of the program occurs, these temporary files remain on your hard drive and can be reopened to recover any work done to the sound files before crashing. Also, the original sound files will remain unchanged until you save your work.

Crash Recovery

When starting the program, any temporary files detected in your temporary file directory will indicate that something went wrong. You have the option to delete these files if no important work has been done, or to just ignore these files. Ignoring the files leaves them intact on your hard drive.

Recover

Use this to change the ending of all file names from FORGExxx.TMP to FORGExxx.WAV. You can then open these .WAV files using the *File Open* dialog. The files will be placed in the directory specified on the *Storage* page of the *Preferences* folder.

Delete

This command deletes all the temporary files regardless of selection in the drop-down list. Use this only when you are sure that no important work is in the file.

Ignore

If you don't want to deal with the temporary files left over from an improper termination at this moment, use this command. However, it is recommended that you either rename the files and check their contents or delete them, since they are taking up valuable disk space.

Object Linking and Embedding (OLE)

32 Bit Only: OLE support is only available in the 32 Bit version of Sound Forge.

Sound Forge for 32 Bit Windows supports Microsoft's Object Linking and Embedding (OLE) technology for improved integration with other applications. This section discussing the implementation details and different uses for the OLE technology supported by Sound Forge. The following topics are covered:

Introduction to Object Linking and Embedding (OLE).

Using OLE with Sound Forge.

Tips, Limitations, etc.

Introduction to Object Linking and Embedding (OLE)

Object Linking and Embedding (OLE) is a technology developed by Microsoft to allow independent applications to behave as though they are tightly integrated. Note that this is not an all encompassing definition, but it will work for the implementation provided by Sound Forge. If you would like a more in depth look at OLE, there are many references available at most book stores that discuss the technology.

By using the OLE support in Sound Forge, you can *embed* digital audio files used by Sound Forge into documents used by other applications (such as Write, Microsoft Word and Excel). Once a digital audio file is embedded into a document, you can then edit the audio with Sound Forge at any time by activating the embedded object (usually by double clicking on it with the mouse).

Using OLE with Sound Forge

Sound Forge is an OLE *server*. What this means is Sound Forge can create OLE *objects* (serve them up if you will) that can be *embedded* (inserted) into applications that are OLE *containers* (the consumers of objects supplied by servers). Don't let the terms scare you. We'll go through some examples that will make everything clear as mud shortly.

The OLE objects that are created by Sound Forge are called *Sound Forge Audio* objects. These objects are really just *links* (oh no, another term!) to a sound file on your hard disk or network server. Therefore, when you embed a Sound Forge Audio object into Write, you are really only inserting a link (a directory and file name) to the actual sound file. We will explain why this is important to know in the Tips, Limitations, etc. section.

There are three methods that can be used to create Sound Forge Audio objects. When we mention an OLE container, we simply mean an application that can *contain* OLE objects (like Microsoft Word and Excel). The three methods are as follows:

1. Dragging and Dropping a link from the OLE Drag Source control in Sound Forge to an OLE container.
2. Using the Copy Object Link command in the Edit menu of Sound Forge to place an Object Link onto the Clipboard. This Object Link can then be Pasted into an OLE container.
3. Invoking the Insert Object command from an OLE container and selecting the Sound Forge Audio object from the list.

Drag and Drop

The most convenient method of embedding a Sound Forge Audio object into a document is to Drag a link from the OLE Drag Source control in Sound Forge and Drop the link on the document that you desire. Note that the application that you drop the OLE object on must be OLE 2.0 compatible. Because of this, you cannot drop Sound Forge Audio objects on Write so the example below uses Microsoft Word for

Windows 95 when running on Windows NT.



Follow these steps to embed a Sound Forge Audio object into WordPad (Windows 95) or Microsoft Word for Windows 95 (Windows NT) using Drag and Drop:

1. Run WordPad (Windows 95) or Word 7.0 (Windows NT).
2. Now run Sound Forge and open a sound file (TUTOR1.WAV in your Sound Forge directory will suffice).
3. Ensure that at least a portion of WordPad or Word 7.0 is visible while Sound Forge is open (if either application is maximized, you will need to restore them and re-size them accordingly).
4. Locate the OLE Drag Source control in the upper right corner of the Data Window. It is on the right end of the ruler and contains the letters OLE.
5. Click and *hold* the left mouse button on the OLE Drag Source control and then Drag the mouse cursor while holding the mouse button down.
6. Now Simply Drag the mouse over the top of WordPad or Word 7.0 and Drop it (release the mouse button). If you did everything correctly, a Sound Forge Audio object will now be embedded in your document.

A newly created sound file must first be saved before you can drag and drop the object into an OLE container. If the sound file in question needs to be saved, you will be prompted to do so when you click on the OLE Drag Source control.

There is an option in the More page of the Preferences dialog to turn off the OLE Drag Source control (hide it). It may be desirable to turn this control off if you do not plan to use the OLE Drag and Drop support of Sound Forge.

Copy Object Link

Another method for creating a Sound Forge Audio object is to invoke the Copy Object Link command under the Edit menu in Sound Forge. This method is sometimes preferable to the Drag and Drop method because you do not need a mouse.

Follow these steps to embed a Sound Forge Audio object into WordPad (Windows 95) or Write (Windows NT) using the Copy Object Link command:

1. Run Sound Forge and open a sound file (TUTOR1.WAV in your Sound Forge directory will suffice).
2. Select the Copy Object Link command under the Edit menu of Sound Forge. If the sound file needs to be saved, you will be prompted to do so.
3. At this point, a Sound Forge Audio object is on the Clipboard and ready to be Pasted into a document.
4. Run WordPad (Windows 95) or Write (Windows NT) and select the Paste command from the Edit menu. If you did everything correctly, a Sound Forge Audio object will now be embedded in your document.

Insert Object

The last method for creating a Sound Forge Audio object is to use the Insert Object command from another application. This method provides a convenient way to get a new Sound Forge Audio object embedded into a document without having to first create it in Sound Forge and copy it into the document.

Follow these steps to embed a Sound Forge Audio object into WordPad (Windows 95) or Write (Windows NT) using the Insert Object command:

1. Run WordPad (Windows 95) or Write (Windows NT).
2. Select the Insert Object command. For WordPad, use the Object option under the Insert menu. For Write, use the Insert Object command under the Edit menu.
3. From the Object Type list, select the Sound Forge Audio item and click OK. Note that WordPad has two options: Create New and Create From File. You will need to select the Create New option.
4. At this time, Sound Forge will be run automatically to create the new object. A new window is automatically opened allowing you to copy or create any new sound data you wish. Note that you *must* create the sound data in the window supplied.
5. After creating any desired sound data, simply save the modifications and close Sound Forge. If you did everything correctly, a Sound Forge Audio object will now be embedded in your document.

After the Sound Forge Audio object has been created and embedded into the document, you can edit it at any time using Sound Forge by double clicking on the object in your document.

Tips, Limitations, etc.

As stated earlier, Sound Forge Audio objects are implemented as *links* to existing sound files on your hard disk or network server. Because of this, there are some limitations you should be aware of before using the OLE support in Sound Forge.

Links can be broken, but not mended

The most important rule to bare in mind when using OLE with Sound Forge is that links can be broken, but not mended. When you create a Sound Forge Audio object, it stores only a link (which is simply a directory path) to the sound file. Sound Forge uses links because sound files tend to be very large and would usually not be acceptable to store along with your documents.

Because Sound Forge uses Object Links, moving, renaming or deleting the sound file associated with a Sound Forge Audio object will break the link. Attempting to edit the object in your document will fail because Sound Forge will not be able to locate the correct sound file.

To fix a broken link, simply delete the Sound Forge Audio object from the document and create a new one using the correct sound file.

If you need to move a sound file that is linked to a document, you can use Sound Forge to save the file to a different location and the object link will automatically be updated. However, you must *open* the sound file that you want to move as an *object* for this feature to work. To open a sound file as an object, simply double click on the Sound Forge Audio object in your document; do *not* open the sound file directly. You can then use Save As to move the sound file. Sound Forge will prompt you about updating the object link after saving the new file.

Sound files do not travel with your documents

When you copy a document that contains one or more Sound Forge Audio objects to another computer, the sound files that are linked to the objects are *not* copied with the document. If you need to copy a

document to another computer, you will need to copy all sound files associated with it as well. However, you must also ensure that the sound files have the *exact same* path on the new computer as they did on the old computer.

Using a Network Server

To ease the problems associated with moving or sharing documents between computers, it is suggested that you use a network server to store all sound files that are embedded in the documents. If you do this, all computers that have access to the network server will be able to open and edit the Sound Forge Audio objects.

There are two methods of ensuring a common network server will work with shared documents. The first is to make sure you use the same drive letter when connecting to the network server on all machines. Remember that the full path must remain constant between computers.

The second and more preferable method is to use Universal Naming Convention (UNC) paths when embedding Sound Forge Audio objects into your documents. A UNC name looks like this:

```
\\server\share\<directory and file name>
```

Therefore, if your sound files are stored on \\myserver\sounds, then open them in Sound Forge using UNC path names rather than a drive letter. You can then create the Sound Forge Audio object using either the Copy Object Link command or the OLE Drag Source control and embed the sound files into your documents. By using UNC names, you eliminate the need to use a common drive letter on all computers that share the document.

If you are running Windows 95, UNC path names are extremely simple to use. Instead of using a drive letter when browsing in the File Open dialog, select Network Neighborhood in the Look In field and browse to the appropriate network server and share containing your sound files. This method of browsing uses UNC names exclusively.

Sound Forge is not an OLE Container

Although Sound Forge can create OLE objects to embed in other applications, the inverse is not true. Therefore, OLE objects created by applications such as Microsoft Word or Excel cannot be embedded into sound files open in Sound Forge.

Process Menu

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Auto Trim/Crop

[Video tutorial](#)

The *Auto Trim/Crop* function is useful for removing unnecessary silence in a sound file and automatically fading in and out the end-points of a phrase.

Function

The *Function* allows you to select the mode in which the preset will run.

Keep Edges Outside of the Selection

Silence between the selection points and the sound material is deleted. Use this to remove silence from the edges of a selection. Data outside of the selection is not deleted.

Remove Edges Outside of the Selection

Silence between the selection points and the sound material is deleted. Anything outside of the selection is also deleted. Use this to trim a sound from a larger file.

Remove Data Beyond Loop Points

All data after the selected loop is deleted. Since most samplers never play audio beyond a loop, data beyond the loop is unnecessary.

Remove Silence Between Phrases (creates regions)

All silence between phrases inside of the selection is deleted. Regions are created between phrases. Use this to delete the silence between sound effects and automatically create regions in the *Regions List*.

Minimum Inter-phrase Silence (0.1 to 3.0 seconds)

Minimum amount of silence which must exist between phrases before a new region is created.

Minimum Length Following Loop End (0 to 48,000 samples)

When using the Remove data beyond loop points option, this number specifies the number of samples which must exist after the loop. This is used to ensure that a sampler can loop properly, since some samplers don't use the exact loop points supplied.

Attack Threshold (-Inf. to 0 dB)

Instantaneous threshold level used for the detection of the trim/crop start point. -Inf. is complete silence, 0 dB is a maximum amplitude level.

Release Threshold (-Inf. to 0 dB)

Instantaneous threshold level used for the detection of the trim/crop end point. -Inf. is complete silence, 0 dB is a maximum amplitude level.

Fade In (0 to 100 ms)

After the trim/crop start point is detected, this is the length of fade in applied to the sound file. This is useful for preventing glitches at end points.

Fade Out (0 to 100 ms)

The length of the fade out applied to the sound file after the detected trim/crop end point.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Channel Converter

[Video tutorial](#)

The *Channel Converter* is used to convert between mono and stereo formats. It can also be used to inter-mix the left and right channels of a stereo file to create interesting pan effects.

Output Channels (Mono/Stereo)

Determines the number of channels that the output sound file will have.

New Left Channel

Left (-200% to 200%)

Determines the amount of the original left channel that will be mixed onto the new left channel.

Right (-200% to 200%)

Determines the amount of the original right channel that will be mixed onto the new left channel.

Invert Left Channel Mix (On/Off)

When checked, the new left channel mix will be inverted (polarity reversed).

New Right Channel

Left (-200% to 200%)

Determines the amount of the original left channel that will be mixed onto the new right channel.

Right (-200% to 200%)

Determines the amount of the original right channel that will be mixed onto the new right channel.

Invert Right Channel Mix (On/Off)

When checked, the new right channel mix will be inverted (polarity reversed).

Convert to specified output channels only (no custom mixing) (On/Off)

When checked, the fader mix levels are ignored. When converting from stereo to mono, both channels are mixed together at 50%. When converting from mono to stereo, the mono channel is placed in both channels at 100%.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Convert to 8 Bit

[Video tutorial](#)

Use this function to convert 16-bit sound files to an 8-bit format. Because the signal-to-noise ratio of 8-bit audio is so small (42 instead of 96 dB), it is suggested that the volume of the sound file be maximized by using the *Volume* or *Normalize* functions before performing the conversion.

Truncate

This method of 16 to 8-bit conversion simply removes the least significant bits of a 16-bit sample. It is the least desirable of the three.

Round

When rounding, 8-bit values are mapped to the nearest 16-bit value, thereby slightly reducing the quantization noise.

Dither

(0.01 to 2.00 bits)

Dithering is used to reduce the audible effects of quantization noise which are most audible in 8-bit samples. This is accomplished by adding a small amount of Gaussian noise to the signal to mask the more obtrusive 8-bit quantization distortion.

The dither depth is the amplitude of the added noise in terms of the least significant bits of the sound file. For most cases, a value between 0.5 and 1.2 is sufficient.

Tip: Use very small amounts of dithering and listen to whether to quantization noise is less obtrusive. Increase the dither amount until the dither noise is louder than the quantization noise.

See also

[Sound Processing and Other Tools](#)
[Converting 16 to 8 bit](#)

DC Offset

[Video tutorial](#)

Use this command to change the baseline of a sound file. A recorded wave that is not centered around the zero baseline in the waveform display is said to have a DC offset. To correct for offsets, a constant value is added to each sample.

DC offsets are usually caused by electrical mismatches between your sound card and microphone. Glitches and other unexpected results can occur when sound effects are applied to files which contain DC offsets.

Automatically Detect and Remove

When this option is checked, the DC offset is calculated for each channel individually and then automatically corrected.

Adjust DC Offset By (-32,768 to 32,767)

Enter the offset value. The offset can be from -32768 to 32,767 for 16-bit data and from -128 to 127 for 8-bit data.

Tips: An easy way to spot DC offset is to zoom close up to a spot in the sound file containing only silence and see if the silence waveform matches the centerline in the waveform display.

An approximation of the DC offset of the waveform can be obtained by running the *Statistics* function under the Tools menu. For example, if a DC offset of 100 exists, then you should apply a -100 DC offset to cancel out the existing offset.

Compute DC Offset From First 5 Seconds Only (On/Off)

When this option is enabled, only the first 5 seconds of a sound file are analyzed when measuring the DC offset. The only time that 5 seconds is not sufficient is if a long fade-in or mute has been applied at the beginning of the file.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Graphic EQ

[Video tutorial](#)

The graphic equalizer can boost or attenuate selected frequency bands to correct or enhance a signal's frequency spectrum.

Band Attenuation/Gain Faders (-60 dB to 15 dB)

Each of the nine sliders corresponds to the gain or attenuation factor that will be applied to the specified frequency region.

The frequency on the bottom of the fader is the center frequency of the frequency band affected by the fader.

The value of the fader affects the frequency band as follows:

+15 dB Maximum boost

0 dB

No effect

-15 dB

Maximum attenuation

Accuracy (Low, Medium, High)

Determines a compromise between filter precision and processing speed. Low precision is not recommended for performing very sharp filtering, when filtering very low frequencies, or when using a high sample rate.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

[Using Filtering](#)

Paragraphic EQ

[Video tutorial](#)

The Paragraphic Equalizer is a parametric EQ with a graphical display. It consists of four fully parametric peak filters, a high-shelf and a low-shelf filter.

Dry Out **(-Inf. to 25 dB)**
Amount of unprocessed signal mixed into the output.

Wet Out **(-25 to 25 dB)**
Amount of processed signal mixed into the output.

EQ Graph

The *EQ graph* displays the amplitude vs. frequency curve generated from the current settings.

Band Filter:

Gain **(-25 to 25 dB)**
Determines the amount of boost or cut for the band. To quickly disable a band, set the Gain to 0.0 dB by double-clicking on the fader handle.

Width **(0.3 to 2.5 Oct.)**
The bandwidth determines the number of octaves (centered on the selected frequency) which will be affected by the filtering, i.e. sets the size of the band. Use a high value to affect a greater range of frequencies and a low value for a more selective filter.

Center Frequency **(20 to 15,000 Hz)**
Determines the center frequency for the band.

Enable Low-shelf **(On/Off)**
When on, frequencies below the low-shelf cutoff frequency will be attenuated or boosted by the amount specified.

Enable High-shelf **(On/Off)**
When on, frequencies above the high-shelf cutoff frequency will be attenuated or boosted by the amount specified.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

[Using Filtering](#)

Parametric EQ

[Video tutorial](#)

The Parametric Equalizer is a set of four frequency selective filters which allow for very precise changes in the frequency content of a sound signal.

Filter Type (Low Pass, High Pass, Band Pass, Band Reject)

Frequency response characteristic of the filter.

High-pass

Frequencies below f_1 are attenuated by the reject ratio. A high-pass filter removes the low frequencies of a channel and can be used to remove low-frequency rumbles such as wind, electrical hum, or traffic noise.

This *High-pass filter* works best with frequencies above 150 Hz.

Band-pass (peak)

Frequencies below f_1 and above f_4 are attenuated by the reject ratio. A band-pass filter restricts frequencies higher and lower than a specified band and is useful when trying to isolate or boost a particular frequency range of, for example, voice.

The *Band pass filter* works best when the frequencies are very close together. When they are far apart, ringing can occur.

Band-reject (notch)

Frequencies between f_1 and f_2 are attenuated by the reject ratio. A band-reject filter attenuates a selected frequency band and is often used to remove narrow-bandwidth noise such as amplifier/microphone feedback or 60 Hz electrical hum.

The *Band-reject filter* works best with frequencies above 150 Hz.

f1 - f4 (1 - Nyquist frequency)

Cutoff frequencies for each filter type corresponding to the graph.

Graph

A visual representation of the frequency response curve of the filter is displayed. The virtual horizontal and vertical axis correspond to frequency and gain.

Reject (-60 to 0 dB)

Amount of attenuation applied to the region specified by the filter type and cutoff frequencies. (0 dB is no attenuation, -60 dB is maximum attenuation)

Total Gain (-60 dB to 20 dB)

Amount of gain applied to the signal after processing. (-15 dB is maximum attenuation, 15 dB is maximum boost)

Accuracy (Low, Medium, High)

Determines a compromise between filter precision and processing speed. Low precision is not recommended for performing very sharp filtering, when filtering very low frequencies, or when using a high sample rate.

Tips

A low pass filter is used to restrict the high frequencies of a channel. Possible uses are electrical noise or hiss reduction. Lowpass filters are also used for anti-aliasing a sample before downsampling.

A high pass filter removes the low frequencies of a channel. This can be used to remove low-frequency rumbles such as wind, electrical hum, or traffic noise.

A band pass filter restricts frequencies higher and lower than a specified band. This is useful when trying to isolate or boost a particular frequency range of, for example, voice.

A band reject filter attenuates a selected frequency band and is often used to remove narrow-bandwidth noise such as amplifier/microphone feedback or 60 Hz electrical hum.

By running a filter several times, you can create sharper cutoff curves.

A zero-phase filter can be created by using the following procedure: Run a filter on a file. Reverse the file by using the **Reverse** function. Run the same filter on the reversed file. Reverse the file once again.

By running a filter in both time directions, filter phase shifts are canceled.

Note: The filters designed are FIR Chebychev linear phase filters with 15 to 512 coefficients (the number

of coefficients determines both the accuracy and processing time of the filter). In some circumstances, the specified filter parameters will be too stringent for the filter design algorithm. This occurs when: filter frequencies are set too close together or too far apart; reject attenuation is set too low; or the total gain is set too high or low.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

[Using Filtering](#)

Fade Graphic

[Video tutorial](#)

The *Graphic Fade* dialog allows you to draw a fade envelope which will be applied across the current data selection. Up to 16 envelope points are allowed.

Draw the envelope by pulling the small square boxes (drag points) up or down. You can create a new drag point by left-clicking on any point of the fade envelope. To delete a drag point, single-click on it with the right mouse button, or double-click on it with the left mouse button. Once you have finished with the envelope, press the *OK* button to apply the fade.

Dither and Noise-shape to Minimize 16-Bit Quantization Noise (On\Off)

When turned on, noise-shaping is applied to the sound file during processing to minimize quantization noise. This effect is most noticeable when the signal is at very low levels, such as near the tails of a fade-in or fade-out.

During a fade-out performed without dithering, sinusoidal waves become very blocky and eventually become square waves. You really have to crank up the volume to hear this, but if you do a fade-out on a 440 Hz sine wave, you'll eventually hear all the overtones associated with a square wave of varying pulse widths. This is most apparent when the sine wave drops from -60 to below -80 dB (try it on a 15 second sine wave that drops from -50 to -Inf. dB).

Noise-shaping works by adding a very small dither signal to the audio. The signal is combined with the existing audio in such a way that the noise that is normally generated from quantization is greatly shifted towards the Nyquist frequency. In a 44,100 Hz sampled file, noise is transferred to around 22,000 Hz. Unless you have canine hearing, audibility of such frequencies is naturally attenuated by a tremendous amount. In a way then, the noise floor of 16-bit audio is lowered.

Since quantization noise is transferred to near the Nyquist Frequency, it is strongly advised that noise-shaping not be used with sample rates below 32,000 Hz. Also, it is not recommended that you perform noise-shaping more than once on a given selection, i.e. save the dithering until the very end. However, noise-shaping is not nearly as effective when performed after a fade instead of during a fade, which is why the option to do so is in this dialog.

Show Wave

Press this check box to draw the current selection's waveform on the envelope graph. This is done automatically if the selection is small. For stereo files, you also have the option of displaying the left, right, or a mix of the two channels.

See also

[Process Menu](#)

[Video tutorial](#)

[Sound Processing and Other Tools](#)

Fade In

[Video tutorial](#)

Use this command to linearly fade a selection from a volume of 0 to a volume of 100%.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Fade Out

[Video tutorial](#)

Use this command to linearly fade a selection from a volume of 100% to a volume of 0.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Insert Silence

[Video tutorial](#)

Use this command to insert a section of silence into a sample.

Silence Length

Enter the length of the silence block in seconds.

Insert at

Start of Sample

Inserts the silence at the beginning of the file.

End of Sample

Inserts the silence at the end of the file.

Cursor

Inserts the silence at the current insertion point.

You cannot insert silence to a single channel of a stereo file since the length of each channel must always remain equal. If you wish to insert silence into a single channel you can use the [Delay/Echo](#) function to delay one channel forwards or backwards in time.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Invert/Flip

[Video tutorial](#)

Use this command to invert the sample about the base line. Although this does not make an audible difference, it can be useful for matching a sample transition when executing certain pastes, mixes, or loops.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Mute

[Video tutorial](#)

Use this command to set a selection to a volume of 0 (silence).

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Normalize

[Video tutorial](#)

Use this command to maximize the volume of a selection without clipping. The *Normalize* function scans the audio and applies a gain to raise its level to a specified (often very high) value.

Normalize Using Peak Level

Checking this option will normalize the sound file using the maximum (instantaneous) sample values detected. A constant gain is then applied to the sound.

Normalize Using Average RMS Power (loudness)

Checking this option will normalize the sound file using the detected average RMS value of the sound file. This is helpful for matching the apparent loudness of different recordings.

Normalize to (-60 to 0 dB)

This value indicates the level to which the sound file will be normalized. For example, when using peaks, if the peak level is -10 dB, and the *Normalize to* setting is set to -3 dB, a constant boost of 7 dB will be applied to the entire file.

When using RMS levels, normalizing to 0 dB means boosting the signal so that it has the same apparent loudness as a 0 dB square wave, meaning incredibly loud. If you were to do so, all of the dynamic range of the signal would be squashed and all the peaks would either be clipped or seriously compressed (see below). The lesson is, normalizing a peak to 0 dB is OK, but normalizing RMS to anything above -6 dB is playing with fire.

Scan Settings

These settings determine the response of the RMS power scan to the dynamics in the sound file.

Ignore below (-Inf. to 0 dB)

Using this control, you can determine the level of material you want to include in the RMS calculation. Any sound material below the threshold will be ignored in the calculation. This is useful to eliminate any silent sections from the RMS calculation. You should set this parameter a few dB above what you consider to be silence.

If you set this value to minus infinity, all sound data will be used. If the value is set too high (above -10 dB), there is a good chance that the RMS value is always below the threshold. In this case, no normalization will occur. Therefore, it is good to test the threshold by using the *Scan Levels* button.

Attack Time (1 to 500 ms)

This control determines how fast the scan should respond to transient peaks in the sound file. A slower attack time will tend to ignore fast peaking material.

Release Time (1 to 500 ms)

This control determines how fast the scan should stop using transient peak material after it has begun to drop in level. A slower release time will increase the amount of material included in the RMS calculation.

Use Equal Loudness Contour (On/Off)

Checking this option causes the scan to take into effect the *Fletcher-Munson Equal Loudness Contours*. Essentially, very low and high frequency material is less audible than mid-range audio. This option forces the scan to weigh this factor into the RMS calculation.

Scan Levels

This button runs the *Peak* and *RMS* scan function on the selection and displays an RMS level as well as the highest peak level found. Use this information to set the *Normalize to* level.

An asterisk (*) following a level indicates that the values displayed are not current. This occurs when you first open the dialog or if you change the selection. To update the values, press the *Scan Levels* button.

When previewing a *Normalize*, the entire file must be scanned even when previewing a small selection. Pressing the *Scan Levels* button stores the current *Peak* and *RMS* values. This allows you to preview different *Normalize to* level settings without re-

scanning the entire file.

If the RMS level never reaches the *Ignore below* threshold, a value of -96 dB will be displayed. You must then lower the *Ignore below* threshold level before processing.

Use Current Scan Levels (do not scan selection) (On/Off)

When this option is turned on, no scan is performed. Instead, the current scan levels shown are used (which persist when you close the dialog). This is useful if you want to use the scan levels from a different selection or file on the current selection. This allows the same gain to be applied to different files (the gain used is based on the *Scan Level* and the *Normalize to* setting).

One possible use is to scan a small selection of a track that has the loudest material or most constant levels and then using that scan to *Normalize* the entire file or different tracks.

If Clipping Occurs

These options allow you to choose how you want the *Normalize* function to handle any clipping that could occur as the sound file is processed when using the *RMS* option.

Apply Dynamic Compression

With this option selected, any peaks that would clip are limited to below 0 dB using non-zero attack and release times to minimize distortion. In other words, a time-varying gain is used to ensure that no hard clipping occurs.

This option is useful for getting very loud (it is common knowledge that louder is better, right?), yet clear sound during the mastering process.

Also, you achieve best results when converting 16-bit audio to 8-bit or other compressed formats if the audio has been maximized before the conversions occur.

Normalize Peak Value to 0 dB

With this option selected, the selection's peak amplitude level is normalized to 0 dB. This applies the maximum possible constant gain which doesn't clip to the selection. Less gain is applied than would be necessary to achieve the *Normalize to RMS* level.

Ignore (saturate)

With this option selected any sound data that would clip is allowed to clip. This may not be a problem if the clipping samples are very short and infrequent, but will sound horrendous if much clipping occurs.

Stop Processing

With this option selected any sound data that would clip cause the *Normalize* function to stop processing and display a message to that effect.

Note: When using *Normalize* on stereo data, if the selection includes both channels, normalization is computed on the loudest sample value found in either channel and the same gain is applied to both. If a single channel is selected in a stereo file then *Normalization* will effect only that channel.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Pan/Expand

[Video tutorial](#)

The *Pan/Expand* window allows you to draw a pan, stereo expansion, or MS mix envelope which will be applied across the current data selection. You can create up to 16 envelope points.

Process Mode

Pan (preserve stereo separation)

Allows you to perform right and left channel panning effects without mixing the channels together. This is used to simulate left/right positioning of stereo recordings.

Pan (mix channels before panning)

Allows you to perform right and left channel panning effects by first mixing the two channels and then changing the volume between channels.

Stereo Expand

In this mode, you can contract or expand the stereo image of a stereo recording, from dead center (mono), to completely panned wide (no center channel). For realistic effects, you should not venture too far from *Normal*, but do experiment.

To move both endpoints at once, you can select all points by pressing *Control+A* while focus is on the envelope. Dragging one of the points then moves all of the points.

This effect works best with true stereo recordings and will not work with mono material.

Mix Mid-Side (MS) Recording to Left and Right Channels

Mid-Side recording (MS) is a microphone technique where one mic is used to record the center, or mid, channel (pointed directly towards the source) and the other mic records the stereo image (pointed 90 degrees away from the source). For proper playback on most systems, MS recordings must be converted to your standard left/right (also called AB) track.

In Sound Forge, to convert an MS-recorded track to a left/right track, you must first ensure that the *Mid track* is in the left track and the *Side track* on the right. The MS mix function then allows you to set the width of the stereo image for the converted track.

Output Gain

The gain applied to the signal after processing.

Show Wave

Press this check box to draw the current selection's waveform on the envelope graph. This is done automatically if the selection is small. For stereo files, you also have the option of displaying the left, right, or a mix of the two channels.

See also

[Video tutorial](#)

[Process Menu](#)

[Sound Processing and Other Tools](#)

Resample

[Video tutorial](#)

Use this function to change the sampling rate of an existing sound file.

New Sample Rate (2,000 to 60,000 Hz)

Sample rate to which the sound file will be converted.

Apply an Anti-alias Filter During Resample (On/Off)

Apply an anti-aliasing low-pass filter to the sound file before resampling when changing to a lower sampling rate. Since the maximum frequency that can be represented by a particular sample rate is one-half of the sampling rate (the Nyquist frequency), high frequencies in a sound file cannot be represented if the sound file is resampled to a lower sampling rate. Therefore, when changing to a lower sampling rate, if the sound file has a strong high-frequency content, anti-aliasing should be used to prevent these high frequencies from becoming low-frequency distortion.

Set the Sample Rate Only (do not resample) (On/Off)

If this option is checked the playback rate is changed without resampling the data. This means that the original pitch of the file is not preserved. When converting between two very close sample rates and speed is of the essence, this option can be turned on.

Resampling data consists of changing the number of recorded samples per second. When changing to a higher sample rate, extra samples are interpolated and the file size increases. When changing to a lower sample rate, some samples are removed and the file size decreases.

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Reverse

Use this command to reverse a selection.

[Video tutorial](#)

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Smooth/Enhance

[Video tutorial](#)

Use this command to add or remove high frequency content in a sound file selection.

The *Smooth* function will smooth out fast-changing transients in a sound and is useful for removing glitches.

The *Enhance* function boosts the very highest frequencies (close to the Nyquist rate) in the sound file, making the sound file sound more vivid. It is useful for compensating for the effects of downsampling or for bringing out very fast transients in a sound file

Operation

(-5 to 5)

Determines the amount of smoothing (high frequency attenuation) or enhancement (high frequency boost) is applied to the sound file.

For a very drastic high frequency boost or cut, repeat the effect 2 or 3 times with the amount set to either limit.

Note that since the enhance function boosts frequencies near the Nyquist rate, the current sample rate will determine what frequencies will be affected. At 44,100 Hz, the effect can be very subtle (unless you repeat it a few times until your ears ring).

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Time Compress/Expand

[Video tutorial](#)

The *Time Compress/Expand* function changes the duration of a sound file without altering the pitch.

Mode

Selects the algorithm used to process the sound. The modes have been labeled for suggested uses, but do not assume that one mode will not work well with a certain type of sound. The A, B, and C modes use different algorithms, with the numbers indicating different internal parameters. Depending on the source sound, different modes can vary the resulting quality drastically, so experimenting with all modes is suggested.

You should be able to achieve excellent results for ratios between 75% and 115%. Beyond this range, you will start to hear artifacts such as echoes, flanging, or drop-outs. The different modes are supplied so that you can determine what works best for your case.

In many cases, many of the modes will work well, but in some instances (such as complex, multi-instrument music or pitched drums) you'll have to try out all of the modes. In general, the settings designed for music will work with the largest variety of material.

Also, running the process a number of times using small increments (like 105%) will create different effects that processing all at once with a large time change.

Final Length

(50% to 500% of original length)

Desired length of the selection. Note that only the A modes allow for expansion above 150%.

Blend Edges with Adjoining Data (On\Off)

When this option is turned on, the beginning and end of the selections will be crossfaded with the original sound data. This is used to minimize glitching.

To change the *Original tempo* displayed at the bottom of the dialog, you must go to *the Edit Tempo* dialog (Special menu).

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

[Changing Pitch and Time Duration](#)

Volume

[Video tutorial](#)

Use this command to change the volume of a selection.

Gain (-Inf. to 20 dB)

Select a *Gain* amount from -Inf. to +20 dB. A value of negative infinity corresponds to a mute (0 %). Negative decibel values lower the volume (attenuation). Positive values are used to raise the volume (boost).

See also

[Process Menu](#)

[Sound Processing and Other Tools](#)

Effects Menu

[ACM Filter](#)

[Amplitude Modulation](#)

[Chorus](#)

[Delay/Echo|Simple](#)

[Delay/Echo|Multi-Tap](#)

[Distortion](#)

[Dynamics|Graphic](#)

[Dynamics|Multi-Band](#)

[Envelope](#)

[Flange](#)

[Gapper/Snipper](#)

[Noise Gate](#)

[Pitch|Bend](#)

[Pitch|Shift](#)

[Reverb](#)

[Vibrato](#)

ACM Filter

Sound Forge has the ability to use *Audio Compression Manager (ACM)* filters to process sound data. No ACM filters come installed with Sound Forge, but ACM filters could potentially be obtained from various manufacturers. The word filter in this case does not necessarily mean sound frequency filtering. ACM filters can perform most DSP-related functions like delay, volume, reverb, and time change.

This menu item appears only if any ACM filters are installed in your computer. No ACM filters are included and none are available from Sonic Foundry, so don't think you're missing out on anything. Several ACM filters are included with Windows 95 and Windows NT 3.51

Filter

Any installed ACM filters will be displayed here.

Attributes

If the selected filter has predefined parameters, they will be displayed here.

Because Microsoft fully documents the *Audio Compression Manager Filter* architecture, anyone who wishes to write a DSP function for use with Sound Forge may do so. However, writing ACM filters is not for faint at heart; it requires a strong computer programming background in C/C++ as well as knowledge of audio-related DSP.

The *Audio Compression Manager Development Kit* can be obtained from Microsoft (please do not call Sonic Foundry). It is available on the *Video for Windows 1.1 Development Kit* as well as the *Developer Network Platform Level 2*.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Installing the ACM](#)

[The Audio Compression Manager](#)

Amplitude Modulation

[Video tutorial](#)

The *Amplitude Modulation* effect applies 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.

Dry Out (0-100%)

Amount of unprocessed signal mixed into the output.

Modulated Out (0-100%)

Amount of processed signal mixed into the output.

Frequency (0.1-5,000 Hz)

Frequency of the gain waveform to be applied to the input signal. To achieve a slow tremolo, use a low frequency (Period = 1/Frequency). With faster frequencies, the amplitude modulation is audible not as a change in amplitude, but as additional frequency side bands.

Shape (Sine or Square)

Shape of the gain waveform. Sine wave amplitude modulation is smooth, while square wave modulation creates drastic on/off changes.

Stereo Pan (On/Off)

When on, the two channels' amplitude gain envelopes are out of phase. This creates the effect of panning back and forth between the two channels. Panning is only available for stereo files.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

Chorus

[Video tutorial](#)

The *Chorus* effect creates the illusion of two or more sound sources playing together. This is done by adding 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.

Input Gain (–Inf. to 0 dB)

Gain applied to the signal before processing.

Dry Out (–Inf. to 0 dB)

Amount of unprocessed signal mixed into the output.

Chorus Out (–Inf. to 0 dB)

Amount of processed signal mixed into the output.

Chorus Size (1 to 3)

Number of times the selection is processed with the chorus algorithm. Larger numbers create a fuller sound.

Feedback (0 to 100%)

Percentage of the processed signal which is fed back for re-processing.

Invert the Chorus Phase (On/Off)

When this option is checked, the processed output will be inverted (180 degree phase) before mixing with the *Dry out*. This can sometimes have a large effect on the final sound.

Chorus Out Delay (0.1 to 100 ms)

Time, in milliseconds, that the chorused signal is delayed from the input signal.

Modulation Rate (0.1 to 20.0 Hz)

Frequency, in Hertz, of the modulating signal.

Modulation Depth (1 to 100%)

Amount of frequency modulation applied to the signal.

Attenuate Frequencies Above (100 to 10,000 Hz/Off)

Cut-off frequency, in Hertz, of the low-pass filter used on the processed signal. Setting to *off* means no low-pass filtering occurs.

A wide variety of non-chorus-like effects can be created with this function. If the modulation depth is high, a vibrato effect will occur. If the delay is small, flanging occurs.
--

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Using Chorus](#)

Simple Delay/Echo

[Video tutorial](#)

The Delay/Echo effect adds a delayed copy of the sound signal to the file. A single delay or decaying multiple delays can be added before or after the sound signal.

Dry Out (-Inf. to 0 dB)

Amount of unprocessed signal mixed into the output.

Delay Out (-Inf. to 0 dB)

Amount of delayed signal mixed into the output.

Multiple Echo (On/Off)

When Off, a single delayed copy is added to the file.

When On, multiple delayed copies are added by using feedback (delaying the delayed signal, and so on...), which creates a decaying echo effect.

Pre-Delay/Echo (ON/OFF)

When Off, the delayed copy is added after the sound, as heard in everyday life echoes.

When On, the delayed copy is added before the sound, as heard in the twilight zone.

Delay Time (0.0000 - 5.0000 Seconds)

Determines the time after or before which the delayed copy of the sound is added to the file. Also, if Multiple Echo is turned On, subsequent echoes occur at intervals specified by the Delay Time.

Decay Time (0.000 - 20.000 Seconds)

Determines the time it takes the multiple echoes to become nearly inaudible. Technically speaking, this is the time it takes an echo to decrease in amplitude by a factor of 30 dB.

If the Decay Time is less than the Delay Time, no echo will be created.

Tips

You can also use the *Delay/Echo* function to shift one channel of a stereo file backward or forward in time. For example, if you have a file with music in the right channel and voice in the left, make a selection in only the left channel around the voice. Set the following parameters: *Dry out* level at 0%, *Delay out* 100%, no *Multiple delays/echoes*, no *Pre-delay/echo*. You can then use the *Delay time* parameter to determine the time shift. If you want to shift the vocals backwards in time, check the *Pre-delay* check box.

To simulate a stereo sound from a mono file, copy the sound to both left and right channels of a stereo file and then do a very small time delay of a single channel (*Dry out* level at 0%, *Delay out* level at 100%, no *Multiple delays/echoes*, no *Pre-delay/echo*). This simple technique often works in making the sound have a more realistic stereo image. When a sound wave comes from your side, your right and left ears receive sounds at different times, which is what this effect simulates.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Using Delay](#)

Multi-Tap Delay

[Video tutorial](#)

The *Multi-Tap Delay* effect is used to simulate multiple echoes, reverberation and other delay-related effects. It accomplishes this with multiple delay taps, pitch modulation, and filtering.

You can use up to eight independent delay taps. Each delay tap consists of a delay time and amplitude specification for a particular echo of the original sound. These are used to simulate the early reflection echoes which are the result of the first sound reflections in a room. Feedback, in combination with filtering and modulation, is then used to simulate the thousands of other reflections that combine to form what we hear as reverberation.

Echogram

The *Echogram* represents the amplitude of sound reflections over time (impulse response) as determined by the current settings. Each vertical line represents an echo of the original sound. The length of each line corresponds to its amplitude (as a percentage of the original sound), while its horizontal distance from the left edge represents the time elapsed after the original sound. The red line is the currently selected tap, blue lines are the other active taps, and black lines are echoes resulting from feedback.

The *Echogram* can be used to get an estimate of the overall decay time. The right-most horizontal point corresponds to a delay time equal to the graph resolution. This value can be changed between 500, 1,000, 3,000, and 5,000 milliseconds.

Input Gain (-Inf. to 0 dB)

Gain applied to the signal before processing.

Dry Out (-Inf. to 0 dB)

Amount of unprocessed signal mixed into the output.

Wet Out (-Inf. to 0 dB)

Amount of processed signal mixed into the output.

Modulator Rate (0.1-10 Hz)

Frequency, in hertz, of the modulating signal. 0.1 Hz is slow while 10 Hz is very fast modulation.

Modulator Depth (0-100%)

Amount of frequency modulation applied to the reverberation. Zero signifies no modulation, and 100% is a large amount of modulation. Modulation is used to create a fuller sound or chorusing effects.

Feedback (0-150%)

Percentage of the processed signal which is fed back for re-processing and thus creating more echoes. Large feedback values allow for longer decay times but can cause ringing.

Low-pass Start Frequency (100-10,000 Hz/ Off)

Cut-off frequency, in Hertz, of the low-pass filter (high frequency damping) used on the processed signal. Setting to *Off* means no low-pass filtering occurs. A high frequency setting (or *Off*) will result in live room reverberation, while lower settings will deaden the sound by simulating the high-frequency damping which many materials impart on sound.

Most building materials and air itself absorb more high frequencies than low frequencies, so using a low-pass filter is often necessary to prevent the reverberation from sounding too live.

Number of Taps (1-8)

Number of delay taps used to create the reverberation.

Current Tap (1-8)

Use to select the delay tap to be modified.

For each active tap there is a corresponding:

Delay (1-2500 ms)

Delay time of the tap. Since sound travels close to 1 foot per millisecond, you can simulate a given room size by setting delays to match reflections from all walls in the room. In a large room, sound must travel further to reach the reflection surfaces. Therefore, setting the echoes further apart creates a sense of being in a larger space.

Gain (-100 to 100%)

Loudness level of the echo.

Pan (-100 to 100%)

Channel placement of the echo, with -100% being hard left and 100% being hard right. Panning echoes to different channels creates the illusion of reverberation coming from all directions in stereo files.

Graph Resolution (Delay ms) (500, 1000, 3000, 5000)

Determines the maximum time in milliseconds displayed on the graph.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Using Multi-tap delay](#)

Distortion

[Video tutorial](#)

The *Distortion* effect can simulate the overloading of an amplifier or mangle a waveform into submission.

Dry Out (-Inf. to 0 dB)

Amount of unprocessed signal mixed into the output.

Distorted Out (-Inf. to 40 dB)

Amount of distorted signal mixed into the output.

Threshold Level (-Inf. to 0 dB)

Signals above this level will be distorted.

Clamp level (-Inf. to 0 dB)

Signals over the *Threshold level* are set or clamped to this level. For normal limiting, set the *Clamp level* to be equal to the *Threshold level*. Setting this level anywhere else will make your speakers do things they've never done before.

Curvature (-Inf. to 0 dB)

Determines the curvature of the limiting above the *Threshold level*. Minus infinity corresponds to hard clipping (no curvature) while zero decibels corresponds to a very smooth curve, with little distortion.

Hard Clip for Maximum Speaker Thrashing (On/Off)

When *On*, signals over the *Threshold level* will be set to the *Clamp level*. When *Off*, signals over the *Threshold level* will be soft clipped according to the *Curvature*.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

Graphic Dynamics

[Video tutorial](#)

Compression, expansion, and limiting affect the dynamic range of the sound file by varying the gain as a function of the input sound level.

A compressor lowers the dynamic range of a signal by reducing the level of high volume signals and applying an overall gain to raise the level up again. Compression is often used in vocals and music to get even volume levels which often results in a smoother, fuller sound.

A limiter is a compressor with a greater ratio of dynamic range reduction. Limiters are used in broadcasting for making sure that the volume level does not go over a certain threshold which overdrives electrical equipment. Limiters are also used by guitarists to create a distorted, sustaining tone.

An expander increases the dynamic range of a signal by decreasing the level of low volume signals. Expanders are commonly used for noise reduction or for emphasizing the dynamics of a piece.

Dynamics Input/Output Graph

This graph maps each possible input level to an output level. The *No Gain* diagonal line shows where the input and output are equal. When the envelope is below the *No Gain* line, signals of the corresponding input level are attenuated (gain is less than 0). On the other hand, when the envelope is above the *No Gain* line, the signal is boosted (gain is greater than 0).

With this in mind, it is possible to draw graphs to generate compression, limiting, noise gating, and expansion effects by moving the envelope points manually or by using the *Threshold* and *Ratio* controls.

Output Gain (-25 to 25 dB)

The gain applied to the signal after processing. If *Auto Gain compensate* is turned on, the output gain is still applied afterwards.

Reset Graph

Press this to reset all envelope points.

Sync Stereo Gain (On/Off)

When on, the gain of the left and right channel will always be the same. This prevents a loss of stereo imaging that can occur if both channels were to be processed independently.

Auto Gain Compensate (On/Off)

When on, a gain is applied during processing to try to keep the maximum input and output levels constant. In general, this gain will be equal to the highest envelope point in the graph. You can still use the *Output gain* control to fine tune the overall gain.

Attack Time (0 to 500 ms)

Time, in milliseconds, required for the gain to change from a pre-threshold level to a post-threshold level.

Release Time (0 to 2,000 ms)

Time, in milliseconds, required for the gain to change from a post-threshold level to a pre-threshold level.

Threshold (-80 to -0.1 dB)

Signal level which differentiates the *Attack* and *Release* modes. In a simple compressor or gate, the threshold is the level at which the gain begins to deviate from 1.

When changing the threshold, all points slide diagonal up and down the *No Gain* line.

Ratio (1.0:1 to Inf.:1)

Ratio of input to output levels above the *Threshold*. This *Ratio* determines how much a signal above the *Threshold* will be boosted or faded as a function of its level.

When changing the *Ratio*, all points above the *Threshold* move closer or further away from the *No Gain* line.

For best results, very low frequencies (below 20 Hz) present in a sound file should be removed before applying the *Graphic* or *Multi-Band Dynamics* functions. This makes it easier to monitor audible frequencies while compressing.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Using Compression and Limiting](#)

[Using Expansion](#)

Multi-Band Dynamics

[Video tutorial](#)

The *Multi-Band Dynamics* dialog allows you to apply compression and limiting to up to four different frequency bands. Band limiting is used to keep a set of frequencies from being too loud. The most common use for band limiting is the de-esser. In this case, the frequencies associated with sibilant sounds (S's) are limited to a set level, without affecting other frequencies.

Because there are so many different parameters available, getting the best results requires a bit of experimentation. For most situations, you will only need to use a single band. Each band can either be a low-shelf, band-notch, or high-shelf. After you've selected the frequency range that will be affected, you must set a compression *Amount*. A value less than 10:1 is considered compression while greater values will create harsher limiting.

The *Threshold* level determines at what point will compression begin. You can find your required *Threshold* level by first setting it a high level (such as -6 dB) and slowly lowering it while previewing until you can hear the band-limiting effect.

You can also get an estimate of the *Threshold* level necessary by first selecting the portion of sound which contains the offending frequencies, bring up the dialog, and after selecting the frequency range pressing the *Get Threshold* button. The *Threshold* setting will change to 6 dB below the detected peak of the current frequency band. You'll probably need to then lower it slightly to increase the effect.

To reset all bands, right-click anywhere on the dialog and select *Reset All*. This can be used in most function dialogs.

Graph

The *Multi-Band Dynamics Graph* gives you an idea of where the active bands are located in the frequency spectrum and the relative threshold and amount levels. The horizontal lines indicate the threshold settings of the active bands, the current band being red. The amount of compression is indicated by how far down the dark gray curve goes down. Therefore, the amount of dark gray is representative of the amount of gain reduction that can occur.

It is recommended that the bands not overlap. When the bands overlap the results are not easily visualized.

Output Gain (- 25 to 25 dB)

The gain applied to the signal after processing.

Sync Stereo Channels (On/Off)

When on, the gain of the left and right channel will always be the same. This prevents a loss of stereo imaging that can occur if both channels were to be processed independently.

Number of bands (1 to 4)

Determines the number of active, independent bands. For most cases, only a single band is necessary.

Current band (1 to number of bands)

Since each band has an independent mode, threshold, amount, attack and release times, you must use this control to switch between all active bands.

Threshold (-Inf. to 0 dB)

When a frequency band level is above the threshold, it will be compressed. A lower threshold will cause more of the signal to be compressed. If the threshold is too low, too much of the sound data will be filtered. If its too high, not enough processing will occur.

Get Threshold

When you press this button, the current selection is analyzed to get an estimate of the necessary threshold. After analyzing, the *Threshold* will be set to 6 dB below the peak level found for the current band.

Amount (1:1 to 50:1)

Compression ratio applied to a band when it is above the threshold level. A high compression ratio means more gain reduction will be applied.

Attack Time (0 to 500 ms)

Time, in milliseconds, required for the gain to change from a pre-threshold level to a post-threshold level.

Release Time (0 to 2,000 ms)

Time, in milliseconds, required for the gain to change from a post-threshold level to a pre-threshold level.

Type

Determines the frequency band type.

Low-shelf

Frequencies below the cut-off frequency will be compressed.

Band-notch

Frequencies surrounding the center frequency will be compressed.

High-shelf

Frequencies above the cut-off frequency will be compressed.

Gain (-25 to 25 dB)

Gain applied to a band after it has been compressed. This is used for compensating for the gain lost from the compression.

Width (0.3 to 3.0 octaves)

Determines the frequency range which will be compressed when using *Band-notch* mode.

Frequency (Below, Center, Above) (50 to 15,000 Hz)

If the band mode is a low- or high-shelf, this parameter determines the cut-off frequency for which the filter starts processing. Otherwise, it is the center frequency of the band.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Using Compression and Limiting](#)

[Using Expansion](#)

Envelope

[Video tutorial](#)

With the *Envelope* effect, the amplitude envelope of a waveform can be forced to match a specified envelope shape. Unlike the Graphic Fade, 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.

Envelope Graph

(2 to 16 points, 0 to 100%)

Specifies the amplitude envelope over time. Click the left mouse button to define a point and drag to change its position. Click the right mouse button over an existing point to remove the point. You can create up to 16 envelope points.

Reset Envelope

Press this button to reset the graph to a simple ADSR (Attack, Decay, Sustain, Release) curve.

Smooth Gain

(On/Off)

to Minimize Distortion on Steep Slopes

When on, prevents the gain from changing too fast, which might result in unwanted distortion. Also, when this option is on the gain will always begin at 0%.

Show Wave

Select this check box to draw the waveform of the current selection on the envelope graph. This is done automatically if the selection is small. For stereo files, you also have the option of displaying the left, right, or a mix of the two channels.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

Flange/Wah-Wah

[Video tutorial](#)

Flange

The flanging effect is heard in many 60s and 70s recordings. It is the result of mixing a modulated delay signal with the original signal to create a sweeping sound much like an airplane taking off.

Phaser

A phase is similar to a flange, except that instead of using a simple delay, frequencies are phase-shifted over time. When used on stereo files, strange effects on the stereo image are created.

Wah-Wah

The wah-wah effect was popularized by many a rock/blues guitarists favorite toy, the wah-wah pedal. It consists of a band-pass filter which attenuates low and high frequencies in varying amounts over time.

Dry Out **(-Inf. to 12 dB)**

Amount of unprocessed signal mixed into the output. A negative percentage means that the signal is also inverted.

Wet Out **(-Inf. to 12 dB)**

Amount of processed signal mixed into the output. A negative percentage means that the signal is also inverted.

Rate **(0.1 to 20 Hz)**

Frequency, in Hertz, of the modulating signal, which determines the period of the flange. Low frequencies create a slow, sweeping flange while higher frequencies create fast flanging.

Depth **(1 to 100%)**

In the *Flange*, this determines the amount of frequency modulation applied to the signal.

In the *Phaser*, this determines the range of frequencies which experience phase shifting.

In the *Wah-Wah*, this determines the sweep range of the band-pass filter.

Center Frequency **(100 to 5,000 Hz)**

Frequency around which the sweep of the *Phaser* and *Wah-Wah* is centered.

Resonance **(1 to 100%)**

Amount of band-pass filtering (in *Wah-Wah*) or phase shifting (in *Phaser*) which takes place. When the resonance is high, the range of frequencies processed is smaller, yet more pronounced.

Flange, Phaser, and Wah-Wah are best heard with long, sustaining sounds so that the entire sweep can be heard.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Using Flange](#)

Gapper/Snipper

[Video tutorial](#)

The Gapper/Snipper cuts chunks from the sample or inserts silence in the sample periodically at a set frequency to create unusual (if not useful) effects.

Insert/Cut

In Cut mode, chunks are removed from the sample. In Insert mode, chunks of silence are inserted into the sample.

Frequency (0.1-1,000 Hz)

Times per second a chunk is inserted or cut from the sample.

Size (1-1,000 ms)

Size of the chunk to be inserted or cut.

Fade (0-50%)

Percentage of the chunk not cut or between inserts that is faded in and out. Zero percent corresponds to no fade, while 50% is maximum fade. Use fading to prevent glitches.

Tips

This process changes the size of the original file. Be aware of the final size indicated, since it is easy to set the parameters to create a very large output.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

Noise Gate

[Video tutorial](#)

A noise gate removes signals below a set threshold. It is used to remove noise from silent breaks in a sample.

Threshold (0 to -40 dB)

Signals below this level will be removed from the sample. Zero dB is a very high level, while -70 dB is very low. Noise levels are usually around -40 dB, but will vary widely. To get an estimate of the noise level, select a region of silence (which also contains noise) and run the [Statistics](#) tool. The maximum value (positive or negative) shown is a good estimate of the threshold noise level required. For example, if the maximum positive value is -40 dB (1.0%) and the maximum negative value is -36 dB (-1.5%), use a number a bit higher than -36 dB (maybe -32 dB) as the noise threshold level.

Attack Time (1-500 ms)

Time it takes the gain of the gate to change from zero to one once the level rises above the threshold. The attack time is best kept to a minimum if percussive attacks are to be preserved. Higher attack times make sounds slowly swell up in volume.

Release Time (1-2000 ms)

Time it takes the gain of the gate to change from one to zero once the level falls below the threshold. The release time is usually kept high to allow for natural sounding decays, otherwise long decays will be cut off.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Using Noise Gate](#)

Pitch Bend

[Video tutorial](#)

Pitch Bend is used to draw an envelope which corresponds to increasing or decreasing the pitch of a sample over time.

Pitch Bend

Pitch Bend is used to draw an envelope which corresponds to increasing or decreasing the pitch of a sound file over time.

Envelope

(2 to 16 points, -Range to +Range)

Specifies *Pitch Bend* over time. The vertical axis represents pitch change, with zero being no change. The horizontal axis represents the length of the selected region.

Click the left mouse button to define a point and drag to change its position. Click the right mouse button over an existing point to remove the point. You can create up to 16 envelope points.

Reset Envelope

This button removes all points except the outermost two.

Range

(-24 to 24 Semitones)

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.

Show Wave

Select this check box to draw the waveform of the current selection on the envelope graph. This is done automatically if the selection is small. For stereo files, you also have the option of displaying the left, right, or a mix of the two channels.

<p><i>Pitch Bend</i> is accomplished in the same manner as changing the playback speed on a tape deck. Therefore, the length of the file will be changed.</p>

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

Pitch Shift

[Video tutorial](#)

This effect changes the pitch of a sample with or without preserving the duration of the sample.

Semitones To Shift Pitch By (-50 to 50)

Number of semitones (half-steps) to raise or lower the pitch. When using *Preserve duration*, the range is limited to -12 to 12 semitones.

Cents To Shift Pitch By (-100.0 to 100.0)

Fine tune above and below a semitone. One hundred cents is equal to one semitone.

Apply An Anti-alias Filter During Pitch Shift (On/Off)

Changing the pitch of a file requires a form of resampling. When shifting by large amounts, an anti-alias filter will minimize distortions due to aliasing.

Preserve Duration (On/Off)

If on, the sound file length will remain the same; otherwise, the length of the selection will change by 1/transposition ratio.

Mode

The *Mode* determines what algorithm is used to do the pitch shifting. There are three different algorithms, A, B, and C, each with different modes tuned for different sound material. It is best to experiment with different modes since the names for these modes are only suggestions.

Blend Edges with Adjoining Data (On/Off)

When this option is turned on, the beginning and end of the selections will be crossfaded with the original sound data. This is used to minimize glitching.

Transposition Ratio

Fraction by which all frequencies will be multiplied. If the *Preserve duration* check box is off, the length of the file will be multiplied by 1/transposition ratio.

Pitch change without preserving duration is accomplished in the same manner as changing the playback speed on a tape deck. Therefore, the length of the file will be changed. A pitch change can be used to accurately change the time duration of a selection if the change in pitch is not audible or important. This is useful if a very small change in time duration is necessary, for example, to correct for playback/record speed discrepancies on a tape deck.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Changing Pitch and Time Duration](#)

Reverb

[Video tutorial](#)

Reverb is an effect to simulate various acoustic spaces. It consists of early reflections, which are the first reflections that arrive back to your ear, and the reverb itself.

Reverberation Mode

The *Reverberation mode* affects the quality of the reverberation tail. Each mode has slightly a different frequency response, ringiness, echo density, and delay time to simulate different environments.

Dry Out (-Inf. to 0 dB)

Amount of unprocessed signal mixed into the output.

Reverb Out (-Inf. to 0 dB)

Amount of processed reverberated signal mixed into the output.

Early Out (-Inf. to 0 dB)

Amount of processed early reflections mixed into the output.

Early Reflections Style

The *Early reflections style* affects the first few echoes heard immediately following the dry source and have a strong influence on the size of the perceived space.

Decay Time (0.5 to 5.0 seconds)

This is the time it takes for the reverb to decrease in level by 60 dB. Or, in other words, this is how long your reverb will last.

Pre-delay (Off to 200 ms)

This is the time it takes from the initial sound until the onset of reverb. This setting helps control the perceived size of the room. A longer *Pre-delay* time will imply a larger room.

Attenuate Bass Frequencies Below (10 to 500 Hz)

Use this option to remove low frequencies in the reverberation. Too much low bass in a reverb will result in a muddy sound.

Attenuate High Frequencies Above (500 to 15,000 Hz)

Selecting high frequencies filters the reverb above the set frequency. This setting has a lot to do with the perceived liveliness of the room. In a very lively room, high frequencies are not attenuated when they bounce off of the walls.

Any generated reverberation will not extend the sound file. To ensure that the reverberation tail does not get truncated at the end of the sound file, there should be enough silence following the last sound. You can extend the file by using *Insert Silence* (Process menu) at the end of the file before applying *Reverb*.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

[Using Reverb](#)

Vibrato

[Video tutorial](#)

Vibrato is the periodic pitch modulation often used by vocalists and instrumentalists. Very fast pitch modulation creates wacky FM effects unlike any vibrato.

Modulation Rate (0.1 to 1,000 Hz)

Frequency, in Hertz, of the modulating signal. Determines the period of the vibrato.

Low frequencies (0.1 to 2 Hz) create a slow vibrato while higher frequencies (2 Hz to 15 Hz) create fast pitch bends. Even higher frequencies (15 Hz and up) modulate the selection so fast that instead of hearing pitch changes, new sideband frequencies are heard.

Modulation Depth (0.1 to 100.0%)

Amount of pitch change applied to the selection.

Trim the End if a Change of Duration Occurs (On/Off)

When on, if the output selection length is shorter than the input length, the selection end is trimmed. This is used to prevent glitches when using a slow vibrato inside a selection.

See also

[Effects Menu](#)

[Sound Processing and Other Tools](#)

Tools Menu

[Auto Region](#)

[Crossfade Loop](#)

[Sampler](#)

[Search](#)

[DTMF Synthesis](#)

[Frequency Modulation Synthesis](#)

[Simple Synthesis](#)

[Statistics](#)

Auto Region

[Video tutorial](#)

The *Auto Region* function automatically creates regions in a sound file for the *Regions List* and *Playlist*. These regions are defined according to fast sound attacks (like drum beats or words) or a selected musical time interval.

Attack Sensitivity (1 to 100%)

Determines 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.

Release Sensitivity (0.1 to 100.0%)

Determines the minimum decrease in sound level which must occur before a region end is created. High values indicate that the sound level does not have to decrease very much before a region end is created, and therefore more regions will be found. When set low, the sound level must decrease to a greater extent before a region end is found. A low setting is useful if you only want a region to be created after very silent breaks.

Minimum Level (0.1 to 100.0%)

Threshold sound level which 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 only the loud instrument attacks in a song (like the bass drum) to be at 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.

Minimum Beat Duration (.01 to 3.00 Seconds)

Specifies the minimum length in seconds 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 fast tempo music. A higher setting will prevent quick sound attacks from being separated into different regions.

Use Release Point for End of Region (On/Off)

When on, a region will end when the sound level drops by a factor determined by the *Release sensitivity*. This is useful if you don't want the silence in between sounds or phrases to be included in the regions.

When off, region ends are only created when attacks are detected.

Build Regions Using the Current Tempo (On/Off)

When on, regions are created according to the current *Beats per minute* (see [Edit Tempo](#) function under the Special menu to change this value) Otherwise, regions are created by using an algorithm that finds sound level changes to determine where a sound attack or release occurs.

Measures, Beats (0 to 100)

Determines the musical time interval between regions according to the specified number of beats in a measure. For example, if you want a region to be created at every beat, set *Beats* to 1 and *Measures* to 0. To have a region created at every measure, set *Measures* to 1 and *Beats* to 0. If you wanted to have a section at every 7 beats, you could set *Beats* to 7 and *Measures* to 0, or set *Beats* to 3 and *Measures* to 1, if you have defined a measure to be 4 beats. You can set the number of beats per measure using the *Edit Tempo* command.

When using the attack-detection algorithm, more beats will be detected if you raise the *Attack sensitivity*, raise the *Release sensitivity*, lower the *Minimum level*, or lower the *Minimum beat duration*. The opposite is true for detecting fewer beats.

See also

[Tools Menu](#)

[Sound Processing and Other Tools](#)

[Creative Uses for Sound Forge](#)

Crossfade Loop

[Video tutorial](#)

The *Crossfade Loop* function is designed to help create loops in material that otherwise might not be loopable. It does so by mixing sound occurring before the loop start point into the end of the loop. This can make the transition from the end to the beginning of the loop much smoother.

You may also wish to crossfade the region following the loop (*Crossfade Post-Loop*) with the beginning of the loop to make the transition from looping to playing the post loop smoother. Often times this is done to compensate for a crossfade inside the loop.

Tip: Although it depends on your application and personal preferences, most musicians agree that the *Crossfade Loop* function is a last resort. It is suggested that the *Loop Tuner* be used before applying the *Crossfade Loop* function to get the loop ends to match as well as possible.

Graph

The *Crossfade Loop* graph shows you where in the selection the crossfade will occur. The three areas (from left to right) which represent sound before, inside, and following the loop points, are shown with different sloped lines. For example, when doing a *Sustaining* loop crossfade with no *Post-Loop*, you can see the loop data fade out as data from the *Pre-Loop* is mixed in to form a crossfade.

Loop to Crossfade

Selects the sample loop that will be affected by the crossfade. Most samples only have one loop which is the *Sustaining* loop. If a sample contains two loops, the second loop is called the *Release* loop.

Loop (1 to 100 %)

Determines how much of the selected sample loop will be crossfaded. When checked, the end of the sample loop is crossfaded with the end of the region before the loop start. This is used for making the end of the loop blend smoothly with the beginning of the loop.

Post-Loop (1 to 100 %)

When checked, the region following the loop end is crossfaded with the beginning of the sample loop. This is used for making the transition from the end of the loop to the rest of the sample following the loop smoother (i.e. when done looping and playback continues).

Post-Loop crossfading is only necessary if the sampler is every going to play beyond the loop points.

Preview Mode

These modes determine how the *Preview* button will work.

Loop the Loop

Plays the loop area only.

Play Loop through Post-Loop

Plays the loop once and then the *Post-Loop*.

Play as One Shot

Plays from the beginning of the file and uses the loop settings to determine how many times to loop each loop.

See also

[Tools Menu](#)

[Sound Processing and Other Tools](#)

Statistics

[Video tutorial](#)

This tool displays statistical information about the selected sound file region.

Cursor Position

This indicates the cursor position in samples from the start of the sound file.

Sample Value at Cursor

This value represents the instantaneous amplitude at this position.

For 16-bit sound files, the range of values is -32,768 to 32,767. For 8-bit sound files, the range of values is -128 to 127.

Maximum/Minimum Position and Sample Value

Maximum and minimum sample values and the location, in samples, where they occur.

This is useful for determining if any clipping occurs in the sound file. It can also be used to determine the noise level of a signal for use on the *Noise Gate* effect. For example, to find the noise amplitude, run the *Statistics* function on a region of noisy silence.

RMS Power

Root Mean Square of the sample values relative to the RMS value of a maximum-amplitude square wave (the loudest possible recording).

When used on short intervals relates to the volume level of the sound file. However, if used on a large selection with large volume variation, this value becomes less meaningful. For another way to measure loudness, use the *Scan Levels* button in the *Normalize* dialog.

The calculation used in Sound Forge 4.0 is slightly different than in Sound Forge 3.0. The value of 0 dB corresponds to a maximum amplitude square, not sine wave. A maximum value sine wave will read at -3.0 dB.

Average Value (DC Offset)

This is a sum of all of the sample values in the selected region divided by the number of samples. If non-zero, this usually indicates that there is a DC offset in the recording process.

Zero Crossings

Times per second that the waveform changes from a negative value to a positive value. This value can be used as a rough estimate of the frequency of the sound data for very simple waveforms.

See also

[Tools Menu](#)

[Sound Processing and Other Tools](#)

DTMF Tones Synthesis

[Video tutorial](#)

Use this to generate the standard dial tones used by the telephone companies.

Dial String

Use this to enter the phone number to be generated along with any necessary pause characters. Unknown characters will be ignored.

Amplitude

(-Inf. to 0 dB)

Peak level of the waveform.

Tone Style to Generate

DTMF

Dual Tone Multi-Frequency signals are used by the standard push-button phones. It uses combinations of 679, 770, 852, 941, 1209, 1336, 1477, and 1633 Hz sine waves.

MF

CCITT R1 signals are used internally by the telephone networks. They are generated by a combination of 700, 900, 1100, 1300, 1500, and 1700 Hz sine waves.

Single Tone Length

(0.001 to 2 seconds)

Output length of individual tones.

Break Length

(0.001 to 2 seconds)

Length of the silence between tones.

Pause Length

(0.001 to 5 seconds)

Length of silence to insert when the pause character is found.

Fade The Edges of Each Tone

When on, the tones are faded in and out a few milliseconds to prevent glitching.

Pause Character

Character which when present in the dial string will cause a pause between successive tones.

Insert New Tone Sequence At

(Cursor, Start of file, End of file,)

Point where the generated waveform will be placed within the existing sound file.

See also

[Tools Menu](#)

FM Synthesis

[Video tutorial](#)

FM Synthesis uses frequency modulation (FM) to create complex sounds from simple waveforms. Frequency modulation is the varying of the frequency of a waveform (the carrier) as a function of the output amplitude of another waveform (the modulator). 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 fast that many additional frequencies, or sidebands, are created.

In Sound Forge, 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.

Total Output Waveform Length (0.001 to 60.000 seconds)

Output length of the generated waveform.

Current Operator (1 to Number of Operators)

Determines the active operator being modified.

Envelope Graph

Amplitude envelope of the active operator over time. Click the left mouse button to define a point and drag to change its position. Click the right mouse button over an existing point to remove the point. You can create up to 8 envelope points for each operator.

Reset

The *Reset* envelope button removes all points except the outermost two.

Amplitude (-Inf. to 0 dB)

Output gain of operator after amplitude envelope. 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, very harsh sounds can result.

Operator Shape (Sine, Square, Saw, Triangle, Noise, Absolute Sine)

Shape of a single period of the waveform.

Frequency (0.01 to Sampling Rate/2)

Frequency of the waveform. This parameter does not affect the noise waveform.

If the frequency is set to 0.00, a DC (zero frequency) waveform is produced regardless of the waveform specified.

When using *Noise*, the frequency determines the high frequency content of the noise. For full bandwidth (white) noise, set the frequency to its highest possible settings.

Feedback (0 to 100%)

Amount of the output of the operator which 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.

Insert Waveform At (Cursor, Start of file, End of file)

Point where the generated waveform will be placed within the existing sound file.

Configuration Figure

Determines the arrangement and number of operators to be used.

The output of operators joined horizontally are simply mixed together. The output of the bottom operators are mixed together to form the final output. Mixing together different simple waveforms is called additive synthesis.

Two operators joined vertically are FM carrier-modulator pairs. Operators which have another operator stacked above them are carriers, with the operator directly above being the modulator.

Operators which have no other operator directly above are simple waveform generators.

When three or more operators are stacked, this means that the highest operator modulates the operator below it, which then in turn modulates the following operator below, and so on.

Start out with the simplest, single operator configuration and try out all the waveforms at different frequencies, with the feedback factor set at zero. Experiment using feedback on each waveform to hear the different effects of self-modulation. Move to the next configuration, with two unmodulated operators mixed together (horizontal connection) to hear two operators simultaneously (additive synthesis).

Change the configuration to two stacked operators (vertical connection) and again experiment using different waveforms and frequencies on the carrier (bottom operator) and the modulator (top operator). With the modulator frequency set low (1 to 5 Hz), lower the modulator amplitude to create slight detuning. Raising it creates big pitch bends.

If the modulator frequency is set high, many unusual sounds can be achieved.

Setting the modulator frequency to 0.00 (DC) allows for direct control of the carrier's pitch by using the modulator's amplitude envelope.

Adding more operators increases the complexity of the waveform. Both FM and additive synthesis can be combined to generate an endless variety of sounds.

See also

[Tools Menu](#)

[Sound Processing and Other Tools](#)

[Using Synthesis](#)

[MIDI Notes and Frequencies](#)

Simple Synthesis

[Video tutorial](#)

Simple Synthesis can be used to generate a simple waveform of a given shape, pitch, and length.

Amplitude **(-Inf. to 0 dB)**

Peak level of the waveform to be generated. When generating *Noise*, the amplitude is affected by the cutoff frequency selected.

Waveform Shape (Sine, Square, Saw, Triangle, Noise, Absolute Sine)

Shape of a single period of the waveform.

Length **(0.001 to 60.0 seconds)**

Output length of the generated waveform.

Frequency **(0.01 Hz to Sampling Rate/2)**

Frequency of the waveform.

When creating noise, the frequency specifies the low-pass cut-off frequency. For white noise, the frequency must be set to its maximum available setting.

Note that aliasing occurs with many of these waveforms when using high frequencies because they are not band-limited.

Insert New Waveform At **(Cursor, Start of file, End of file)**

Point where the generated waveform will be placed within the existing sound file.

See also

[Tools Menu](#)

[Sound Processing and Other Tools](#)

[Using Synthesis](#)

[MIDI Notes and Frequencies](#)

Find

[Video tutorial](#)

The *Find* tool searches for clicks and pops, volume levels, or silence breaks in a sound signal.

Glitch

Clicks and pops are unwanted sounds often found in vinyl recordings or from bad splice-editing, and are seen as sharp glitches in the normal curve of a waveform.

The *Glitch* search algorithm looks at a waveform, starting at the cursor position, moving forward in time, and determines when a glitch occurs depending on the threshold and sensitivity settings. Then, the cursor position is moved to the detected glitch to allow for further editing. In *Pencil Tool* mode (Edit menu), you can draw a smooth curve over the glitch to remove it.

Threshold Slope (-60 to 0 dB)

Minimum slope (steepness) of the glitch. A high setting will only detect glitches with a high slope, while a low setting will detect both low and high-sloped glitches.

Sensitivity (0 to 100)

With a high setting, the glitch finder will detect as a glitch any part of the waveform with a slope above the *Threshold slope*.

A lower setting forces the *Glitch* finder to further verify that the high-sloped waveform is in fact a glitch and not part of the original, smooth waveform.

If no glitches are detected, but you can hear clicks or pops, try lowering the *Threshold slope* and/or raising the *Sensitivity*.

If the *Glitch* search algorithm is falsely detecting glitches, lower the *Sensitivity* and/or raise the *Threshold slope*.

To find the next glitch, you can use the *Control+Y* shortcut key to repeat the last operation.

Level Equal To or Above

Use this to find the next point in time after the cursor position when the waveform reaches a particular level.

Threshold Level (-96 to -6 dB)

Determines the sound level to search for.

End of Silent Region

Use this to move the cursor to the end of the next silence break.

Threshold Level (-96 to -10 dB)

Determines the silence sound level. The *Silence search* will move the cursor to the next position where the level goes below and then back above the silence threshold level.

Largest Peak

Use this to move the cursor to the next largest peak amplitude.

Searches progress from the current cursor position forward, not the entire file. To find the largest peak in the entire file, set the cursor to the beginning of the sound file or select all before processing.

See also

[Tools Menu](#)

[Sound Processing and Other Tools](#)

Sampler

[Video tutorial](#)

The **Sampler** dialog (**Tools** menu) allows you to send and receive data to and from a sampler. To set up sampler/computer communications, press the **Configure** button.

Configuration

This drop-down list contains Preset Sampler setups (saved in the Sampler Configuration dialog). Once you save different configurations, the settings can be conveniently selected from this list.

Logical send/receive sample number (0 - 16,383)

This is the number your sampler uses as its location reference for samples sent or received. This number can be biased for specific samplers with the Sampler Bias option in the Sampler Configuration dialog.

Actual send sample number

This text field contains the actual sample number to be used by your sampler to store the sample. This number is based on the Logical sample number combined with the Sample Bias for sending, set in the Sampler Configuration dialog.

Actual receive sample number

This text field contains the actual sample number to be retrieved from your sampler into the current data window. This number is based on the Logical sample number combined with the Sample Bias for receiving set in the Sampler Configuration dialog.

Send Sample

Press this button to begin transmitting the active Data Window's sample to the sampler.

Get Sample

Press this button to begin receiving a sample from the sampler. A new window is created prior to reading the data from the sampler.

Configure

Opens the Sampler Configuration dialog where you can determine the settings for communications with your samplers.

Sampler Configuration Dialog

The Sampler Configuration dialog is used to create and save sampler communications configurations.

Name

The name of the sampler configuration that is saved as a Preset.

Send to

Lists supported samplers and methods of communication for sending samples. If your sampler is not listed here, try Generic MIDI/SDS or Generic SCSI/SMDI.

MIDI output (MIDI/SDS)

Select the MIDI interface connected to the input of the sampler to which you wish to send samples.

MIDI input (MIDI/SDS)

Select the MIDI interface connected to the output of the sampler to which you wish to send samples.

SCSI Host (SCSI/SMDI)

Select the SCSI card used to send data between the computer and sampler.

SCSI Sampler (SCSI/SMDI)

Select the SCSI compatible sampler which will receive the data.

Sample Bias (-16,383 - 16,383)

This is a numbering offset that allows you to add or subtract a value to the Logical sample number to compensate for differing sampler storage schemes. It is often easiest to set the Sample Bias so a Logical sample number of zero corresponds to the first available sample storage number in your sampler.

MIDI Channel (MIDI/SDS) (1-16)

The MIDI channel the sample data will be sent on when using SDS.

Wait for request (MIDI/SDS)

Selecting this option tells the Sampler Tool to wait for a request (handshake) from the sampler before beginning an SDS sample transfer.

Open loop (MIDI/SDS)

Selecting this option will cause the Sampler Tool to send SDS sample data immediately upon clicking the **Send Sample** button. This is an unconditional transfer of sample data (no handshake). When using Open Loop for send, the sampler's MIDI Out does not have to be connected to your computer's MIDI In. However, *open loop decreases sample dump performance..*

Receive from

Lists supported samplers and methods of communication for receiving samples. If your sampler is not listed here, try Generic MIDI/SDS or Generic SCSI/SMDI.

MIDI output (MIDI/SDS)

Select the MIDI interface connected to the input of the sampler from which you wish to receive samples.

MIDI input (MIDI/SDS)

Select the MIDI interface connected to the output of the sampler from which you wish to receive samples.

SCSI Host (SCSI/SMDI)

Select the SCSI card used to receive data from your sampler.

SCSI Sampler (SCSI/SMDI)

Select the SCSI compatible sampler which will send the data.

Sample Bias (-16,383 - 16,383)

This is a numbering offset that allows you to add or subtract a value to the Logical sample number to compensate for differing sampler storage schemes. It is often easiest to set the Sample Bias so a Logical sample number of zero corresponds to the first available sample storage number in your sampler.

MIDI Channel (MIDI/SDS) (1-16)

Select the MIDI channel on which the sample data will be received when using SDS.

Send request (MIDI/SDS)

Selecting this option tells the Sampler Tool to send a request to your sampler asking for a sample when you press the **Get Sample** button. If you turn this option off, you will always have to initiate transfers to the Sampler Tool from your sampler after pressing the **Get Sample** button. In other words, you will have to press a button on your sampler to start the actual transfer.

Open loop (MIDI/SDS)

Selecting this option will cause the Sampler Tool to wait for sample data after pressing the **Get Sample** button. You must initiate all sample transfers from your sampler. This is an unconditional transfer of sample data (no handshake). When using Open Loop for receive, the sampler's MIDI In does not have to be connected to your computer's MIDI Out. However, *open loop decreases sample dump performance.*

See also

[Tools Menu](#)

[Using the Sampler Tool](#)

Appendix

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Sound File Formats

The file formats recognized by Sound Forge are listed below along with the extension used to identify them to the user. An arbitrary extension has been assigned for each of the formats used by computers other than the PC. Many of these formats when found on another type of computer will tend to use the extension .SND. Thus, even if a file has the .SND extension it may not be a Sounder/ SoundTool file.

Ad Lib Sample (.SMP)

This format is used by the Ad Lib Gold card, for its sampled instruments. It supports, 8-/16- bit, stereo/mono, and 4-bit Yamaha ADPCM compression.

Amiga SVX File (.SVX/.IFF)

This file type is found on the Commodore Amiga and is similar in substance to the Microsoft .RIFF Wave format.

Covox 8-Bit File (.V8)

This format is used with Covox software. It is an 8-bit mono uncompressed file format.

Creative Labs VOC File (.VOC)

This is one of the more commonly found sample sound formats found on PC-compatible computers. The .VOC format supports packed data which Sound Forge will unpack prior to importing the file. The .VOC format also supports information for silence, looping, and varying sample rates. When Sound Forge imports a .VOC file it uses the first sample rate found and ignores such information as looping and silence. Sound Forge will save .VOC files as unpacked data with none of the additional information such as looping or silence. Sound Forge gives you the ability to save .VOC files in either the old or new .VOC standard. The new .VOC standard supports 16-bit data. Refer to the *Preferences* section in the Reference chapter for information on configuring Sound Forge to always use the new .VOC standard.

Dialogic VOX File (.VOX)

This format is used with specialized voice data boards. It is a 4-bit mono ADPCM file format which expands to 16-bit data.

☞ Indicates this file format supports the saving of sample looping information.

Gravis Patch File ☞ (.PAT)

This format is used by the Gravis UltraSound Card. Sound Forge can only read and write Gravis Patch files which have single samples stored in them. Patch files with multiple samples are not recognized. Sound Forge recognizes Gravis Patch files which are of version 1.10. Older versions, such as 1.00, are not supported.

Intervoice (.IVC)

This file format is for use with Intervoice telephony systems. It is a mono format with a variety of compression schemes available.

Macintosh AIFF ☞ (.AIF/ .SND)

This format is used on the Apple Macintosh to save sound data files. An .AIFF file is best when transferring files between the PC and the Mac using a network, since most network software will delete any resource information in a Macintosh file. Sound Forge can also read an .AIFF with a Mac binary header attached but will identify the file as a Macintosh Resource instead.

Macintosh Resource (.SND)

This format is used on the Apple Macintosh to save sound resources. When reading a file of this type it must have a Mac-Binary header. This header is attached to binary files transferred with common Macintosh file transfer programs. Sound Forge does not save to this file format.

MIDI SDS File ☞ (.SDS)

This format is a file which contains the exact data that would be transferred to a sampler when using the MIDI Sample Dump Standard (SDS) protocol. It is a series of MIDI SysEx messages. MIDI SDS supports mono files only.

NeXT / Sun (Java) File (.AU/.SND)

This is the standard format of sound files found on the NeXT and Sun Sparc station computers. This format has a large number of sub types of file data. Sound Forge reads and writes the most common data formats for these files, 8-bit linear, 16-bit linear, and u-Law.

RealAudio (.RA)

This format is a compressed format designed for real time streaming over the Internet. It is a mono format and can be compressed for 14.4 and 28.8 kilobaud transfers.

SampleVision File ☺ **(.SMP)**

This format is used with Turtle Beach's SampleVision software. It is a 16-bit mono file format. The .SMP format is meant to be used with sampling keyboards.

☺ Indicates this file format supports the saving of sample looping information.

Sonic Foundry Sample Resource **(.SFR)**

This format is an old format defined by Sonic Foundry and is supplied for compatibility. It is not recommended for use in saving files.

Sound Designer 1 ☺ **(.DIG/ .SD)**

This format is used by Sound Designer 1.

Sounder/SoundTool File **(.SND)**

This format is used with the shareware applications Sounder and Sound Tool. It is very popular on bulletin boards and various online services.

Video for Windows **(.AVI)**

This is a Microsoft digital video file format. It supports multiple channel audio, but Sound Forge will only allow you to open one audio and one video stream at a time. When saving as an .AVI, many compression schemes are available for both the video and audio streams.

Wave (Microsoft) File ☺ **(.WAV)**

This is the format for sampled sounds defined by Microsoft for use with Windows. It is an expandable format which supports multiple data formats and compression schemes.

Raw Files **(.*)**

Raw files can be composed of any sound data. If the data is any form of 8- or 16-bit PCM (uncompressed) or G.711 A-Law or u-Law (a form of companded compression), then Sound Forge can open and save the file.

☺ Indicates this file format supports the saving of sample looping information.

Summary Information Fields

IARL Archival Location

Indicates where the subject of the file is archived.

IART Artist

Lists the artist of the original subject of the file. For example, "Bach."

ICMS Commissioned

Lists the name of the person or organization that commissioned the subject of the file. For example, "XYZZY Records."

ICMT Comments

Provides 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 newline characters.

ICOP Copyright

Records the copyright information for the file. For example, "© Copyright Sonic Foundry, Inc. 1996." If there are multiple copyrights, separate them by a semicolon followed by a space.

ICRD Creation Date

Specifies 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

Specifies the size of the original subject of the file. For example, "8.5 in h, 11 in w."

IDPI Dots Per Inch

Stores dots per inch setting of the digitizer used to produce the file, such as "300."

IENG Engineer

Stores the name of the engineer who worked on the file. If there are multiple engineers, separate the names by a semicolon and a blank. For example, "Sayre, Jack; Feith, John."

IGNR Genre

Describes the original work, such as, "landscape," "portrait," "still life," etc.

IKEY Keywords

Provides a list of keywords that refer to the file or subject of the file. Separate multiple keywords with a semicolon and a blank. For example, "Madison; aerial view; scenery."

ILGT Lightness

Describes the changes in lightness settings on the digitizer required to produce the file. Note that the format of this information depends on hardware used.

IMED Medium

Describes the original subject of the file, such as, "computer image," "drawing," "lithograph," and so forth.

INAM Name

Stores the title of the subject of the file, such as, "Madison From Above."

IPLT Palette Setting

Specifies the number of colors requested when digitizing an image, such as "256."

IPRD Product

Specifies 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."

ISFT Software

Identifies the name of the software package used to create the file, such as "Sound Forge."

ISHP Sharpness

Identifies the changes in sharpness for the digitizer required to produce the file (the format depends on

the hardware used).

ISRC Source

Identifies the name of the person or organization who supplied the original subject of the file. For example, "Sonic Foundry, Inc."

ISRF Source Form

Identifies 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

Identifies the technician who digitized the file. For example, "Palmer, Curt."

Sound Forge and the Microsoft Audio Compression Manager

Sound Forge for 32 Bit Windows Note: The Microsoft Audio Compression Manager (ACM) is currently not available under **Win32s**. This is a limitation of Win32s, not Sound Forge for 32 Bit Windows.

The Microsoft Audio Compression Manager (more commonly referred to as the ACM) is a standard interface for Audio Compression and Filtering in Windows and Windows NT. This interface allows applications like Sound Forge to use compression and filtering algorithms provided by other companies. Sound Forge fully supports audio compression and filtering through the ACM. This enables you to use any ACM compatible compression and filter driver with Sound Forge. The best part of this support is you don't have to learn anything new to use it! Sound Forge transparently opens compressed WAV files and provides all available compression formats for WAV files in the Save As dialog. There are no silly Import and Export commands.

There are roughly three pieces to the ACM (though these pieces are tightly coupled). It is not necessary for you to know any details about these pieces to use the ACM from Sound Forge, so this information is supplied for those who want to know more. That being said, the three pieces of the ACM are:

[Audio data compression and decompression](#)

[Audio data filtering](#)

[Transparent Playback and Recording of non-hardware supported audio files](#)

See also

[Setting up the ACM](#)

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:

1. - 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 entirely dependent on the algorithm.
2. - 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. The result is opening and saving compressed files will usually take longer than uncompressed files.
3. - Compressed files are not as portable as uncompressed files. If you are distributing .WAV files in a compressed format, you must make sure 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 inconvenient.

In Sound Forge, 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 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 will always save changes to the file using the selected compression algorithm; you do not need to re-select the compression format each time you save. Of course you can change the compression format at any time with the *Save As* dialog. You can also revert to an uncompressed format.

Experimentation is probably the most important step in deciding whether or not to use audio compression for your projects.

Audio data filtering (Digital Signal Processing)

The second piece of the *ACM* provides access to *Digital Signal Processing (DSP)* algorithms. The *ACM* refers to these DSP algorithms as Filter drivers. Though the term Filter in this context is more generic than just frequency filtering (like *Parametric* or *Graphic EQ*). An *ACM* filter can implement almost any type of audio *DSP* algorithm. Reverb, distortion, and time change are just a few of several possibilities.

Sound Forge does not come with any *ACM* filters. You must obtain *ACM* filters from other sources. However, since the *ACM* filter architecture is fully documented by Microsoft, it is possible that many types of filters will become available on BBBs and other places for use with Sound Forge. If any *ACM* filters are installed in your computer, you will see *the ACM Filter* menu item under the Tools menu.

Sound Forge allows you to use any *ACM* filter installed and enabled on your computer to process sound files. Simply select the *ACM* filter option under the Tools menu to choose the filter and its attributes that you wish to apply to your sound data. The *ACM Filter* tool works exactly like all other Sound Forge *DSP* functions including the option to save presets and set custom selections.

Transparent Playback and Recording of non-hardware supported audio files

The third part of the *ACM* is probably the most significant (or at least the coolest). It is called the *Sound Mapper* and allows audio data formats that are not directly supported by your sound card to be played and recorded. Sound Forge lets you use the *Sound Mapper* by selecting it for *Playback* and *Record* in the *Wave* folder of the *Preferences* dialog. Additional configuration of the *Sound Mapper* for your system can be done using the *Windows Control Panel*.

If the *ACM* is not installed on your computer, the *Sound Mapper* option will be called the *Wave Mapper* instead. The *Wave Mapper* and *Sound Mapper* can be thought of as the same device (to a computer program, they are one and the same). However, the *Sound Mapper*, which requires the *ACM* to be installed, is much more powerful than the *Wave Mapper*. Keep this in mind when you hear people talk about the *Wave Mapper* and *Sound Mapper*.

Lets go through some examples of what the *Sound Mapper* can do. If you have a sound file that is recorded at a bizarre sample rate like 22,257 Hz (perhaps a file that came from an external sampler), 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.

The *Sound Mapper* can play this sound file correctly without forcing you to resample the file first. The *Sound Mapper* will map the sound to the best format possible and perform the resampling in real time. Yes Virginia, there is a Santa Claus!

Other examples include the ability to play 16-bit sounds on an 8-bit-only sound card. Or playing stereo sounds on a mono-only sound card. And believe it or not, you can record stereo files on a mono-only sound card. However, this last example is essentially useless because the *Sound Mapper* will simply duplicate the mono input of your sound card in both channels. So keep in mind that the *Sound Mapper* cannot create something that is better than what the sound card can supply.

The usefulness of the *Sound Mapper* for non-compressed sound files can be summarized as convenience. You dont have to convert the sound to a supported format before you listen to it. Some of the tasks that the *Sound Mapper* can perform (like format conversions while recording) are available not because they are useful in real world situations, but because the developers of the *ACM* knew it could be done. Much like climbing a mountain for no better reason than its there.

So what about compressed sound files? Yes, the *Sound Mapper* allows you to play (and sometimes record) compressed sound files. Even on sound cards that do not support compression directly. So you can play a sound file that is compressed with Microsoft ADPCM or The DSP Groups TrueSpeech on any sound card without decompressing the sound file first. Many multimedia and entertainment software titles use compressed sound files to save disk space. The software then plays the files through the *Sound Mapper*.

We stated above that the *Sound Mapper* can sometimes record compressed sound files. The reason for not always being able to record compressed files (even when the compressed file can be played) is simple. Compressing sound data can be very computationally expensive (it takes a long time); the amount of time required is completely dependent on the compression algorithm and how it is implemented. Decompressing sound data is almost always faster than compressing the same sound data which makes it possible to play it back in real time.

It should be noted that Sound Forge does not play and record compressed sound files directly. Rather, all compression and decompression is performed while opening and saving the sound files. This limitation is actually insignificant; Sound Forge is the perfect tool for authoring compressed sound files for the *ACM* and many software authors are using Sound Forge for exactly this. Sound Forge saves compressed sound files using the best quality possible something that cannot always be done in real time. So saving compressed sound files with Sound Forge will usually sound better than those recorded with audio compression.

Notes on Sound Forges sound file compression support:

- The *Open* dialog allows you to preview compressed .WAV files if you have an appropriate ACM driver installed. However, you must have your *Playback* device set to the *Sound Mapper* for this to work (*Preferences|Wave* in the Options menu).
- When saving uncompressed audio data to a compressed format with the *Save As* dialog, it is a good idea to close the file and re-open it after saving. Since Sound Forge does the compression and decompression during saving and loading you will not be able to hear what the file sounds like with compression until after youve saved and reloaded the file.

Setting up the Microsoft Audio Compression Manager

The Microsoft Audio Compression Manager is part of Windows 95 and NT. If you are running Windows 3.x, Sound Forge will install *Video for Windows* in your computer, which also installs the ACM.

In Windows 95, to view the installed audio compression and decompression drivers, go to *Control Panel\Multimedia\Advanced\Audio Compression Codecs*. Double-clicking on any of the codecs allows you to enable or disable each one.

The Sonic Foundry Virtual MIDI Router (VMR)

The *Sonic Foundry Virtual MIDI Router (VMR)* is a software-only driver that allows you to control Sound Forge from other MIDI-capable applications and vice versa under Windows 95 and Windows 3.x without using MIDI hardware.

A MIDI Router simply transfers MIDI data from one port to another. The *Sonic Foundry VMR* driver does exactly this but requires no hardware. The driver provides up to four devices for MIDI output and input. Each of the output devices sends all MIDI data to its corresponding input device.

If one application sends MIDI data through the #3 *Sonic Foundry MIDI Router output* device, another application can receive this MIDI data as input from the #3 *Sonic Foundry MIDI Router input* device.

The first character of the device name is a device number from 1 to 4. The device number is placed at the beginning of the name for compatibility with sequencer software that tries to display device names in very small places.
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The *Sonic Foundry VMR* can be very useful for synchronizing two MIDI capable applications. You can trigger Sound Forge from your MIDI sequencer, or you can drive your MIDI sequencer from Sound Forge using SMPTE/MTC. This and more can be accomplished without using MIDI hardware for routing.

See also

[Installing the Sonic Foundry Virtual MIDI Router](#)

[Configuring the Sonic Foundry Virtual MIDI Router](#)

Installing the Sonic Foundry Virtual MIDI Router

Note: A *Virtual MIDI Router* is applicable to Windows 95 and Windows 3.x only, not Windows NT.

The *Sonic Foundry Virtual MIDI Router* is included on the Sound Forge CD. To install the *Sonic Foundry Virtual MIDI Router*, follow the instructions below for the operating system you are running.

Windows 95

1. Open *Control Panel* from the *Settings* menu in the *Start* menu.
2. Open the *Add New Hardware Item* in the *Control Panel*.
3. Click the *Next>* button in the first page of the *Add New Hardware Wizard*.
4. In the *Detect New Hardware* page of the *Wizard* set the *Detect New Hardware option* to *No*.
5. Click the *Next>* button.
6. In the *Select type of hardware* page select *Sound, video, and game controllers*.
7. Click the *Next>* button.
8. In the *Manufacturer and model* page click the *Have Disk...* button.
9. Insert the Sound Forge CD into your CD-ROM drive.
10. In the path list box, type in *E:\extras\sfvmid* (when E is your CD-ROM drive) and click *OK*.
11. Select the *Sonic Foundry Virtual MIDI Router* from the models list and click *OK*.
12. Click the *Finish* button in the final *Add New Hardware* page.
13. Once you have restarted Windows, all programs that support MIDI (including Sound Forge) can use the *Virtual MIDI Router*.

Windows 3.x

1. Open *Control Panel* and double-click the *Drivers* icon.
2. Click the *Add* button.
3. Select *Unlisted/Updated* driver and click *OK*.
4. You can now insert the Sound Forge CD. Click the *Browse* button and select the drive in which you inserted the CD. Double click the *Extras* directory, then the *Win16* directory, then the *VMR* directory. Click *OK*.
5. The driver should now be listed as *Sonic Foundry Virtual MIDI Router*. Click once on the driver to select it and click *OK*.
6. Windows will now install the driver. When it is finished you will be prompted to restart Windows to complete the driver installation. There are no further setup procedures.
7. Once you have restarted Windows, all programs that support MIDI (including Sound Forge) can use the *Virtual MIDI Router*.

Configuring the Sonic Foundry Virtual MIDI Router

After installing the *Sonic Foundry Virtual MIDI Router (VMR)*, a new MIDI device is created with the name *Sonic Foundry MIDI Router*. If you want to install additional devices for MIDI routing (the *VMR* supports up to four) you need to configure the *VMR*.

Note: When only one routing device is used, the name of the driver is displayed as *Sonic Foundry MIDI Router*. When multiple ports are used each port is labeled starting with a number (1 - the number of ports) to identify each connection.

The following procedures describe configuring the number of *VMR* devices for Windows 95 and Windows 3.x.

Windows 95

Warning: If you have a large number of MIDI devices in your system, or a multi-port MIDI board such as an eight port MIDI interface, you may have problems installing additional *VMR* MIDI interface ports. Windows 95 will crash on startup if more than 11 MIDI devices are installed. If you have trouble with MIDI devices causing system errors after installing the *VMR*, you should either reduce the number of ports in use by the *VMR*, remove the *VMR*, or reduce the number of MIDI devices in your system.

1. Double-click on *My Computer*.
2. Open the *Control Panel* folder.
3. Double click on the *Multimedia* icon.
4. Select the *Advanced Properties* page.
5. Expand the *MIDI Devices and Instruments* list by selecting it and double-clicking.
6. Select the *Sonic Foundry MIDI Router* device and click the *Properties* button.
7. In the *General* page of the *Properties* for the *Sonic Foundry MIDI Router* click the *Settings* button.
8. In the *Configure Sonic Foundry VMR* dialog select the number of ports you would like to enable from the *Ports* combo list.
9. Click *OK* in the dialog.
10. Click *OK* in the *Sonic Foundry MIDI Router Properties* dialog.
11. Click *OK* in the *Multimedia Properties* dialog.
12. Restart Windows for the new number of devices to take effect

Windows 3.x

1. Open the *Program Manager* group main.
2. Double-click on the *Control Panel* icon to open the *Control Panel*.
3. Double click on the *Drivers* icon to open the *Drivers* dialog.
4. Select the *Sonic Foundry Virtual MIDI Router* from the list of displayed drivers.
5. Click the *Setup* button.
6. In the *Configure Sonic Foundry VMR* dialog, select the number of ports you would like to enable from the *Ports* combo list.
7. Click *OK* in the dialog.
8. Press the *Restart Now* button to restart Windows and to have the settings take effect.

SMPTE Time Code

The Society of Motion Picture and Television Engineers (SMPTE) time code may be one of the most misunderstood concepts among individuals within the music industry. After working with SMPTE time code for years, many people are still confused by the concept, so don't feel bad if you haven't got it all figured out. Hopefully this discussion will clear the mud.

The biggest problem with SMPTE time code is that, depending on whether you sit on the video or audio side of the fence, SMPTE time codes may mean different things to you. When dealing with SMPTE you will probably see five, perhaps six, different types of time codes formats (six is for the people who are really confused).

Here is a description of each SMPTE time code format:

SMPTE 25 EBU

This SMPTE code runs at 25 frames per second and is also known as SMPTE EBU (European Broadcasting Union). The reason for having this rate is that European television systems run at exactly 25 frames per second.

SMPTE 24 Film Sync

This SMPTE code runs at 24 frames per second and is also known as SMPTE Film Sync. This rate matches a nominal film rate of 24 frames per second (the slowest speed possible for apparent continuous motion).

OK, those two are easy. Now things start to get a little crazy.

SMPTE 30 Non-Drop (as used in the audio world)

In the US, the 60 Hz power system makes it easy to generate a time code rate of 30 frames per second. This rate is commonly used in audio environments and is typically known as 30 Non-Drop. You will probably use this rate when synchronizing audio applications like a multi-track recorder or your MIDI sequencer.

If all you care about is working with audio and not dealing with video, stop reading right here. We mean it! All you really need to know is that there are three different SMPTE rates you might want to use: SMPTE 24, SMPTE 25, and SMPTE 30 Non-Drop. However, be aware that SMPTE 30 Non-Drop in the video world runs at 29.97 frames per second.

True SMPTE 30 Drop and SMPTE 30 Non-Drop (as used in the video world)

If you are planning to work with video, the frame rate of exactly 30 frames per second is never used. When NTSC color systems were developed, the frame rate was changed by a tiny amount to eliminate the possibility of crosstalk between the audio and color information. Even though it is still referred to as SMPTE 30 Drop or Non-Drop, the actual frame rate that is used is exactly 29.97 frames per second. This poses a problem since this small difference will cause SMPTE time and real time (what your clock reads) to be different over long periods. Because of this, two methods are used to generate SMPTE time code in the video world: Drop and Non-Drop.

In SMPTE Non-Drop, the time code frames are always incremented by one in exact synchronization to the frames of your video. However, since the video actually plays at only 29.97 frames per second (rather than 30 frames per second), SMPTE time will increment at a slower rate than real world time. This will lead to a SMPTE time versus real time discrepancy. Thus, after a while, we could look at the clock on the wall and notice it is farther ahead than the SMPTE time displayed in our application.

SMPTE Drop time code (which also runs at 29.97 frames per second) attempts to compensate for the discrepancy between real world time and SMPTE time by dropping frames from the sequence of SMPTE frames in order to catch up with real world time. What this means is that occasionally in the SMPTE sequence of time, the SMPTE time will jump forward by more than one frame. The time is adjusted forward by two frames on every minute boundary except 00, 10, 20, 30, 40, and 50. Thus when SMPTE Drop time increments from 00:00:59:29, the next value will be 00:01:00:02 in SMPTE Drop rather than 00:01:00:00 in SMPTE Non-Drop. In SMPTE Drop, it must be remembered that certain codes no longer exist. For instance, there is no such time as 00:01:00:00 in SMPTE Drop. The time code is actually 00:01:00:02.

When synchronizing audio to video, it is crucial that the SMPTE time code (30 Drop or Non-Drop) used in your sequencer or digital audio workstation is the same as the SMPTE time code striped onto the video. Only then will the SMPTE times on the video screen and computer monitor match exactly during playback.

In the audio world, people have started to call 30 Non-Drop (which runs at 29.97 frames per second) 29.97-Non-Drop to distinguish it from the 30 Non-Drop used between audio applications (which runs at a true 30 frames per second). SMPTE 30 Drop (as used in video) may also be referred to as SMPTE 29.97 Drop just to reiterate that the frame rate is actually 29.97 frames per second. It just depends on who you talk to.

However, you must remember that there is no difference between 30 Drop and 29.97 Drop time code. There are those who have tried to say that there is such a thing as a SMPTE time code which actually runs at 30 frames per second *and* generates drop frames, but such a time code would not show real world time and would also be incompatible with the 29.97 NTSC frame, so its use is unclear.

How Sound Forge deals with the issue

For SMPTE 24, SMPTE 25, and SMPTE 30 Drop time codes, Sound Forge behaves as you would expect. The 24, 25 and 30 Drop frame rates are calculated and generated at exactly 24, 25, and 29.97 frames per second respectively. In 30 Drop rate, the dropped frames are inserted to make up for the time discrepancies with real world time.

For SMPTE 30 Non-Drop, the default method for Sound Forge is to calculate all SMPTE values at exactly 30 frames per second. If you are working with video and need SMPTE Non-Drop running at 29.97 frames per second, you can turn off the **Use 30 frames per second for SMPTE Non-Drop** option in the **MIDI/Sync Preferences** folder. This will cause Sound Forge to use 29.97 frames per second instead of 30 frames per second for SMPTE 30 Non-Drop.

Some calculations for your benefit:

SMPTE 30 fr = 33.3333 ms per frame; SMPTE 29.97 fr = 33.3667 ms per frame

1 minute = 60 seconds * 30 = 1,800 frames per minute

1 minute = 60 seconds * 29.97 = 1,798.2 frames per minute

Difference per minute = 1.8 frames.

1 hour = 3,600 seconds * 30 = 108,000 frames per hour

1 hour = 3,600 seconds * 29.97 = 107,892 frames per hour

Difference per hour = 108 frames.

Thus in drop frame we need to catch up by 108 frames per hour

For example, in Non-Drop mode the SMPTE time code of 01:00:00:00 will equate to a real world time of 01:00:03:18. Therefore, there is a lag behind real world time of 3 seconds and 18 frames after 1 hour.

Here is a nifty chart to show how SMPTE Drop works. As time progresses, two frames are dropped each minute except at the 00, 10, 20, 30, 40, and 50 minute points. Notice that after one hour, 108 frames have been dropped. This number corresponds to the number of drop frames we calculated above needed to match SMPTE 30 (29.97 fps) time to real time.

Number of dropped frames versus minutes elapsed over one hour

Minutes	00	10	20	30	40	50
	xx:00:00:00	xx:10:00:00	xx:20:00:00	xx:30:00:00	xx:40:00:00	xx:50:00:00
0	0 frames	18 frames	36 frames	54 frames	72 frames	90 frames
1	2	20	38	56	74	92
2	4	22	40	58	76	94
3	6	24	42	60	78	96
4	8	26	44	62	80	98
5	10	28	46	64	82	100
6	12	30	48	66	84	102
7	14	32	50	68	86	104
8	16	34	52	70	88	106
9	18	36	54	72	90	108

General MIDI Patch Map

1-8

PIANO

1	Acoustic Grand Piano
2	Bright Acoustic Piano
3	Electric Grand Piano
4	Honky-tonk Piano
5	Electric Piano 1
6	Electric Piano 2
7	Harpsichord
8	Clav

25-32

GUITAR

25	Acoustic Guitar (nylon)
26	Acoustic Guitar (steel)
27	Electric Guitar (jazz)
28	Electric Guitar (clean)
29	Electric Guitar (muted)
30	Overdriven Guitar
31	Distortion Guitar
32	Guitar Harmonics

49-56

ENSEMBLE

49	String Ensemble 1
50	String Ensemble 2
51	Synth Strings 1
52	Synth Strings 2
53	Choir Aahs
54	Voice Oohs
55	Synth Voice
56	Orchestra Hit

73-80

PIPE

73	Piccolo
74	Flute
75	Recorder
76	Pan Flute
77	Blown Bottle
78	Shakuhachi
79	Whistle
80	Ocarina

97-104

SYNTH EFFECTS

97	FX 1 (rain)
98	FX 2 (soundtrack)
99	FX 3 (crystal)
100	FX 4 (atmosphere)
101	FX 5 (brightness)

9-16

CHROM PERCUSSION

9	Celesta
10	Glockenspiel
11	Music Box
12	Vibraphone
13	Marimba
14	Xylophone
15	Tubular Bells
16	Dulcimer

33-40

BASS

33	Acoustic Bass
34	Electric Bass (finger)
35	Electric Bass (pick)
36	Fretless Bass
37	Slap Bass 1
38	Slap Bass 2
39	Synth Bass 1
40	Synth Bass 2

57-64

BRASS

57	Trumpet
58	Trombone
59	Tuba
60	Muted Trumpet
61	French Horn
62	Brass Section
63	Synth Brass 1
64	Synth Brass 2

81-88

SYNTH LEAD

81	Lead 1 (square)
82	Lead 2 (sawtooth)
83	Lead 3 (calliope)
84	Lead 4 (chiff)
85	Lead 5 (charang)
86	Lead 6 (voice)
87	Lead 7 (fifths)
88	Lead 8 (bass+lead)

105-112

ETHNIC

105	Sitar
106	Banjo
107	Shamisen
108	Koto
109	Kalimba

17-24

ORGAN

17	Drawbar Organ
18	Percussive Organ
19	Rock Organ
20	Church Organ
21	Reed Organ
22	Accordion
23	Harmonica
24	Tango Accordion

41-48

STRINGS

41	Violin
42	Viola
43	Cello
44	Contrabass
45	Tremolo Strings
46	Pizzicato Strings
47	Orchestral Strings
48	Timpani

65-72

REED

65	Soprano Sax
66	Alto Sax
67	Tenor Sax
68	Baritone Sax
69	Oboe
70	English Horn
71	Bassoon
72	Clarinet

89-96

SYNTH PAD

89	Pad 1 (new age)
90	Pad 2 (warm)
91	Pad 3 (polysynth)
92	Pad 4 (choir)
93	Pad 5 (bowed)
94	Pad 6 (metallic)
95	Pad 7 (halo)
96	Pad 8 (sweep)

113-124

PERCUSSIVE

113	Tinkle Bell
114	Agogo
115	Steel Drums
116	Woodblock
117	Taiko Drum

102	FX 6 (goblins)	110	Bagpipe	118	Melodic Tom
103	FX 7 (echoes)	111	Fiddle	119	Synth Drum
104	FX 8 (sci-fi)	112	Shanai	120	Reverse Cymbal

121-128

SOUND EFFECTS

121	Guitar Fret Noise	124	Bird Tweet	127	Applause
122	Breath Noise	125	Telephone Ring	128	Gunshot
123	Seashore	126	Helicopter		

PERCUSSIVE Channel 10 Note Values and Patches

35	Acoustic Bass Drum	51	Ride Cymbal 1	67	High Agogo
36	Bass Drum 1	52	Chinese Cymbal	68	Low Agogo
37	Side Stick	53	Ride Bell	69	Cabasa
38	Acoustic Snare	54	Tambourine	70	Maracas
39	Hand Clap	55	Splash Cymbal	71	Short Whistle
40	Electric Snare	56	Cowbell	72	Long Whistle
41	Low Floor Tom	57	Crash Cymbal 2	73	Short Guiro
42	Closed Hi Hat	58	Vibraslap	74	Long Guiro
43	High Floor Tom	59	Ride Cymbal 2	75	Claves
44	Pedal Hi-Hat	60	Hi Bongo	76	Hi Wood Block
45	Low Tom	61	Low Bongo	77	Low Wood Block
46	Open Hi-Hat	62	Mute Hi Conga	78	Mute Cuica
47	Low-Mid Tom	63	Open Hi Conga	79	Open Cuica
48	Hi-Mid Tom	64	Low Conga	80	Mute Triangle
49	Crash Cymbal 1	65	High Timbale	81	Open Triangle
50	High Tom	66	Low Timbale		

MIDI Notes and Frequencies

Note	MIDI	Hz	Note	MIDI	Hz
C-1	0	8.176	E 4	64	329.63
C#-1	1	8.662	F 4	65	349.23
D-1	2	9.177	F# 4	66	369.99
D#-1	3	9.723	G 4	67	391.99
E-1	4	10.301	G# 4	68	415.31
F-1	5	10.913	A 4	69	440.00
F#-1	6	11.562	A# 4	70	466.16
G-1	7	12.250	B 4	71	439.88
G#-1	8	12.978	C 5	72	523.25
A-1	9	13.750	C# 5	73	554.37
A#-1	10	14.568	D 5	74	587.33
B-1	11	15.434	D# 5	75	622.25
C 0	12	16.352	E 5	76	659.26
C# 0	13	17.324	F 5	77	698.46
D 0	14	18.354	F# 5	78	739.99
D# 0	15	19.445	G 5	79	783.99
E 0	16	20.601	G# 5	80	830.61
F 0	17	21.826	A 5	81	880.00
F# 0	18	23.124	A# 5	82	932.32
G 0	19	24.499	B 5	83	987.77
G# 0	20	25.956	C 6	84	1046.5
A 0	21	27.50	C# 6	85	1108.7
A# 0	22	29.135	D 6	86	1174.7

B 0	23	30.867	D# 6	87	1244.5
C 1	24	32.703	E 6	88	1318.5
C# 1	25	34.648	F 6	89	1396.9
D 1	26	36.708	F# 6	90	1480.0
D# 1	27	38.890	G 6	91	1568.0
E 1	28	41.203	G# 6	92	1661.2
F 1	29	43.653	A 6	93	1760.0
F# 1	30	46.249	A# 6	94	1864.7
G 1	31	48.999	B 6	95	1975.5
G# 1	32	51.913	C 7	96	2093.0
A 1	33	55.000	C# 7	97	2217.5
A# 1	34	58.270	D 7	98	2349.3
B 1	35	61.735	D# 7	99	2489.0
C 2	36	65.406	E 7	100	2637.0
C# 2	37	69.295	F 7	101	2793.8
D 2	38	73.416	F# 7	102	2960.0
D# 2	39	77.781	G 7	103	3136.0
E 2	40	82.406	G# 7	104	3322.4
F 2	41	87.307	A 7	105	3520.0
F# 2	42	92.499	A# 7	106	3729.3
G 2	43	97.998	B 7	107	3951.1
G# 2	44	103.82	C 8	108	4186.0
A 2	45	110.00	C# 8	109	4434.9
A# 2	46	116.54	D 8	110	4698.6
B 2	47	123.47	D# 8	111	4978.0
C 3	48	130.81	E 8	112	5274.0
C# 3	49	138.59	F 8	113	5587.7
D 3	50	146.83	F# 8	114	5919.9
D# 3	51	155.56	G 8	115	6271.9
E 3	52	164.81	G# 8	116	6644.9
F 3	53	174.61	A 8	117	7040.0
F# 3	54	184.99	A# 8	118	7458.6
G 3	55	195.99	B 8	119	7902.1
G# 3	56	207.65	C 9	120	8372.0
A 3	57	220.00	C# 9	121	8869.8
A# 3	58	233.08	D 9	122	9397.3
B 3	59	246.94	D# 9	123	9956.1
C 4	60	261.63	E 9	124	10548.1
C# 4	61	277.18	F 9	125	11175.3
D 4	62	293.66	F# 9	126	11839.8
D# 4	63	311.13	G 9	127	12543.9

Note: The chart shows the musical pitch and frequency associated to each MIDI Note in Sound Forge. A4 is at 440 Hz and middle C is C4 at 261.63 Hz. These values can differ, often by one octave, from values used by other manufacturers. To avoid having negative values, some MIDI sequencers and sound modules use C0 as MIDI Key 0, which then makes middle C C3 instead of the more standard C4. The frequencies below are based on equal temperament tuning.

RIFF Wave Chunks

The following is an overview of the defined RIFF Wave File chunks which Sound Forge 3.0 stores and recognizes. Although Microsoft recommends that the audio data be stored after all other chunks, the only chunks stored by Sound Forge prior to the data are the **fmt**, **fact**, and **PAD** chunks.

The reasoning proposed for storing the data after all other chunks is if the data is modified, the other chunks do not have to be rewritten to disk. However, since the auxiliary chunks are usually much smaller than the data, the associated time to rewrite them is usually negligible. For example if a play list chunk is modified to add a new region we have to rewrite the complete contents of the data to insert the new Playlist chunk. However if it is stored at the end of the data only the play list chunk and any chunks following it need to be rewritten. If we were operating with 3 minutes of 44k stereo data then we would need to rewrite 30 megabytes of data to insert 1 region change. The auxiliary chunks typically are less than 100k so we only have to move 100k with changes to the audio data itself.

Another reason for storing additional chunks after the data is that some non-robust software packages assume that all that will be found in a WAV file prior to the data is the **fmt** chunk. This allows you to still read WAV files to some degree with these packages.

Chunk	Use
PAD	This chunk is used by Sound Forge to align the DATA chunk on a sector boundary for use with products like Digidesign's Session 8.
fact	Used to specify the length of audio data in samples when data is stored in a format other than PCM.
cue	Used to specify the start of Markers and Regions.
plst	Used to specify a Playlist within a single wave file.
adtl	Used to specify region lengths and to provide text names for regions and markers.
smpl	Used to specify information for files used with samplers. Includes information on release and sustain loops.
tlst	Used to specify external trigger events for regions and Playlist entries.
inst	Unsupported, but documented for consistency.

For more information on the above chunks, refer to the RIFF specification available from Microsoft.

The following chunk has been proposed by Sonic Foundry but has not been registered with Microsoft. This chunk is used to save trigger information in WAVE files created by Sound Forge.

tlst (Trigger List) Chunk

Added: ??

Author: Sonic Foundry

Defined for : WAVE form

The <tlst-ck> trigger list chunk specifies a list of triggers which can be used to trigger playback of a series of cue points or play list entries. The <tlst-ck> is defined as follows:

```
|<tlst-ck>|    ()    tlst(  
                <dwTriggers:DWORD>    // count of triggers  
                <trigger(s)>...)    // trigger table
```

```
<trigger>    struct  
              {  
                FOURCC    fccListReference;
```

```

    DWORD    dwName;
    DWORD    dwType;
    DWORD    dwTriggerOn;
    DWORD    dwFunction;
    DWORD    cbExtra;
}

```

The <trigger> chunk:

fccListReference	Specifies the list which this entry references. A trigger can reference either the <cue-ck> or the <plst-ck>.
dwName	When the trigger references the <cue-ck> this field specifies the cue point name. This value must match one of the names listed in the <cue-ck> cue-point table. When the trigger references the <plst-ck> this field references a play list entry by index. This value must be less than the total count of <plst-ck> entries.
dwType	Specifies the type of trigger. 0 - SMPTE Trigger 1 - MIDI Command Trigger. 2 - MIDI SysEx Trigger. (SysEx message)
dwTriggerOn	Specifies the value which will cause a trigger to be generated. This value will change value depending on the type of trigger.

For triggers of type SMPTE (0)

Specifies a SMPTE time offset for the cue. The format of this value is 0xhhmmssff. hh is an unsigned Hours value [0..23]. mm is an unsigned Minutes value [0..59]. ss is an unsigned Seconds value [0..59]. ff is an unsigned Frames value [0..numFrames].

For triggers of type MIDI Command (1)

Specifies a MIDI Command.

Specified as :

Channel	Command	Param1	Param2
Byte 0	Byte 1	Byte 2	Byte 3

Where Byte 0 is the MSB and Byte 3 is the LSB.

For channels a value of FF signals trigger on all channels.

For any other param other than the MIDI command byte FF signals a don't care value.

For triggers of type MIDI Sysex (2)

Unused, should be set to 0.

dwFunction	Specifies the function of this trigger.
------------	---

- 0 - Play
- 1 - Stop
- 2 - Queue

Trigger functions which are unsupported or unknown should be ignored.

cbExtra Specifies the size in bytes of additional trigger information. For the case of a MIDI Sysex Trigger, this is the size of the matching MIDI sysex command immediately following the structure.

Any value within a sysex that is set to FF is treated as a don't care. This allows response on multiple channels, values, etc.

Examples:

Play when any MIDI note on is received:

FF 90 FF FF

Play when MIDI note 60 is received on channel 1.

00 90 3C FF

Play when MIDI note 60 is received on channel 1, stop when note off received for same note

00 90 3C FF

00 80 3C FF

Queue when MIDI note 60 is received on channel 2, Play when note off received for same note

01 90 3C FF

01 80 3C FF

Markers, Regions, and Playlist Entries as used by Sound Forge 3.0

When creating or reading markers, regions and playlist entries in WAV files, the RIFF documentation can be a little confusing. We have added this section to give software developers a better idea of how to deal with these chunks.

Markers

Markers are a position within a wave file which can also have an associated text name. Each marker in Sound Forge is stored as an entry within the <cue> chunk. The entry within the <cue> chunk stores the position as well as a DWORD name field. The name field is used to reference the text name entry which can be found within the <adtl> chunk. For instance if we had a Marker at 0 which had a name of Start there would be an entry within the <cue> chunk with a Sample Offset of 0 and a Name entry of 0x0001 (we have chosen 0x0001 for the example. The Name entry can be any value from 0x00000000 to 0xFFFFFFFF) Within the <adtl> chunk we would have an entry of type <labl> (label chunk) with a name of 0x0001 and a text field of Start.

Regions

Regions are much like a marker but they also include an associated length as well as a start position and text name. Each region is stored as an entry within the <cue> chunk which stores the position as well as a DWORD name field. The name field is again used to reference the text name entry which can be found within the <adtl> chunk. The difference between the marker and the region is that instead of using a <labl> chunk in the <adtl> chunk we use a <ltxl> chunk which allows us to store a length as well as a text name. For instance if we had a region at 0 with a length of 1000 samples and a name of Beat 1 there would be an entry within the <cue> chunk with a sample offset of 0 and a name entry of 0x0001. (Again we are using the name value of 0x0001 as an example.) Within the <adtl> chunk we would have an entry of type <ltxl> with a name of 0x0001, a text field of Beat 1, and a length of 1,000. Within the <ltxl> chunk there is also a field of type FOURCC which specifies the purpose of this entry. For all regions in Sound Forge the FOURCC type is set to rgn .

Playlist Entries

Playlist regions are stored a little differently. They specify a reference to a <cue> for the start of each play list region and a length and number of loops for each entry. Thus, our play list entries can actually have lengths that are different than those stored within the region list. To handle this, for every playlist entry there will exist a corresponding region of the exact same length when a wave file is created by Sound Forge. This is to simplify the overall use and interface to the user. If we were to have a playlist which contained our previous example of a region of Beat 1 playing 4 times, it would be stored in the play list with a Name field of 0x0001, a length of 1,000 samples, and a loop count of 4.

Tips and Ideas

[Pseudo-quantizing digital audio with MIDI sequencers](#)

[Realistic MIDI drum parts with MIDI triggering](#)

[Randomizing a Drum Loop](#)

Pseudo quantizing digital audio with MIDI sequencers

There are many ways to use Sound Forge with MIDI sequencers. The following two tips deal with drum patterns, though any type of material could be adapted to these techniques. These examples are some of many possible uses for the MTC sync and MIDI triggering capabilities of Sound Forge and MIDI sequencers. For information on setting up the Sync/triggers with specific sequencers, see **Appendix M: Using Sound Forge with MIDI Sequencers**.

Using MIDI note triggering to pseudo-quantize a Regions.

1. Record some drum tracks into Sound Forge that are in a consistent tempo and measure scheme.
2. Split up the sound file into regions that are a measure in length. In the Regions List, assign a different MIDI Note On trigger to each region.
3. Select a region and open the **Edit Tempo** dialog in Sound Forge's Special menu. By selecting **Measure**, you can now see what the approximate tempo (BPM) is according to the number of beats you have entered and the length of the selection. Examine a number of different regions in this way and take an average BPM.
4. Now, set the sequencer's tempo to match the BPM found in Sound Forge's **Edit Tempo**.
5. Next, create a MIDI track with whole notes that correspond to the note numbers selected in the Sound Forge Regions List for each separate region. These notes should fall on downbeats of the measures so there is one note per measure. It is a good idea to make their lengths a few clock pulses shy of the next downbeat. Also, it may be necessary to set a negative time offset for the whole track to allow for any slight delay that MIDI note triggering is prone to have.
6. Now, playing back the track in the sequencer while triggering Sound Forge should give you a pseudo-quantized version of the sound file. Each new measure in the sequencer should trigger a measure of audio material in Sound Forge. Although this method has potential for less accurate start points than MTC SMPTE triggering, it will allow for easier tweaking and alignment within the sequencer.

Using MTC synchronization to pseudo-quantize the Playlist.

1. Record some drum tracks into Sound Forge that are in a consistent tempo and measure scheme.
2. Split up the sound file into regions in the Regions List and Playlist that are a measure in length.
3. Select a region and open the **Edit Tempo** dialog in from the **Special** menu. By selecting **Measure** you can now see what the approximate tempo (BPM) is according to the number of beats you have entered and the length of the selection. Check a number of different regions in this way to find an average BPM.
4. Now, in the sequencer, set the tempo to match the tempo found in Sound Forge. Next, there should be a way (in the sequencer) to see what SMPTE time will correspond to whatever measure downbeat you choose. These times can be entered as the Sound Forge Playlist regions SMPTE start times. This procedure will allow the Playlist regions to start at the downbeat of the sequencer measures when Sound Forge is set to follow MTC. You will undoubtedly need to tweak the start times and possibly the sequencer tempo, but should be able to get it all to sound correct.
5. This method of triggering is quite accurate, since Sound Forge knows what region is to be triggered next and can pre-load the data accordingly. It is a little precarious, though, to have to adjust the SMPTE start times every time the sequencer tempo is changed due to tweaking.

Realistic MIDI drum parts with MIDI triggering

If you do a lot of MIDI sequencing, you know how unconvincing and static MIDI drum parts can be, especially if you're not a drummer. Good results can be achieved through the use of triggering Sound Forge regions from a MIDI sequencer to go along with drum parts that have been created in the sequencer. If you have a sampler, here is a procedure that yields quite convincing drum tracks.

1. First, record into Sound Forge some drum tracks/sounds from an audio source that has an abundance of drums alone material.
2. Run **Auto Region** on the sound file to automatically breakup the events into easily manipulated regions. To audition each region, open the Regions List and click on the play button for each entry.
3. Create some MIDI drum tracks in your sequencer without worrying too much about fills, flams, etc. These things will be added by MIDI triggering Sound Forge.
4. Now that you have a basic drum track in the sequencer, go to Sound Forge and choose events that are distinctly Drummer-esque from your Regions List (you might need to edit the lengths and positions of the regions). These are the types of events that in themselves do not make good single drum samples or are too long to reasonably use with a sampler with limited amounts of RAM (for example, drum fills, flams, long crash cymbals and events that have a certain amount of Human variation).
5. For each region you want to use, select a MIDI Trigger event in the Edit Marker/Region dialog. It is recommended that you use **MIDI: Note On - Queue / Note Off - Play** as this will ensure that the timing will be accurate. The only drawback to this method of triggering is that the actual the start of the region playback will occur on the *release* of the MIDI note in the sequencer. That is, the place in time the triggered region should start playing will have to line up to the end of the MIDI event. In some sequencers, this is a little easier to deal with than others. This is a way for Sound Forge to pre-load the region data for accurate playback at the trigger point. See the Using Sound Forge section on MIDI triggering for a detailed explanation of setting up Sound Forge MIDI triggers.
6. Now, in your sequencer, assign a new track with the same MIDI channel as specified in the MIDI Triggers for the regions. At this point each of your drum regions in Sound Forge can be triggered from your sequencer with MIDI notes. You can decorate your drum parts with these events to add variation and Drummer-isms that are not possible with single drum samples and MIDI alone.

Randomizing Playlist regions

Some very interesting rhythmic patterns can be "discovered" by using the Region Scrambler (a.k.a. the Egg Beater) technique on a drum track outlined below:

1. Open a file with a drum pattern. Run the **Auto Region** tool using the "Drum Beats" Preset (minor tweaks may be necessary).
2. In the Playlist window, randomly rearrange the regions. The fastest (and most fun) way to do this is to randomly drag and drop each region up and down the Playlist. Do this for about 10 seconds or until the regions are completely scrambled.
3. Now, listen to the rearranged regions in the Playlist.
4. Another thing you can do is to raise or lower the play count in some regions. You can do so by using the Numpad + and Numpad - key.
5. Repeat steps 2-4 until you're positive you've created the most complex rhythmic pattern known to man.
6. Create a new file from the playlist by using the **Convert to New** function. Now, play your new drum loop in Looped mode. Tap along to your new groove and try to figure out its time signature.
7. Cool, eh?

Using CSOUND, MTU, IRCAM, BICSF and EBICSF Files

This section discusses how to open and save files of the type CSOUND, MTU, IRCAM, BICSF, EBICSF with Sound Forge.

Introduction

Although Sound Forge supports a large number of sound file formats directly, it does not support the CSOUND, MTU, IRCAM, BICSF or EBICSF file types. However, by using the *Raw File Type* capabilities of Sound Forge, sound data from these file types can be extracted.

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. It also can contain information such as the name of the sample and comments on what is included in the file. This is the format used by the MTU system and is often referred to as MTU files. IRCAM files support two types of data formats: 16-bit linear PCM and floating point data.

BICSF or EBICSF files (Berkeley/IRCAM/CARL Sound File or Extended BICSF) are an extension of the IRCAM format. The difference being that instead of using the standard IRCAM header they replace the first 28 bytes of the header with a standard NeXT/Sun header. This allows the IRCAM format to continue to hold additional information in its 1024 byte header and still allow the files to be read by software that supports the NeXT/Sun file format (such as Sound Forge).

Opening and Saving files

BICSF and EBICSF files

When reading BICSF or EBICSF Sound Forge will identify them as NeXT/Sun files. This is due to the fact that the header of the BICSF file has been modified to allow it to be read as a NeXT/Sun File. Sound Forge has no difficulty reading these files as long as they are in one of the supported NeXT/Sun data formats.

IRCAM, CSOUND and MTU files

When reading any of these types you must import the files as a *Raw data file*. The best way to do this is to set up the correct parameters in the Sound Forge Raw File Type dialog box and then save presets labeled accordingly. Sound Forge's *Raw File* import will allow you to read these files as long as they are mono or stereo and are stored in 16-bit linear format. Sound Forge will not allow you to read the floating point format IRCAM files.

Opening an IRCAM file

To open an IRCAM sound file in Sound Forge, follow these steps:

1. Open the *File Open* dialog box.
2. Select the *Raw File* type.
3. Select an IRCAM file to load.
4. Press the *OK* button in the *File Open* dialog. This will bring up the *Raw File Type* dialog box.
5. Select the following parameters:
 - Sample rate*: Set the most common sample rate you use (typically 44,100).
 - Sample type*: 16-bit PCM
 - Format*: Signed
 - Channels*: *Mono* or *Stereo* (depends on the file)
 - Byte order*: *Big endian* or *Little endian* (depends on where the file was generated).
 - Header*: 1,024 bytes
 - Trailer*: 0 bytes
6. Before selecting *OK* to load the file press the *Save As* button to create a preset. Type in an appropriate preset name and press *OK*.

Now whenever you wish to load an IRCAM file, you can load it as *Raw File Type* and then select the preset you saved to set all the necessary parameters to load the file. Note that if any of the files are not the same as your preset, you should change them in the raw dialog box. You may even want to create multiple presets.

It is important to note that the *Byte order* of files is not always the same for files generated by CSOUND. The CSOUND executables for the PC generate files which use Little Endian byte ordering. CSOUND for other platforms tends to generate files with Big Endian ordering. MTU files also use Big Endian ordering.

The best policy is to try a file first in Big Endian. If the sound data seems to be garbage, then try loading the file using Little Endian. Again you may want to save presets for byte ordering, as well as mono/stereo, if you are receiving CSOUND files from a variety of other computers.

Saving files with Sound Forge

When saving files of the above types you will not be able to save them as IRCAM, CSOUND or MTU. You will have to save any files of these types as one of the file formats supported by Sound Forge.

If you wish to save a file for use with software which supports the BICSF/EBICSF format you should save the file as a NeXT/Sun file which will not save any of the additional information found in a BICSF/EBICSF file but should allow the software to read the data as a NeXT/Sun file.

Troubleshooting

1. When playing files from my SCSI drive, why does the sound have a warbling effect? [ANSWER](#)
2. When saving files as Macintosh AIFF and transferring them to my Macintosh, why does the Macintosh not recognize them as sound files? [ANSWER](#)
3. After running an effect on a file, why is there a loss in volume level? [ANSWER](#)
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15. When playing Sound Files with rates above 44,100 Hz on my Turtle Beach Multisound or Tahiti, my system crashes. [ANSWER](#)
16. When playing Sound files at 8,000 Hz on my Turtle Beach Multisound, the play pointer moves but I do not hear any output. [ANSWER](#)
17. When playing Sound Files at rates other than 11, 22, or 44 kHz the files sound wrong (the pitch is wrong). [ANSWER](#)
18. When making a selection with the mouse, why does the selection change to another location? [ANSWER](#)
19. Microsoft ADPCM files created using Sound Forge crash when I play them in Media Player on my Turtle Beach sound card. Why? [ANSWER](#)
20. When opening an AVI file, if the audio stream format is 44.1 kHz stereo file Microsoft or IMAADPCM, the audio track does not open correctly. [ANSWER](#)

21. After installing the Sonic Foundry Virtual MIDI Router in Windows 95, my system is having trouble with MIDI devices or the VMR. [ANSWER](#)

22. When I play a sound file, I sometimes hear a click at the beginning or end of the selection. [ANSWER](#)

22. While drawing the Waveform Display, my system locks up completely. [ANSWER](#)

Q: When playing files from my SCSI drive, why does the sound have a warbling effect?

A: When playing files from a SCSI drive that is connect to a SCSI adapter that utilizes a DMA channel (such as most Adaptec adapters) you may experience a warbling effect in the sound playback. This can occur if your sound card uses DMA and its DMA channel is set to a higher number than your SCSI adapter.

To fix this problem, set the DMA channel used by your sound card is a lower number than that used by your SCSI adapter. So, if your SCSI adapter is set to DMA channel 5 and your sound card is set to DMA channel 7, try reconfiguring them so the sound card uses DMA channel 5 and your SCSI adapter uses DMA channel 7.

See also

[Troubleshooting](#)

Q: When saving files as Macintosh AIFF and transferring them to my Macintosh, why does the Macintosh not recognize them as sound files?

A: The Macintosh uses special information within its operating system to keep track of the types of files to which it has access. Sound Forge for Windows can save the audio data in the correct file format, but it cannot tell the Macintosh what type it is. To do this you must use a program on the Macintosh, like ResEdit, which allows you to change the Type of the file. You simply need to set the Type field to AIFF for the Macintosh to recognize the file as a sound file.

See also

[Troubleshooting](#)

Q: After running an effect on a file, why is there a loss in volume level?

A: Most effects that can alter the volume level of a file have two fader controls, *Dry Out* and *Effect Out* (where Effect is the name of the function, like *Reverb*). These controls should be adjusted to get the proper balance between processed and non-processed sound and also determine the combined volume level. For example, if you run an effect with both controls set at 40% and the level after processing is too low, simply raise both the Dry and Effect Out faders an equal amount to raise the final level. However, take care to keep the output signal from [clipping](#) .

See also

[Troubleshooting](#)

Q: Why do some effects take so long to process a file?

A: Some processing functions used by Sound Forge are very math intensive. Examples include EQ, pitch change without change of duration, and the reverb effect. This means that millions of integer and floating-point arithmetic operations must be performed for each second of sampled data. If your computer's CPU does not have a math co-processor (in the Intel architecture that means if you have an i486 SX instead of an i486 DX or Pentium), floating-point operations will drastically slow you down and there is not much you can do except save up for a faster processor.

Of course, running effects on stereo, 44,100 Hz, hour long files will slow you down no matter what machine you are using. If you are going to be deleting or inserting space into these huge files, we recommend that you break the files into smaller ones and then mix the smaller files together once you are done processing them. This is especially painful when you insert or delete from the beginning of a large file, since Sound Forge must then shift the rest of the file over.

One way to get around disk related editing delays is to use the Regions List and Playlist to make all arrangements non-destructively and then use the [Convert to New](#) function on your Playlist. Another useful feature in Sound Forge is the [Preview Cut/Cursor](#) function under the *Edit* Menu--use this function to verify your Cut and Clear operations before actually changing the file.

See also

[Troubleshooting](#)

Q: When I start Sound Forge my Play and/or Record buttons/options are grayed out. Why can I not play or record?

A: When Sound Forge starts, it checks to see if you have a wave playback and record device installed in your system. If you do not then it disables the appropriate operations. If you are sure that you have a sound card installed in your system, yet Sound Forge does not recognize this, you need to check a couple of things. First check the [Wave folder](#) in the *Preferences* dialog. The two settings of importance are the *Playback* and *Record* drop-down list boxes. These options determine the wave playback and record devices which will be used by Sound Forge. By default Sound Forge uses the Wave Mapper (may also be called the Sound Mapper) device which should automatically pick an appropriate device for use by Sound Forge. If either the Playback or Record device is set to (None) then Sound Forge can not execute the corresponding operation. To learn more about configuring your wave devices refer to the Wave Preference folder in the Reference section.

See also

[Troubleshooting](#)

Q: Why can I not get Sound Forge to respond to levels from my Microphone or my CD Player?

A: Most sound cards come with multiple inputs which you can record sound from. These may include microphone, CD, auxiliary, MIDI and line inputs. Although some cards record from all of these inputs at the same time, many cards make you choose which device is currently your record device (also known as the input source). These cards come with software which allows you to pick which device is the input source. You will need to **refer to your sound card manual** to determine how to set the input source.

Most sound cards also come with some kind of application which controls the input level of the recording device. You may need to **increase the level of your input sources** in order to record from them. Again, you will need to refer to your sound card manual to learn how to adjust these levels.

See also

[Troubleshooting](#)

Q: I have sound files that do not want to play on my system. Sound Forge loads them fine, why will it not play them?

A: Sound Forge allows you to load many different types of sound files, including ones that your system may not be capable of playing. There are a variety of reasons that your sound card may not be capable of playing a file.

Some sound cards only allow playback and recording at specific sample rates. Typically these rates are limited to 11,025, 22,050, and 44,100 Hz. If you try to play a file that is 32,000 Hz the sound card will not allow you to play it.

If you have an older sound card it may only support Mono playback. If this is the case simply convert any Stereo files to Mono and you will be able to play them.

If you have an older sound card it may only support 8 bit playback. If this is the case simply change the sample size of the file from 16 to 8 bits.

See also

[Troubleshooting](#)

Q: Why will Sound Forge not open my compressed files like Microsoft ADPCM?

A: Sound Forge has built in support for the Microsoft Audio Compression Manager (ACM). This allows you to open files that are compressed with a variety of algorithms including Microsoft ADPCM, IMA ADPCM, and other third party compression schemes like True Speech from The DSP Group.

If you cannot open a Microsoft ADPCM file, you may not have the Microsoft ACM installed in your system or you do not have the Microsoft ADPCM driver installed and enabled for the ACM. Please refer to the Appendix's on Installing the Microsoft Audio Compression Manager and Sound Forge and The Microsoft Audio Compression Manager for more information on how to install and use the ACM with Sound Forge.

See also

[Troubleshooting](#)

[The Audio Compression Manager](#)

[Installing the ACM](#)

Q: Why is it when I play my sound files the play pointer that Sound Forge displays seems to be off from what I am hearing?

A: For Sound Forge to display the current position of the pointer relative to playback of audio data it must rely on your sound card driver to provide the correct position. Many sound card drivers do not do a very good job of this. Typically the position these cards provide is both off by a fixed amount as well as reported only in large discrete steps. Sound Forge provides a way to compensate for this by setting various options in the [Wave Preferences](#) folder (File menu). Refer to the Wave Preferences folder in the Reference section for more information.

See also

[Troubleshooting](#)

Q: When I record or play back files I hear what sounds like small clicks in the sound.

A: When Sound Forge is recording or playing audio data it needs to move large amounts of data between your hard drive and your sound card. This can be a tremendous amount of work for your computer depending on the kind of system you have. If Sound Forge can not move the data fast enough then you will hear small gaps between blocks of data which typically sound like clicks or puttering. There are a number of things you can do to reduce the overhead during the time Sound Forge is moving sound data, all of which are covered in the Using Sound Forge section called [Optimizing Sound Forge](#).

See also

[Troubleshooting](#)

[Clicks at start or end of playback, Q&A](#)

Q: When I record (or play) data at high sample rates, I cannot get Sound Forge to stop by pressing the Stop button.

A: Sound Forge tries very hard to keep recording and playback of audio smooth (no gaps in your sound). So there may be delays between the time that you press the Stop button and the time that Sound Forge actually stops. Note that you can always **press and hold the Escape key** to stop record or playback no matter how bad the situation is. Just be a little patient and wait a few seconds. These delays are usually only observed when one or more components of your system can not keep up with the data rate. Typical faults are:

1. Very slow (older) hard drives.
2. Poorly written Sound Card Drivers.
3. Setting your Storage directory to a slow device like a Network drive or a Compressed Hard drive.

If at all possible, use the fastest **uncompressed** storage device available with Sound Forge. For more information on configuring your system, see the section on [Optimizing Sound Forge](#).

See also

[Troubleshooting](#)

Q: When using SMPTE/MTC to drive my sequencer, my sequencer sometimes loses sync.

A: Depending on the performance of your system (and your sequencer) you may experience SMPTE sync drop-out problems. This is typically due to too many things going on at once (i.e. more than your computer is capable of maintaining). For example, if you have wave playback, synchronization messages, sequencing, and perhaps more going on at the same time, you may experience sync drop-out.

To compensate for this, many sequencers allow you to set a freewheel time, which is a slack time between SMPTE messages during which the sequencer will continue to play even if SMPTE/MTC messages are being missed. By increasing this freewheel time you allow Sound Forge and your sequencer to be less tightly dependent on SMPTE messages, but this may also decrease your synchronization accuracy. The best recommendation is to start with a small freewheel time and, if you experience sync drop-out problems, increase the freewheel time until the drop-outs go away.

See also

[Troubleshooting](#)

Q: After converting WAV files to Redbook audio using the software which came with my Compact Disc recorder, there are audible clicks at the end of the track.

A: Not all software for converting WAV files to Redbook audio handles the WAV file structure correctly. The software simply skips the first 44 bytes of data in the WAV file and then considers everything else in the file to be audio data. This means that if you have saved Summary Information, Sampler Information, or Regions and Playlist data in the WAV file it will be interpreted as audio data.

If you are experiencing this problem simply, use the **Save As** command in Sound Forge to save a new WAV file with the *Save Summary Information*, *Save Sampler Information*, and *Save Regions/Playlists* check boxes **unchecked**. This tells Sound Forge to save only the minimum amount of information needed for a WAV file. Also saving a file with *Sector aligned data for Digidesigns Session 8 (File Preferences)* turned on can cause a glitch at the start of the sound file in other software.

See also

[Troubleshooting](#)

Q: My Compact Disc recorder uses a file type with an RBK extension. What is an RBK file and can I open it with Sound Forge?

A: RBK files are simply Raw PCM sound data files. These files are used by some Compact Disc recorders. These files are used to store Redbook audio onto a Compact Disc. RBK files are data only and contain no other information. They are stored as 16 bit, stereo data with a sample rate of 44.1 kHz. There are a number of things that make them different than a normal 16 bit raw file:

1. The channels are stored in reverse order.
2. The data is stored in Big Endian format (Motorola).
3. The data length of the file has to be a multiple of 2352 bytes (1 CD sector). The size of a CD sector is calculated as follows;

1 second of 44.1k, 16 bit, stereo Audio = 176,400 bytes.
CD audio has 75 sectors of data for each 1 second of audio.
Sector size = $176,400/75 = 2352$ bytes.

To open an existing RBK file, simply use the RAW file option in Sound Forge's Open dialog box and select the RBK preset. Remember that you will have to swap the channels if you want to maintain the correct left/right channel settings.

See also

[Troubleshooting](#)

Q: When playing Sound Files with rates above 44,100 Hz on my Turtle Beach Multisound or Tahiti, my system crashes.

A: This is caused by a severe bug in the Turtle Beach Multisound and Tahiti sound card drivers. You will experience the same crash no matter what software you use to play sample rates above 44,100 Hz-- including Media Player and even Turtle Beach's own Wave application.

Contact Turtle Beach for updated drivers.

See also

[Troubleshooting](#)

Q: When playing Sound files at 8,000 Hz on my Turtle Beach Multisound, the play pointer moves but I do not hear any output.

A: This is caused by a bug in the Turtle Beach Multisound driver.

Contact Turtle Beach for updated drivers.

See also

[Troubleshooting](#)

Q: When playing Sound Files at rates other than 11, 22, or 44 kHz the files sound wrong (the pitch is wrong).

A: This is due to poorly written sound drivers that incorrectly play sounds with sample rates that are not supported by the sound card hardware. For example, a sound that is supposed to play at 13,000 Hz might get played at 11,025 Hz instead. This will cause the pitch to be lower and the playback time slower than it should be. The Microsoft Windows Sound System Version 2.0 and many 'compatible' sound cards contain this unfortunate bug.

Contact your sound card manufacturer for updated drivers.

See also

[Troubleshooting](#)

Q: When making a selection with the mouse, why does it change to another location?

A: If you have the [Snap to Zero Crossings](#) or [Snap to Time](#) option on (*Options* menu), the selection points will be automatically moved to the nearest zero crossing or time unit. If you turn both of these options off, selections made will not be modified by Sound Forge.

See also

[Troubleshooting](#)

Q: Microsoft ADPCM files created using Sound Forge crash when I play them in Media Player on my Turtle Beach sound card. Why?

A: Turtle Beach has implemented the Microsoft ADPCM format in the DSP of their audio cards. Depending on the version of the drivers you are using, you may crash when playing files created in Sound Forge on these cards.

This is a bug in the Turtle Beach drivers. Contact Turtle Beach for updated drivers.

See also

[Troubleshooting](#)

Q: When opening an AVI file, if the audio stream format is 44.1 kHz stereo file Microsoft or IMA ADPCM, the audio track does not open correctly.

A: This is due to a known bug in Video for Windows. It only happens if the file is both 44.1 kHz, stereo, and compressed using Microsoft ADPCM. To get around this problem, resample the audio to 22 kHz before saving.

See also

[Troubleshooting](#)

Q: After installing the Sonic Foundry Virtual MIDI Router in Windows 95, my system is having trouble with MIDI devices or the VMR.

A: If you have a large number of MIDI devices in your system or a multi-port MIDI board, such as an eight port MIDI interface, you may have problems installing additional VMR MIDI interface ports. Windows 95 has a limit on the number of MIDI devices which it can support. If you have 12 or more MIDI ports, you will likely crash while booting up Windows 95. If you have trouble with MIDI devices causing system errors after installing the VMR, you should either reduce the number of ports in use by the VMR, remove the VMR, or reduce the number of MIDI devices in your system.

See also

[Troubleshooting](#)

Q: When I play a sound file, I sometimes hear a click at the beginning or end of the selection.

A: If you hear a click during playback start or stop, the first thing to check is whether there is a waveform glitch or sudden spike at either end of your selection (or file). Zoom In Full and ensure that the selection start and end points are located in complete silence and no glitches exist.

If you're certain that the selection endpoints contain no audio but still hear a pop, the most likely culprit is your sound card's software driver. We have found that some versions of several sound card's drivers cause random glitching when starting or stopping playback.

Sound Forge has been tested with many different sound cards by thousands of users, and we are very confident in its correct implementation of standard Windows audio functions. Almost all of the technical support calls we receive are due to hardware conflicts or poorly written sound card drivers. If you experience any sound recording or playback problems, follow these steps:

1. Ensure that you are using unique interrupts (IRQs) and DMA channels between all of your hardware.
2. Try playing the sound file with Media Player (mplayer.exe or mplayer32.exe), which is included with Windows 3.x, 95 and NT. If the playback glitching exists with Media Player also, there is definitely a problem with the sound card hardware or driver.
3. Contact your sound card manufacturer and request for the latest drivers for your sound card.

See also

[Troubleshooting](#)

[Hear clicks during playback, Q&A](#)

Q: While drawing the Waveform Display, my system locks up completely.

A: If you experience a system lockup frequently in Sound Forge, it is likely that your video card is causing this problem. This problem is known to happen with ATI's Mach64 card when using High Color (16-bit) in Windows 95.

Because Sound Forge's drawing routines are highly optimized, they increase the chance of causing little-known video card problems to arise. To solve this problem, turn on *Compatible Draw Mode (for Broken Video Drivers)* under the *General Preferences* (Options menu).

See also

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ADPCM

Adaptive Delta Pulse Code Modulation (ADPCM) is a method of compression audio data. Although the theory for compression using ADPCM is standard, there are many different algorithms employed. As an example, Microsofts ADPCM algorithm is *not* compatible with the International Multimedia Associations (IMA) approved ADPCM.

A-Law

A-Law is a companded 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 non-linear 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.

Aliasing

Aliasing is 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 backwards 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.

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.

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 which can strongly alter the timbre of a sound.

Attack

The attack of a sound is the initial portion of the sound. Percussive sounds (like 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 (ACM) 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, as well as apply DSP algorithms to audio data.

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.

Bit

A bit is 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

Bit Depth refers to the number of bits used to represent a sample. For example 8 bit and 16 bit are two common Bit Depths. Sound Forge uses the term Sample Size when referring to Bit Depth.

Byte

A 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 in Sound Forge 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.

Chorus

Chorus 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 it from a sample so that you may past it or mix it into another sample. Sound Forge maintains it's own clipboard for cutting and pasting sample data allowing you to keep memory usage to a minimum. However, this does not change the normal operation of the clipboard and you can still cut and paste between Sound Forge and other Windows applications.

Clipping

Clipping is what 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.

Cutoff-frequency

The cutoff-frequency of a filter is the frequency at which the filter changes its response. For example, in a lowpass filter, frequencies greater than the cutoff frequency are attenuated while frequencies less than the cutoff frequency are not affected.

CODEC

Acronym for Coder/Decoder. Commonly used when working with data compression.

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.

Data Window

Each opened sound file in Sound Forge has its own Data Window. On 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, Ruler, Playbar and other tools that give you information and allow you to navigate throughout the entire sound file.

DC Offset

DC Offset is a phenomenon that occurs when hardware such as a sound card adds DC current to a recorded audio signal. This current causes the audio signal to alternate around a point above or below the normal -infinity dB (center) line in the sound file. To visually see if you have a DC offset present you can zoom all the way into a sound file and see if it appears to be floating over the center line. Sound Forge can compensate for this DC offset by adding a constant value to the samples in the sound file.

Decibel

The decibel (abbreviated as dB) is 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 $\text{dB} = 20 \times \log (V1/V2)$ 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, 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.

Destructive Editing

Destructive editing is the type of editing whereby all cuts, deletes, mixes and other processes are actually processed to the sound file. Anytime you delete a section of a sound file in Sound Forge the sound file on disk is actually re-written without the deleted section. This is different than non-destructive editing. See Non-destructive Editing.

Direct Edit Mode

Opening a file from the Open dialog with the *Operate directly on the sound file* box checked, opens the file in direct edit mode. This simply means that no temporary file is created for the file. The first time you open a file in direct edit mode, a peak data file will be created and stored in the directory that the file resides. From then on opening and saving the file will be almost instantaneous regardless of the size of the file. Please note, however, that any edits you make to the file are actually changing the file. With the unlimited Undo capability of Sound Forge this shouldn't be a problem, but keep in mind that if the system crashes for any reason, the file will be saved in the state that it is in when it crashes and no undo will be available.

Device Driver

A Device Driver is a program that allows Windows or DOS to connect different hardware and software together. For example, a sound card device driver is used by Windows software to control sound card recording and playback.

Digital Signal Processing

Digital Signal Processing (DSP) is a very 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.

Drag-and-Drop

Drag-and-Drop is a quick way to perform certain operations using the mouse in Sound Forge. 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. For more information on where you can use Drag-and-Drop, look in Appendix A: Shortcuts.

Dynamic Range

Dynamic Range is 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). For example, orchestral music has a wide dynamic range while thrash metal has a very small (always loud) range.

Equalization (EQ)

Equalizing a sound file is a process by which certain frequency bands are raised or lowered in level. EQ has various uses. The most common use for Sound Forge users is to simply adjust the subjective timbre of a sound.

File Associations

This dialog allows you to associate sound file extensions (such as .wav, .au, .snd, etc.) with Sound Forge. This dialog is opened from the File page of the Preferences dialog.

File Format

A file format specifies the way in which data is stored on your floppy disks or hard drive. In Windows, the most common file format is the Microsoft WAV format. However, Sound Forge can read and write to many other file formats so you can maintain compatibility with other software and hardware configurations.

Frequency Modulation

Frequency Modulation (FM) is a process whereby the frequency (pitch) of a sound is varied over time. Slow frequency modulation results in pitch-bending effects. Fast frequency modulation creates many different side-band frequencies.

Frequency Spectrum

The Frequency Spectrum of a signal refers to its range of frequencies. In audio, the frequency range is basically 20 Hz to 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.

Insertion Point

The Insertion Point is like the cursor in a word processor. It is where pasted data will be placed or other data 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 any where in the data window.

MCI

The Media Control Interface (MCI) is a standard way for Windows programs to communicate with multimedia devices like sound cards and CD players. If a device has a MCI device driver, it can be easily controlled by most multimedia Windows software.

Media Player

Media Player is a Windows 3.1 program that can play digital sounds or videos using MCI devices. Media Player is useful for testing your sound card setup. For example, if you can't hear a sound play through your sound card in Sound Forge, try using Media Player. If you can't play the sound using Media Player, check the sound card's manual (do not call Sonic Foundry's Technical Support until you've called the sound card manufacturer).

MIDI

Musical Instrument Device Interface (MIDI) is a standardized way for music equipment to communicate. Technically, it is a serial communications protocol using a 31.25 kHz baud rate. MIDI is used to send control, timing, and data information between devices for things like triggering notes, time synchronization, and sample dumps.

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.

Non-Destructive Editing

This type of editing involves a pointer-based system of keeping track of edits. When you delete a section of audio in a non-destructive 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. Sound Forge is a destructive editor except for when using the playlist.

Normalize

To normalize a file means to raise its volume so that the highest level sample in the file reaches a user defined level. Use this function to make sure you are fully utilizing the dynamic range available to you. Sound Forge also allows normalization to RMS power. This means that a scan will be done on the sound file and it will be raised in level so its RMS power will be equal to the normalization level. This is helpful for making multiple files perceptually as loud as each other.

MTC

MIDI Time Code (MTC) is an addendum to the MIDI 1.0 Specification and provides a way to specify absolute time for synchronizing MIDI capable applications.

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.

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.

Peak Data File

A peak data file is what is created by Sound Forge when a file is opened in direct edit mode for the first time. This file stores the information regarding the graphic display of the waveform so that opening a file is almost instantaneous in direct edit mode. This file is stored in the directory that the file resides in and has a .sfk extension. If this file is not in the same directory as the file, or is deleted, it will be recalculated the next time you open the file in direct mode.

Playbar

The Playbar is the bar at the bottom of each Data Window. On the left of the Playbar are the following buttons: Go To Start, Go To End, Stop, Play Normal, Play Loop, Play Sample. On the right are the Selection Status Fields which display the selection start, end, and length.

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 when doing a fade out.

Reactive Preview

Reactive Previews allow for the tweaking of parameters in a function dialog while the preview is playing. When a parameter is changed the preview will automatically rebuild and continue playback.

Region

A region in Sound Forge 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

To resample means to recalculate 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 will interpolate extra sample points in the sound file. This increases the size of the sound file but does not increase the quality. When down-sampling one must be aware of aliasing. See Aliasing.

Root Mean Square

The Root Mean Square (RMS) of a sound is a measurement of the intensity of the sound over a period of time. The RMS power of a sound corresponds to the loudness perceived by a listener when measured over small chunks of time.

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, 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 either percents or dBs.

Ruler Tags

Ruler Tags are the icons on the Ruler which represent the location of Markers, Regions, and Loop points in the Waveform Display.

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.

SDS

The MIDI Sample Dump Standard (SDS) is a way to transfer samples between music equipment. Samples transferred with SDS are sent across MIDI cables at the MIDI data rate of 31.25 kHz baud. The slow data rate of MIDI makes SDS a painful experience when transferring large samples. SMDI is a much preferred sample transfer method for musicians.

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 which contains the amplitude value of a waveform measured over time.
- ◆ A sound which 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 manual, 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.

Sampled Sound

Sampled sounds are sounds which have been recorded in a digital format and saved on your system for playback through a sound board. A sampled sound is sometimes referred to as a *sample*. However, a *sample* can also refer to a single point in a sampled sound.

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 playback samples while changing the sample pitch.

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.

Sample Size

The Sample Size, or Bit Depth, refers to the number of bits used to represent a single sample. Sound Forge uses either 8 or 16 bit samples. While 8 bit samples take up less memory (and hard disk space), they are inherently noisier than 16 bit samples.

Shortcut Menu

A Shortcut menu, also called a Context menu, is a context sensitive menu which 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 for quick access to many commands. An example of a Shortcut menu can be found by right clicking on any Waveform Display in a Data Window.

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 values and uses zero to represent silence. This is a format option when opening and saving RAW sound files.

SMDI

SCSI MIDI Device Interface (SMDI) is a standardized protocol for music equipment to communicate. 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.

SMPTE

The Society of Motion Picture and Television Engineers time code. This code is used to synchronize time between devices. The time code is calculated in Hours:Minutes:Second:Frames, where Frames are 1/30 of a second.

Status Bar

The Status Bar is the bar at the bottom of the Sound Forge main screen. The Status Bar fields on the right of the Status Bar contain information about the active Data Window (Sample Rate, Sample Size, Stereo/Mono, Total Size). On the right side of the Status Bar, help and processing information is displayed.

Tempo

Tempo is the rhythmic rate of a musical composition, usually specified in Beats Per Minute (BPM).

u-Law

u-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 non-linear 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.

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.

Video for Windows (AVI)

AVI files are a format of digital video for Windows. Sound Forge allows you to open these files and edit the audio embedded within them.

Virtual MIDI Router (VMR)

A Virtual MIDI Router (VMR) is a software only router for MIDI data between programs. No MIDI hardware or cables are required for a VMR, so routing can only be performed between programs running on the same PC. A VMR is normally used to synchronize two MIDI capable programs (for example, a VMR allows Sound Forge to drive a sequencer with SMPTE/MTC). Sonic Foundry supplies a VMR with Sound Forge called the Sonic Foundry Virtual MIDI Router.

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. The Waveform Display in each Data Window represents exactly this.

Waveform Display

The Waveform Display is the part of the Data Window which shows a graph of the sound data waveform. The vertical axis corresponds to the amplitude of the wave. 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 left most 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.

Zero Crossing

A Zero-Crossing is the point where a fluctuating signal crosses the zero amplitude axis. 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. Sound Forge's fades are accomplished using 64-bit arithmetic, thereby creating no audible zipper noise.

Using the Sampler Tool

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[SCSI/SMDI Hardware Setup](#)

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[Tips and Ideas](#)

[Akai S1000 MIDI/SDS](#)

[Digidesign SampleCell II](#)

[Kurzweil K2000 MIDI/SDS](#)

[Kurzweil K2000 SCSI/SMDI](#)

[Peavey SP MIDI/SDS](#)

[Peavey SP SCSI/SMDI](#)

Introduction

Without the convenience of a large computer monitor, an advanced operating system like Windows, and an intuitive interface, working with samples can be a nightmare. With the *Sampler Tool*, you can use Sound Forge's powerful and easy-to-use sample editing capabilities to create and edit samples and then transfer them to your external and internal samplers.

This chapter provides all the information you need to transfer (dump) samples between your computer and sampler using the Sound Forge *Sampler Tool*. In addition, we have included a section titled Tips and Ideas that may stir up some new, and perhaps unconventional, ways of using your sampler.

The documentation that came with your sampler undoubtedly has insights on sampling and loop tuning as well, so you might want to find it, dust it off, and browse through it. Many of these manuals can be very useful when you read them.

External Samplers

The MIDI SDS is used to send and receive digital samples with your normal MIDI hardware and cable connections. Because of the limited bandwidth of the MIDI protocol and the large amounts of data needed to store digital samples, performing a sample dump using the normal MIDI SDS can be slow, often taking minutes to transfer short samples. SDS is also limited to mono-only samples, though some samplers allow you to combine two mono samples to create a single stereo sample.

- ◆ **MIDI Sample Dump Standard (SDS):** The MIDI Sample Dump Standard (SDS) is used to send and receive digital samples with your normal MIDI hardware and cable connections. Because of the limited bandwidth of the MIDI protocol and the large amounts of data needed to store digital samples, performing a sample dump using the normal MIDI Sample Dump Standard (SDS) can be slow, often taking minutes to transfer short samples. SDS is also limited to mono only samples, though some samplers allow you to combine two mono samples to create a single stereo sample.
- ◆ **SCSI MIDI Device Interface (SMDI):** The SCSI MIDI Device Interface (SMDI) is a relatively new standard for music hardware and software to communicate using SCSI hardware and cables. Because SCSI hardware has a much larger bandwidth than MIDI, *SMDI sample dumps are many times faster than SDS dumps*. Another advantage to using SMDI is that it supports mono and stereo sample dumps (unlike SDS).

Internal Samplers

Internal samplers are cards that are installed inside of your computer. Unlike most sound cards, *internal samplers* allow you to download your own sounds into the card's memory. These sounds can then be played at different pitches for use as a musical instrument, just like an external sampler.

The Sampler Tool currently supports only one internal sampler, the Digidesign SampleCell II. Refer to the SampleCell II documentation for installation instructions. Once installed, use the procedures outlined in the section labeled **Digidesign SampleCell II - Sample transfer procedures**.

Internal samplers that are not directly supported by the Sampler Tool could use the MIDI Sample Dump Standard (SDS). Though no Windows compatible samplers support SDS (that we know of), it would be very simple for a manufacturer to implement this feature. If nothing else, SDS could be supported using an Open Loop transfer which would eliminate the complexity of supplying an input device. Open Loop would also make it possible for an internal sampler to accept data at far greater speeds than MIDI normally allows.

If you own a Windows compatible internal sampler, you may want to contact the manufacturer about supporting SDS in their Windows drivers. SDS may not be the best solution, but it is far better than no solution.

Sampler Hardware Setup

If you do not have your sampler and MIDI or SCSI card set up for transferring samples, you should refer to the **MIDI/SDS Setup and Troubleshooting** or **SCSI/SMDI Setup and Troubleshooting** chapters in this manual. These chapters will help you install the necessary components for transferring samples using the Sample Tool. There are also suggestions for trouble shooting problematic hardware configurations.

See also

[Using the Sampler Tool](#)

Configuring the Sampler Tool

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Creating a Sampler configuration

After you have your sampler hardware installed and functioning correctly, you must configure the *Sampler Tool* for sending and receiving samples from your sampler.

If you do not have your sampler and MIDI or SCSI card set up for transferring samples, you should refer to the MIDI/SDS Setup and Troubleshooting or SCSI/SMDI Setup and Troubleshooting later in this chapter. This information will help you install the necessary components for transferring samples using the *Sampler Tool*. There are also suggestions for troubleshooting problematic hardware configurations.

Select *Sampler* from the Tools menu and click the *Configure* button. Note that you must have at least one data window open to access the Tools menu. The *Sampler Configuration* dialog lets you select your sampler and sample transfer mode for both sending and receiving. The most common configuration is to have both the send and receive sampler and transfer mode the same. This is not a requirement, however, and there are situations where differing send and receive setups can be useful. One example of this might be if you want to transfer samples from one sampler to another with editing in between. Multiple configurations can be saved as presets to match different scenarios.

The first step to configuring your sampler setup is to choose the sampler and transfer mode (MIDI/SDS or SCSI/SMDI) from the *Send to* and *Receive from* drop-down lists. Several samplers and transfer modes are listed by name in these lists. For example, Peavey SP (SCSI/SMDI) specifies a *Peavey SP SCSI/SMDI* transfer protocol. If your sampler is listed, select it from the list.

If your sampler is not listed in the *Send to* and *Receive from* drop-down lists, try using the *Generic (SCSI/SMDI)* or *Generic (MIDI/SDS)* options. As long as your sampler properly supports the transfer mode (protocol) you select, the *Sampler Tool* should be able to work with your sampler. If you have trouble and you are certain your sampler is properly set to send or receive samples, contact your sampler's manufacturer. They may have an updated ROM available to correct the problem.

See also

[Using the Sampler Tool](#)

Internal sampler configuration options

Currently, only the Digidesign SampleCell II internal sampler is directly supported by the Sampler Tool. If you have a SampleCell II and you have selected it as the *Send to* sampler, then there are no further configuration options. A complete discussion on the SampleCell II can be found in the Appendix section on Sample Transfer Procedures for Digidesign SampleCell II.

The SampleCell II is only available for the *Send to* option; you cannot receive samples from the SampleCell II.

See also

[Using the Sampler Tool](#)

MIDI/SDS configuration options

If you choose a sampler that uses the MIDI/SDS transfer protocol in the *Send to or Receive from* drop-down lists, then you need to specify a few more options. These options include the *MIDI input* and *MIDI output* ports, the *MIDI channel* to transfer on, and whether handshaking should be used. You can also configure how a transfer will be initiated if handshaking is enabled. These options are discussed in detail below.

MIDI Input and Output Ports

Select the *MIDI input* and *MIDI output* ports that are connected to your sampler from the drop-down lists. If *Open loop* (discussed later) is being used for sending samples, then the *MIDI input* port can optionally be set to *(None)*. If *Open loop* is being used for receiving samples, then the *MIDI output* port can optionally be set to *(None)*. If *Open loop* is not being used, then you must supply both a *MIDI input* and *MIDI output* port.

The MIDI Channel

Now select the *MIDI channel* to use when transferring samples. This channel must match the channel configured in your sampler for SDS transfers.

A channel must be specified even if the sampler is set to Omni (any channel) mode.

Handshaking

The last few options determine how the sampler and the *Sampler Tool* interact while transferring samples. These options are *Wait for request* (send) and *Open loop or Send request* (receive) and *Open Loop*.

Open loop is a term used to describe a unidirectional communication protocol much like a one-sided conversation. When the *Open loop* option is enabled, the data source will send all data to the destination without listening for directions 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 communication makes open loop transfers prone to errors.

If the *Open loop* option is not enabled, we refer to the communication protocol as closed loop. A closed loop allows information to flow in both directions; the two communicators are said to be handshaking because they both agree on how and when data is transferred. When using closed loop transfers, the data is sent from the source in small packets (one packet at a time). The destination receives the packet and keeps the data if it is valid or discards the packet and asks for the data to be re-sent. The source will not send the next packet until the destination asks for it. This form of communication makes closed loop transfers much more reliable than open loop transfers.

It should be noted that open loop transfers are not only more prone to errors, but they can be slower than closed loop transfers, especially when sending samples with the *Sampler Tool*. The reason for this is due to intentional delays placed between data packets. These small delays allow slow samplers to keep up with the data being sent, and since these delays are not fine-tuned to exact requirements for the sampler, idle time is bound to occur. A closed loop transfer guarantees the most efficient timing between packets making transfers faster than open loop transfers.

Beware of using the *Open loop* option to receive samples from a sampler. Since the *Sampler Tool* cannot control the flow of data packets being sent, there is a high probability that data will be missed. Using Closed Loop transfers guarantees that no packets will be missed.

You may be asking why *Open loop* is an option if it is so unreliable. Good question. The answer is that there are cases when using *Open loop* is either necessary or more convenient than using closed loop. If a sampler does not support closed loop (handshaking), then you must use *Open loop* for transfers. You may also want to enable *Open loop* if you are out of MIDI cables or are having problems with a MIDI port. See the section titled MIDI/SDS Requirements and Troubleshooting for further details on open loop transfers.

If you choose to use the recommended closed loop transfer, then you also have the option of determining how a sample transfer is initiated. For sending samples, you can specify that the *Sampler Tool* should wait for the sampler to request the sample transfer before sending the sample by enabling the *Wait for request* option. This means that the *Sampler Tool* will not send the sample (after you press *Send Sample*)

until the sampler requests the sample to be sent. This usually requires you to press a button on the sampler to start the transfer.

For receiving samples, you can specify that the *Sampler Tool* should send a request to the sampler asking for a sample to be sent by enabling the *Send request* option. This means that the *Sampler Tool* will send a request for the specified sample to the sampler as soon as you press *Get Sample*. If you do not enable the *Send request* option, you must initiate the sample transfer from the sampler after you press *Get Sample*.

Typical Setup for MIDI/SDS Transfers

Send to Configuration (typical)

Connect and select a *MIDI input* and *MIDI output* device. Use your preferred *MIDI channel*. Disable *Open loop*. Disable *Wait for request*.

Receive from Configuration (typical)

Connect and select a *MIDI input* and *MIDI output* device. Use your preferred *MIDI channel*. Disable *Open loop*. Enable *Send request*.

The *Sample bias* fields will be discussed in the *Selecting a Sample Bias for your Sampler* section later in this chapter.

See also

[Using the Sampler Tool](#)

SCSI/SMDI configuration options

If you choose a sampler that uses the SCSI/SMDI transfer protocol in the *Send to* or *Receive from* drop-down lists, then you need to specify two other options. These options are the *SCSI host* and *SCSI sampler*.

The *SCSI host* drop-down list contains all *Sampler Tool* compatible SCSI cards found on your computer. You need to select the SCSI card that is connected to your sampler from this list. Note that most computers will have only one SCSI card listed.

Once you have selected the SCSI host, the *SCSI sampler* list will contain all SCSI devices connected to that host. Select your sampler from this list (the number preceding the device name is its SCSI ID). Depending on your SCSI setup, you may notice that hard drives, CD-ROM's and other SCSI devices are listed along with your sampler. This is to help troubleshoot SCSI ID conflicts should one occur. If you accidentally select a device other than a sampler, no harm is done. However, a message box will be displayed stating the problem if you try to send or receive samples from any device other than a sampler.

If your SCSI card or sampler does not show up in the *SCSI host* or *SCSI sampler* drop-down lists, refer to the section titled SCSI/SMDI Requirements and Troubleshooting to determine the cause. The problem may be as simple as turning on your computer and sampler in the wrong order. It may also be that your SCSI card is not compatible with the *Sampler Tool* or your SCSI card and sampler are installed incorrectly.

The above picture shows a typical configuration for the *Kurzweil K2000* (SCSI/SMDI) transfers. The fields that are grayed are for MIDI/SDS transfers only.

See also

[Using the Sampler Tool](#)

Selecting a Sample Bias for your Sampler

Those of you that have worked with samplers from different manufacturers know that the sample numbering scheme can vary dramatically between samplers. In an effort to simplify the confusion caused by working with different samplers, the *Sampler Tool* lets you specify a sample bias. A sample bias is simply a value that gets added to the logical sample number to derive the actual sample number used for sending or receiving. Don't panic, we'll explain everything.

When you want to send or receive a sample from a sampler, you must specify a sample number which is the location that the sample is stored within the sampler. This is pretty simple. However, not all samplers allow you to use any sample number for storing and retrieving samples.

Samplers that have factory-defined samples burned into their ROMs usually define a range of sample numbers for these samples. This range cannot be used to store and retrieve your own samples. These samplers will define one or more sample number ranges that are available for your own samples. For example, the Kurzweil K2000 uses sample numbers 1 through 199 as factory-defined samples and does not allow access to sample number zero. Sample numbers 200 through 999 are available for user-defined samples. Also note that a sample number of 200 specified in the *Sampler Tool* actually addresses sample number 201 on the Kurzweil K2000. (Confusing, isn't it?)

The Akai S1000, on the other hand, uses sample numbers 0 through 3 to store its factory-defined samples. Sample numbers above 3 are available for user-defined samples. And the list of inconsistencies goes on and on.

Wouldn't it be nice if you could simply use sample numbers 0 through n for your samples no matter what sampler you are using? This is what the *Sample bias* allows you to do. Once you figure out what sample number range is available for your custom samples, and determine whether the sampler counts in the same manner as the *Sampler Tool*, you can enter the sample bias and forget about it.

For the above examples, a sample bias of 199 works perfectly for the Kurzweil K2000 (remember it adds one, so sample number 199 actually addresses sample number 200), and a sample bias of 4 makes the Akai S1000 a bit easier to manage. Once you have entered the sample bias for your sampler, you can use sample numbers 0 through n as the *Logical send/receive sample number* and the *Sampler Tool* will convert it to the *Actual send/receive sample number* automatically.

You can also use the sample bias to define different bases for multiple projects on which you are working. For example, if you are composing two different pieces that use different samples, you might want to create a sampler configuration for each of these projects (see the Saving Sampler Configurations section below). Within each configuration, you can define a different sample bias to keep your samples separated in the same sampler.

If you are still a little fuzzy on how the *Sample bias* option works, don't sweat it. As soon as you use it once or twice, it'll become old hat. Try different bias values to see what happens (do this before you start working on a critical project). The worst that can happen is you will overwrite a sample that you didn't mean to, and all writeable samples should be restorable from your disks. The section titled Sending and Receiving Samples also contains more information on using the *Sample bias*.

See also

[Using the Sampler Tool](#)

Saving Sampler configurations

After filling in all options for a sampler configuration, you can save it as a preset by selecting the *Save As* button in the *Sampler Configuration* dialog. Type a descriptive name into the *Save Preset* dialog and press *OK*. Once a sampler configuration preset is saved, you can choose the configuration by name from the *Sampler* dialog.

The following picture shows a configuration being saved for the *Kurzweil K2000 (SCSI/SMDI)*. Note that a preset name can be anything you want, so use whatever names with which you are most comfortable.

You can create and save as many sampler configuration presets as you want. You might want to create configurations for all of your samplers and presets for specific projects you are working on. Having presets for each of your samplers and projects allows you to quickly switch back and forth without reconfiguring the *Sampler Tool*.

If you want to get rid of a preset, select it from the *Name* drop-down list at the top of the *Sampler Configuration* dialog and press the *Delete* button.

See also

[Using the Sampler Tool](#)

Sending and Receiving Samples

Once your sampler setup is configured, you should be able to send and receive samples by using the *Send Sample* and *Get Sample* buttons in the *Sampler* dialog. Select the sampler you want to *Send to* or *Receive from* in the *Configuration* drop-down list, enter the sample number you want to transfer in the *Logical send/receive sample number* field, and press the *Send Sample* or *Get Sample* button.

Note that the *Actual send sample number* and *Actual receive sample number* are equal to the *Logical send/receive sample number*. This is because no sample bias is being used for this particular configuration.

Other areas of interest are the *Send to* and *Receive from* descriptions where all information relevant to the sampler configuration is printed. The fields on the bottom called *MIDI unity note* and *Fine tune* display the current values for the active sample. The *C5 (60)* displayed for the *MIDI unity note* represents that the sample is tuned for Middle C. The *Fine tune* value is set to *0.0 cents*, so no minor tuning differences exist. These values are set using the *Edit Sample* command under the *Special* menu and may be used by samplers that support tuning information for samples. It is important to note that Sound Forge does not use this information, it is for samplers only.

As another example, the following picture shows a setup ready to send or receive sample number 199 (which is really 200) from a Kurzweil K2000 using SCSI/SMDI through an Adaptec AHA SCSI card.

The most notable difference between this example and the previous one is the use of a Sample bias of 199 for the send and receive sample numbers.

To send or receive a sample after selecting the sampler setup from the *Configuration* drop-down list, you need only press the *Send Sample* or *Get Sample* button. The transfer will begin and a *Progress Meter* will be displayed in the status bar of Sound Forge. You can cancel a transfer at any time by pressing the *Close* button to the left of the *Progress Meter* or by pressing the *Escape* key.

Sending a sample will always send the entire contents of the active data window. Any selection contained in the waveform display is ignored. When receiving a sample, the entire contents of the active data window are replaced with the new sample data (a warning will be displayed before any data is replaced).

If you're having trouble, verify that the sampler is set up correctly for sending or receiving samples by referring to the sampler's documentation on sample dumps. For further information on using your sampler with the *Sampler Tool*, refer to the Appendix at the end of this manual. These sections contain detailed information about several samplers that have been used successfully with the *Sampler Tool*. You may want to read some or all of these sections even if your sampler is not mentioned. There are tips you may find helpful when working with your sampler.

See also

[Using the Sampler Tool](#)

MIDI/SDS hardware requirements and setup

To use the MIDI/SDS protocol with an external sampler that supports the MIDI Sample Dump Standard (SDS), you need a MIDI card (with MIDI input and output ports) installed in your computer.

Using MIDI cables, connect the MIDI out port of your sampler to the MIDI in port of the card and the MIDI in port of the sampler to the MIDI out port of the card. This is the same configuration you would use to connect a MIDI keyboard to your computer for sequencing.

Internal samplers do not require a MIDI card and MIDI cables. However, you may need to use open loop protocol when sending samples to an internal sampler. The documentation that comes with your sampler should specify the requirements for performing SDS transfers if it is supported.

MIDI/SDS troubleshooting

When using MIDI/SDS to send or receive samples, it is usually in your best interest to **not** use the *Open loop* option. The *Open loop* transfer is slower and has a much greater chance of sending or receiving incorrect data. Reasons for using *Open loop* are:

- Your sampler does not support closed loop transfers (handshaking).
- You don't have enough cables to connect both the MIDI in and MIDI out ports.
- Open loop can also help with troubleshooting SDS hardware setup problems. If you can't get the *Sampler Tool* to transfer data to (or from) your sampler, it may be helpful to try turning on the *Open loop* option to see if single cable transfers work. If open loop works but closed loop does not, you probably have one of the following problems.
- Your sampler does not support closed loop transfers (handshaking).
- Your MIDI cable or connection for MIDI input (send) or MIDI output (receive) to your computer is bad.
- Your MIDI card is not receiving MIDI input (send) or sending MIDI output (receive). Interrupt conflicts are a common problem for MIDI input.

See also

[Using the Sampler Tool](#)

SCSI/SMDI hardware requirements and setup

To use the SCSI/SMDI protocol with an external sampler that supports the SCSI MIDI Device Interface (SMDI) under Windows 3.x, you must have an ASPI compatible SCSI card for your machine as well as the WINASPI software provided by Adaptec. The best way to acquire this setup is to buy an Adaptec SCSI controller with the EZ-SCSI software. The EZ-SCSI software is required by the Sampler Tool for accessing your Adaptec SCSI card.

Windows 95 includes ASPI drivers, therefore you will only need a compatible SCSI adapter.

If you own an older Adaptec card and it did not come with EZ-SCSI, you will need to contact your Adaptec dealer for information on purchasing EZ-SCSI. You may also contact Adaptec Pre-Sales directly at 1 (800) 959-7274.

If you do not own a SCSI card and are looking for an inexpensive solution for transferring samples to your SMDI compatible external sampler, the Adaptec AVA-1505 TotalCD kit is an excellent choice. This kit comes with EZ-SCSI Lite which provides everything you need for using the *Sampler Tool* with an external sampler (except a SCSI cable).

Cabling

To connect your SCSI card to your external sampler, you will need to use a SCSI cable. There are several types of SCSI cables, so make sure you get the right one for your SCSI card and sampler.

There are three types of connectors found commonly on external SCSI cables. One type is the Centronics 50 pin. This connector is the beast of connectors and if you own an Adaptec card other than an AVA-1505 or AVA-1515, this is probably the connector you will need to connect to your SCSI card. If you own an AVA-1505 or AVA-1515, the connector on your SCSI card is a DB-25 connector. This connector is a 25-pin connector and looks like the connector found on serial and parallel cables. The DB-25 connector is most likely used by your sampler as well. The third type of connector is the SCSI-2 type connector that is used with the popular Adaptec 2940 series of PCI SCSI adapters. This connector looks similar to a DB-25, but has smaller pins and many more of them.

If you are buying a cable to connect your sampler to your system, it is important that you get the terminology right when ordering. You will need to know the type of connector found on your SCSI adapter, as well as the connector on the sampler itself. Most samplers have DB-25 connectors and most SCSI adapters these days use Centronics or SCSI-2 connectors. Determine what your system uses and order appropriately.

Warning: Do not use a serial, modem, or printer cables to connect your SCSI devices.
You may damage your equipment by using these cables.

Once you have a SCSI card installed (refer to the card's manual for installation instructions), connect the card and the sampler's SCSI ports with a SCSI cable. Both your computer and your sampler must have the power turned off before connecting or disconnecting SCSI cables! Failing to turn the power off before connecting and disconnecting SCSI cables can damage your computer and/or hardware. SCSI connections are much more fragile than MIDI connections.

See also

[Using the Sampler Tool](#)

SCSI/SMDI troubleshooting

The following is a list of common problems encountered with SCSI and samplers

[Conflicting SCSI IDs](#)

[Occasional Transfer failures](#)

[Sampler is recognized but transfers are unreliable](#)

[Adaptec 1540/1542CF Doesn't Recognize a Sampler](#)

[SCSI/SMDI Menu Doesn't Show Under Windows 95](#)

See also

[Using the Sampler Tool](#)

Conflicting SCSI ID's.

When connecting devices on a SCSI chain, each device must have a unique device identifier (ID). SCSI allows you to set up to eight unique ID values from 0 to 7. Typically device ID 7 is used for the internal SCSI controller card which leaves ID's 0 through 6 available for other devices. Also, if you have a bootable SCSI hard drive its ID must be set to 0.

Typical SCSI configuration:

ID	Device
0	Hard Drive
1	CD-ROM Drive
2-6	Available for use by samplers
7	SCSI Controller Card

See also

[Using the Sampler Tool](#)

[SCSI/SMDI Troubleshooting](#)

Occasional Transfer Failures.

If you occasionally receive messages such as, *The SCSI Device is not responding* or *A problem was encountered while transferring the sample*, you may just have an upset SCSI bus. Try bringing up the *Sampler Configuration* dialog and reselecting your SCSI host from the drop-down list. Doing this causes a series of SCSI commands to be executed that in many cases will settle an upset SCSI bus. If this fails to work, power down and restart all equipment (computer, samplers, etc.).

If the above mentioned messages are encountered frequently, you may have a more serious problem. The following section discusses several possibilities.

See also

[Using the Sampler Tool](#)

[SCSI/SMDI Troubleshooting](#)

The sampler is recognized but doesn't transfer reliably.

The following is a list of causes for unreliable SCSI transfers.

Synchronous Transfer Mode

Some samplers (like the Kurzweil K2000) 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 often have the option of using a synchronous transfer mode. If there is a host vs. device synchronous transfer option, make sure you select host. Refer to your SCSI device manuals for more information.

SCSI Termination

If your SCSI chain is not properly terminated, you may experience unreliable SCSI transfers. Refer to your SCSI card and SCSI device manuals for more information.

Long or Cheap SCSI Cables

SCSI cables that are very long or not properly shielded can cause unreliable operation. Keep in mind that SCSI cables are not as forgiving as MIDI cables and overly long cables will not work reliably. Also beware of using cables that are not certified SCSI cables (such as tantalizingly cheap parallel printer cables); the extra money spent on a true shielded SCSI cable can save you endless hours of frustration.

See also

[Using the Sampler Tool](#)

[SCSI/SMDI Troubleshooting](#)

Adaptec 1540/1542CF Doesn't Recognize a Sampler

If you own the Adaptec 1540/1542CF (not the C version, the newer CF version) and it won't recognize your sampler you may need to make a change in the configuration of your Adaptec controller.

Some samplers do not want to operate when the *Reset SCSI Bus at Power-On* option of the Adaptec controller is enabled. This is the default operation for the 1540/1542CF and you will have to disable this option to allow your system to work with your sampler.

See also

[Using the Sampler Tool](#)

[SCSI/SMDI Troubleshooting](#)

[Disabling the Reset SCSI Bus option](#)

Disabling the Reset SCSI Bus option

You should review your 1540/1542CF owners manual before proceeding with this change.

1. Reset your machine by pressing the hardware reset button or turning the machine off and then back on.
2. While your system is booting it will display the message, *Press Control+ A for the SCSISelect Utility*. Press *Control+A* when this message appears.
3. The screen will display the main screen of the *SCSISelect* utility and show the current IO port in use by the SCSI card. Press *Enter* to proceed to the Options menu.
4. In the Options menu press *Enter* on the *Configure/View Host Adapter Settings* option to proceed to the *Settings* screen.
5. Move the cursor using the arrow keys until the *Advanced Configurations Options* item is selected. Press the *Enter* key to proceed to the *Advanced Options* screen.
6. Move the cursor using the arrow keys until the *Reset SCSI Bus at Power-On option* is selected. Press *Enter* to change this option.
7. Select the *Disabled* setting by moving the selection with the arrow keys, and then press *Enter*.
8. Press the *Escape* key twice to return to the previous screens.
9. Answer *Yes* to the *Save Changes Made?* prompt.
10. Press the *Escape* key again to exit the *SCSISelect* utility and answer *Yes* when the *Exit Utility* prompt is displayed.
11. *SCSISelect* will reboot your machine after you have made the changes.

Note: Disabling the *Reset SCSI Bus at Power-On* option may keep other devices on the SCSI chain from resetting correctly when using the *Soft Boot* feature of your machine (i.e. pressing *Control+Alt+Del*). On some systems this will cause the system to hang when pressing *Control+Alt+Del*. To guarantee that devices are reset when rebooting with this option disabled, you should use the *Reset* button on your computer or turn the power off and back on to reset the machine.

See also

[Using the Sampler Tool](#)

[SCSI/SMDI Troubleshooting](#)

SCSI/SMDI Menu Doesn't Show Under Windows 95

The first thing to do is to check for correct SCSI termination and multiple devices on the SCSI chain using the same SCSI ID. If you've already done this and have an Adaptec SCSI card it is possible that you need to update the mini-port drivers in your system.

Adaptec has a series of updated mini-port drivers available for Windows 95 on both CompuServe and on the world wide web.

To Find the Drivers on CompuServe

1. Log on to CompuServe and type in GO ADAPTEC. (With WinCIM press the GO button and type in ADAPTEC.)
2. In the library section labeled Windows 95 download the file WIN95MPD.EXE

To Find the Drivers on the Web

1. Go to the Adaptec site at <http://www.adaptec.com>.
2. Navigate to the support section which contains updated drivers or you can go directly to the files page at the web address http://www.adaptec.com/techsupp/BBS_95.html
3. Click on the file name WIN95MPD.EXE to download the updated driver package.

Updating Your Drivers

After you have downloaded the file WIN95MPD.EXE you need to create a temporary directory or folder on your system and run the WIN95MPD.EXE program. This program is a self-extracting archive which creates a series of mini-port drivers (all identified by the extension *.mpd*) and a readme.txt file which explains how to install the new drivers.

Follow the directions in the readme.txt file to update your drivers for your Adaptec SCSI card.

See also

[Using the Sampler Tool](#)

[SCSI/SMDI Troubleshooting](#)

Sampler Tips and Ideas

[Using FM synthesis with your sampler](#)

[Change the games but not the rules](#)

[Back-up and catalog your samples](#)

See also

[Using the Sampler Tool](#)

Using FM synthesis with your sampler

One nice use for the **FM Synthesis** Tool in Sound Forge is to Turn your sampler into an FM Synth! Well, not quite. This procedure will not turn your sampler into a true FM synthesizer, or at least the types that have become most familiar. While Sound Forge will produce four operator sounds, this is still sampling. To get different pitches across the keyboard, the sampler will vary the sample rate of the original sample rather than produce a unique timbre based on the frequencies dictated by that pitch (key).

Also, any dynamic envelope and/or velocity control of the FM is not variable like that of true FM synthesizers. Nonetheless, sending samples to your sampler that were produced using the **FM Synthesis** tool of Sound Forge can yield some very cool results. If you have a sampler with synthesis capabilities, this technique becomes quite powerful.

1. First, create a sample using the **FM Synthesis** tool found in the **Tools** menu. The sample can be rather short (0.2 - 1.0 sec) and it is probably a good idea to make sure that some portion of the envelope is static so that a good loop is easy to attain.
2. Once you have created the sample, create a sustaining loop and edit it with the Loop Tuner.
3. When you have your sample the way you want it, follow the **sampler** tool procedure to transfer the sample to your sampler.
4. From there any further manipulation will need to take place inside your sampler. That's about it! Simple but effective.

See also

[Using the Sampler Tool](#)

Change the game but not the rules

It's easy to see that you can expand the processing power of a sampler by using **Get Sample** to retrieve a sample from the sampler, processing it with Sound Forge, and then using **Send Sample** to send the sample back to the same location in the sampler. This in itself is no mind blowing discovery. However, there may be some inspiring side effects waiting to be uncovered.

Most samplers will replace samples already in memory locations with new samples received from Sound Forge if the **Logical send/receive sample number** matches the original location. Other samplers will automatically put a received sample in an empty location regardless of the sampler number. The following idea is only possible with a sampler that replaces a sample.

When the Kurzweil K2000 receives a sample via SCSI/SMDI, it will *replace* samples that have the same incoming send sample number (this is not true if you are using MIDI/SDS with the K2000). The interesting part of this is *all Keymaps that reference that sample location will remain intact* even if the sample has been replaced with an entirely different one. This can lead to some very unique and sometimes spine tingling sounds.

See also

[Using the Sampler Tool](#)

Back up and catalog your samples on your computer

Using the Sampler Tool, you can use your computer to store and catalog samples. Saving your samples as WAV files on your computer's hard drive or normal floppy disks allows you to store detailed Summary Information with each sample. Summary Information can include descriptive titles, subject details, general comments, and more.

Unfortunately, Keymaps and other sampler specific information are not transferred from the sampler to the Sampler Tool, so none of it gets saved with WAV files. If this is the case with your sampler and you still want to save your samples as WAV files, you might be able to save Keymaps and other information *without* the sample data on a floppy disk. By saving this special information without the sample data (or a very small sample if the sampler requires it), you could store quite a few on a singly floppy disk. Later, the Keymaps can be recombined with the sample data stored on your computer.

See also

[Using the Sampler Tool](#)

Akai S1000 - MIDI/SDS procedure

The S1000 is capable of sending and receiving samples via MIDI/SDS in all methods implemented by Sound Forge. Provided that your MIDI connections are correct, the S1000 is basically ready to receive or send a sample.

A few things to keep in mind:

- ◆ The S1000 structures its sample memory in such a way that the **Logical send/receive sample number** in Sound Forge can be a little confusing. The problem is that the S1000 will *name* the newly received sample according to the **Logical send** number specified in Sound Forge. However, in the S1000, a new sample is placed into the *next* available memory location, not the **Logical send** location.
- ◆ For example, if you send the S1000 a sample with a **Logical send number** of 0, send another with the **Logical send number** of 2, and then send yet another with the **Logical send number** of 1, the S1000 will now have three samples next to each other in memory, one named MIDI 0 another named MIDI 2 and another named MIDI 1. Now, if you wanted to use Sound Forge's **Get Sample** to retrieve these from the S1000, MIDI 0 would have to be referred to as **Logical receive number 4**, MIDI 2 would be **Logical receive 5**, and MIDI 1 would be referred to as **Logical receive number 6**. What's going on here?
- ◆ First of all, the S1000 has four waveforms permanently stored in memory location 0-3; sine, square, sawtooth, and pulse. The S1000 doesn't necessarily refer to these waveforms as being at these locations. We are referring to them as such because if you wanted to **Get sample** from Sound Forge, these are the **Logical receive numbers** you would use to get these waveforms. The S1000 is simply taking the **Logical send** number and using it as a naming convention for received samples. However, the **Logical send** number has no bearing on where it puts the sample.
When the S1000 receives a new sample from Sound Forge, it puts it into its next available memory location, deriving its name from the **Logical send** number. In our example, it put our first sample sent in location number 4 and named it MIDI 0, the next one in location number 5 and named it MIDI 2, and finally put MIDI 1 in location number 6. So bear in mind that when invoking **Get Sample** in Sound Forge, the **Logical receive sample number** does not necessarily correspond to the name of the sample in the S1000.
- ◆ The S1000 will replace any sample in its memory with an incoming sample of the same name. That is, if you have sent a sample into the S1000 with Logical send number 2 and the S1000 has named this MIDI 2, then if you send another sample and do not change the Logical send number, the original MIDI 2 will be replaced with a new MIDI 2.
- ◆ In order to audition the sample from a MIDI source (such as the Sound Forge keyboard), it will be necessary to go the S1000 **EDIT SAMPLE** page and select the sample that you want to hear. You will need to be on this page in order to audition a sample that has not yet been incorporated into a program.
- ◆ While the S1000 is receiving the sample, the MIDI LED will flicker indicating that the transfer is in process.
- ◆ The S1000 supports SDS handshaking. Therefore, any method of MIDI/SDS transfer will work. It is recommended that you turn **Open Loop** off and **Send request** on. Be sure that the MIDI IN and OUT are connected between the S1000 and Sound Forge to use **handshaking** (Closed loop).
- ◆ On the S1000's **MIDI EXCL** page, set the **type of transmission** to **SINGLE SAMPLE** and **sample protocol** to **STANDARD**.

To transfer MIDI/SDS to the S1000 from Sound Forge

1. Having set the necessary Start/Sustain loop/Release loop points within Sound Forge. Select **Sampler** from the **Tools** menu.
2. Select **Configure**.
3. In the **Send to** section, select Akai S1000 (MIDI/SDS).
4. In the **MIDI output** section choose the MIDI driver that is connected the S1000's MIDI input.
5. Now select the MIDI Channel. (This should match the MIDI channel found in the S1000's MIDI/EXCL page).

6. Hit OK.
7. Hit **Send Sample**. The S1000 should now receive the sample.

To transfer MIDI/SDS to Sound Forge From the S1000

1. Having set the necessary Start/Sustain loop/Release loop points within the S1000. Select **Sampler** from the **Tools** menu.
2. Select Configure.
3. In the Receive from section, select Akai S1000 (MIDI/SDS).
4. In the MIDI output section choose the MIDI driver that is connected to the S1000's MIDI input.
5. In the MIDI input section choose the MIDI driver that is connected to the S1000 MIDI output.
6. We recommend that you select **Send request (MIDI/SDS)**.
7. Now select the MIDI Channel. (This should match the MIDI channel found in the S1000's MIDI/EXCL page).
8. Hit OK.
9. Hit **Get Sample**. Sound Forge should now receive the sample.

See also

[Using the Sampler Tool](#)

Digidesign SampleCell II - Sample transfer procedures

The SampleCell II is capable of receiving samples via internal interaction between Sound Forge and the SampleCell software.

A few things to keep in mind:

- ◆ **You must have SampleCell software version 1.02 or above to transfer samples from Sound Forge to your SampleCell II card.** If you have an earlier version contact Digidesign to acquire an update.
- ◆ At present, Digidesign does not provide Windows NT compatible software. Transferring samples to the SampleCell II is therefore not supported in Windows NT.
- ◆ The SampleCell can only receive samples from Sound Forge. You can not retrieve samples from the internal memory of the SampleCell. This is usually not a problem since the SampleCell does not allow you to record new samples. All samples in its memory are supplied from files you already have on your computer
- ◆ Sending a sample to the SampleCell from Sound Forge will replace any previous samples sent by Sound Forge.
- ◆ In order to audition the sample from a MIDI source (such as the Sound Forge keyboard), it will be necessary to set the MIDI output device to SampleCell 1. If you have more than one SampleCell the number will be adjusted accordingly for each card in your system.
- ◆ You must be running the SampleCell application software to transfer samples from Sound Forge to the SampleCell.
- ◆ All samples sent to the SampleCell from Sound Forge have the name FRGSCNEW.
- ◆ The SampleCell ignores the Sample Number and Bias fields used by Sound Forge.

To transfer samples to the SampleCell from Sound Forge

1. Start the SampleCell software by double clicking on the Sample Cell icon.
2. Having set the necessary Start/Sustain loop/Release loop points within Sound Forge. Select **Sampler** from the **Tools** menu.
3. Select Configure.
4. In the Send to section, select **Digidesign SampleCell II**. You do not have to set any other options in the dialog. They are ignored by the SampleCell.
5. Hit OK.
6. Hit **Send Sample**. The SampleCell should now receive the sample.

When transferring samples to the SampleCell, a temporary WAV patch file is created prior to sending the data to the SampleCell. You should not save instruments or patches based on files transferred from Sound Forge to the SampleCell, as the SampleCell bases its instruments on the original WAV file name. The downloading of samples from Sound Forge is meant to be used to create and tune single WAV sounds within an instrument. If you save an instrument based on a temporary file created by Sound Forge, when you attempt to reload the instrument at a later time the SampleCell software will not be able to find the temporary file.

When creating samples for the SampleCell, do as much work as you can in Sound Forge. Download the sample to the SampleCell to listen to how it sounds on the SampleCell. If you are happy with your work ,save the file using the Save As item from the File menu and then load the new WAV file from SampleCell software to create new instruments.

See also

[Using the Sampler Tool](#)

Kurzweil K2000 - MIDI/SDS procedure

The K2000 is capable of receiving a sample via MIDI/SDS. The K2000 will not send a sample via MIDI/SDS. In its manual there is mention of the K2000 bulk dump. This protocol is not supported by Sound Forge. Therefore, the Sound Forge Get Sample procedure does not work with the K2000 for MIDI/SDS transfers.

Sending samples to the K2000 via MIDI/SDS is really quite simple. Provided that your MIDI connections are correct, the K2000 is basically ready to receive a new sample. The K2000 can be sitting on any screen and will actually operate normally while the transfer is going on in the background.

A few things to keep in mind:

- ◆ The K2000 will automatically assign the incoming sample to the next available - Sample/RAM location if the **Logical Send/Receive** sample number is left at 0. This location is most likely going to be in the 200-299 range (unless these are already filled). If you choose a **Logical Send/Receive** sample number of 0-199, the K2000 will automatically add 200 to the selected number (plus any necessary to reach the next available sample number). If you choose a number of 200 or more, the K2000 will add 1 (again, plus any necessary to reach the next available sample number).
- ◆ In order to audition the sample from a MIDI source (such as the Sound Forge keyboard), it will be necessary to create a Keymap within the K2000. This is done by hitting Preview in the Master/Sample page or by calling up the sample within the Keymap section of a pre-existing program.
- ◆ While the K2000 is receiving the sample, the MIDI LED will flicker indicating that the transfer is in process.

To transfer MIDI/SDS to the K2000 from Sound Forge

1. Having set the necessary Start/Sustain loop/Release loop points within Sound Forge. Select **Sampler** from the **Tools** menu.
2. Select Configure.
3. In the Send to section, select Kurzweil K2000 (MIDI/SDS).
4. In the MIDI output section choose the MIDI driver that is connected to the K2000's MIDI input.
5. Now select the MIDI Channel. (This should match the Basic MIDI channel found in the K2000's MIDI/RECV page).
6. Hit OK.
7. If you now hit Send Sample, the K2000 will start receiving the data.
8. To examine the new sample within the K2000, simply go to the Master/Sample page of the K2000. The sample that appears there as a default should be your newly received sample. Hit EDIT, and look to see that the waveform appears in the K2000's screen. Hitting preview from the Master/Sample page will create a new program based on this sample and will allow you to audition the sample from an external MIDI source.

See also

[Using the Sampler Tool](#)

Kurzweil K2000 - SCSI/SMDI procedure

Sound Forge is capable of sending and receiving samples via SCSI/SMDI to the Kurzweil K2000. This method is very fast and very simple once you have configured the SCSI hardware properly.

A few things to keep in mind:

- ◆ Unlike in MIDI/SDS sample transfers, the K2000 will not automatically adjust for sample number locations (adding 200, etc.). There are two ways of handling this fact;
 1. You can select the specific Logical send/receive sample number as you would like it to appear in the K2000. Since the K2000 will add 1 to whatever number you have entered here, set the Sample bias in the Configure menu to -1.
 2. If you set the Sample bias to 199, you can then refer to the Logical send receive sample number as 0 through x. This will make the Sound Forge send of sample number 0 locate to the K2000 at 200, sample number 1 locates to 201 and so on.
- ◆ When sending a sample to a K2000 RAM location that is already filled, the K2000 will not warn you that the location is already filled. Instead, it will replace the sample with the new one being sent. Be very careful when doing a lot of sample transfers (SCSI/SMDI), so that you do not wipe out your hard work by neglecting to select a new Logical send/receive sample number for each separate sample.
- ◆ When Choosing SCSI/SMDI in the Sampler dialog or if it was last set to SCSI/SMDI, Sound Forge will verify that the SCSI bus is operational and may therefore take a second or two to respond to any user interaction.
- ◆ When the SCSI/SMDI method is selected in the Configuration dialog, you only need to be concerned with setting the SCSI host and SCSI sampler. Any SCSI device that appears in these lists, is being recognized by Sound Forge and should work correctly.
- ◆ Make sure that your SCSI adapter is supported by Sound Forge for sample transfer procedures. If you are unsure as to whether your adapter is supported for SMDI transfers with Sound Forge, see **SCSI/SMDI Setup and Troubleshooting** at the beginning of this section.
- ◆ The K2000's SCSI ID is set in its MIDI/RECV page.

To transfer SCSI/SMDI to the K2000 from Sound Forge

1. Having set the necessary Start/Sustain loop/Release loop points within Sound Forge. Select **Sampler** from the **Tools** menu.
2. Select Configure.
3. In the Send to section, select Kurzweil K2000 (SCSI/SMDI).
4. In the SCSI host section, select your SCSI adapter.
5. In the SCSI sampler section, choose the K2000 connected to your computer's SCSI bus.
6. Save your configuration if you like and/or hit OK.
7. Check to see that your Configuration settings are confirmed without error and now hit Send Sample. The K2000 will receive the sample very quickly.
8. To examine the new sample within the K2000, go to the Master/Sample page of the K2000. The sample that appears there as a default should be your newly received sample. Hit EDIT, and look to see that the waveform appears in the K2000's screen. Hitting preview from the Master/Sample page will create a new program based on this sample and will allow you to audition the sample from an external MIDI source.

To transfer SCSI/SMDI from the K2000 to Sound Forge

1. Having set the necessary Start/Sustain loop/Release loop points within the K2000. Select **Sampler** from the **Tools** menu.
2. Select Configure.
3. In the Receive from section, select Kurzweil K2000 (SCSI/SMDI).
4. In the SCSI host section, select your SCSI adapter.
5. In the SCSI sampler section, choose the K2000 connected to your computer's SCSI bus.
6. Save your configuration if you like and/or hit OK.

7. Check to see that your Configuration settings are confirmed without error and now hit Get Sample (remember to select the specific Logical send/receive sample number minus 1, unless you have adjusted the Sample bias). Sound Forge will receive the sample very quickly.
8. When the transfer is complete, the sample will appear in the active sample window.

Known problems with the K2000

The following is a list of known problems with transferring samples to K2000:

- ◆ **The K2000 reboots when in the Master/Sample page:** This is a bug in the K2000's ROM. The K2000 will reboot if you transfer a sample to the same sample number twice in a row while the K2000 is on the **Master/Sample** page. The best way to get around this problem is to not transfer samples while you are using the **Master/Sample** page of the K2000.

See also

[Using the Sampler Tool](#)

Peavey SP - MIDI/SDS procedure

The Peavey SP is capable of sending and receiving samples via MIDI/SDS in all configuration methods implemented by Sound Forge.

Provided that your MIDI connections are correct, the SP is basically ready to receive a new sample. The SP can be sitting on any screen and will begin to receive the sample upon initiating the transfer from Sound Forge.

A few things to keep in mind:

- ◆ Make sure the SP's Get slave samp# is using: midi (sds) - this is located in the SP's Util menu.
- ◆ If you have set up the transfer configuration as a Wait for request, you will need to go to the SP's Util menu and instigate the request for sample dump. This is done by hitting Exec twice from the Util menu on the SP after telling Sound Forge to Send Sample.
- ◆ In order to audition the sample from a MIDI source (such as the Sound Forge keyboard), it will be necessary to go to the wave menu on the SP and select the appropriate wave number that you are trying to audition (the last received sample should appear on this screen).
- ◆ If you are sending a sample Open loop, it is not required that the sampler's MIDI out be connected to Sound Forge. However, it is recommended that you use handshaking (no Open loop) for faster and more reliable communications.

To transfer MIDI/SDS to the Peavey SP from Sound Forge

1. Having set the necessary Start/Sustain loop/Release loop points within Sound Forge. Select **Sampler** from the **Tools** menu.
2. Select Configure.
3. In the Send to section, select Peavey SP (MIDI/SDS).
4. In the MIDI output section choose the MIDI driver that is connected to the SP's MIDI input.
5. Select the MIDI Channel (this should match the omni/poly channel found in the SP's Global menu, unless the SP is set to omni mode).
6. In Sound Forge hit the Send Sample button. The transfer will begin and both Sound Forge and the SP will display the amount of data left to go until the procedure is complete.
7. When the transfer is complete, you can hit the wave button on the SP and immediately audition the sample from the Sound Forge keyboard or whatever MIDI source is connected to the SP.

To transfer MIDI/SDS from the Peavey SP to Sound Forge

1. Having set the necessary Start/Sustain loop/Release loop points within the Peavey SP. Select **Sampler** from the **Tools** menu.
2. Select Configure.
3. In the Receive from section, select Peavey SP (MIDI/SDS).
4. In the MIDI input section choose the MIDI driver that is connected to the SP's MIDI output.
5. In the MIDI output section choose the MIDI driver that is connected to the SP's MIDI input.
6. Select the MIDI Channel (this should match the omni/poly channel found in the SP's Globl menu, unless the SP is set to omni mode).
7. In order for Sound Forge to automatically get the sample without invoking the transfer from the SP's front panel, it is necessary to turn Wait for request on in the Receive section of the **Sampler Configure** dialog. Otherwise, you will have to hit exec on the SP after hitting the Get Sample button in Sound Forge.
8. In Sound Forge, hit the Get Sample button. The transfer will begin and both Sound Forge and the SP will display the amount of data left to go until the procedure is complete.
9. When the transfer is complete, the sample will appear in the active Data Window.

See also

[Using the Sampler Tool](#)

Peavey SP - SCSI/SMDI procedure

Sound Forge is capable of sending and receiving samples via SCSI/SMDI to the Peavey SP. This method is very fast and simple once you have configured the SCSI hardware properly.

A few things to keep in mind:

- ◆ Make sure the SP's Get slave samp# is using: SCSI (SMDI). This is located in the SP's Util menu. Though the SP is capable of detecting what method is used with the incoming sample, it is highly recommended that the user select this manually as crashes can occur when first sending MIDI/SDS followed immediately by SCSI/SMDI.
- ◆ The SCSI cable from your computer must be connected to the SP's SCSI B port.
- ◆ When Choosing SCSI/SMDI in the Sampler dialog or if it was last set to SCSI/SMDI, Sound Forge will verify that the SCSI bus is operational and may therefore take a second or two to respond to any user interaction.
- ◆ When SCSI/SMDI method is selected, you only need to be concerned with setting the SCSI host and SCSI sampler in the Configuration dialog. Any SCSI device that appears in these lists, is being recognized by Sound Forge and should work correctly.
- ◆ Make sure that your SCSI adapter is supported by Sound Forge for sample transfer procedures. If you are unsure as to whether your adapter is supported for SMDI transfers with Sound Forge, see **SCSI/SMDI Setup and Troubleshooting** at the beginning of this section.
- ◆ The Peavey SP's SCSI ID is set by default to Device 0. This setting is made by means of jumpers within the unit itself. Every time the SP is powered on, it will reset itself to this setting. You can temporarily change the SCSI ID to another number from the SP's UTIL menu and will remain until the unit is powered off. If you find that you consistently want a SCSI ID other than 0, you should change the jumper settings inside the unit according to the directions in the Peavey SP manual. **If you have a bootable SCSI hard drive its ID will also be 0 so you must change the ID of the SP for proper operation.**

To transfer SCSI/SMDI to the Peavey SP from Sound Forge

1. Having set the necessary Start/Sustain loop/Release loop points within Sound Forge. Select **Sampler** from the **Tools** menu.
2. Select Configure.
3. In the Send to section, select Peavey SP (SCSI/SMDI).
4. In the SCSI host section, select your SCSI adapter.
5. In the SCSI Sampler section, choose the Peavey SP connected to your computer's SCSI bus.
6. Save your configuration if you like and/or hit OK.
7. Check to see that your Configuration settings are confirmed without error and hit Send Sample. The Peavey SP will receive the sample very quickly.
8. When the transfer is complete, you can hit the wave button on the SP and immediately audition the sample from the Sound Forge keyboard or whatever MIDI source is connected to the SP.

To transfer SCSI/SMDI from the Peavey SP to Sound Forge

Having set the necessary Start/Sustain loop/Release loop points within the Peavey SP. Select **Sampler** from the **Tools** menu.

1. Select Configure.
 2. In the Receive from section, select Peavey SP (SCSI/SMDI).
 3. In the SCSI host section, select your SCSI adapter.
 4. In the SCSI Sampler section, choose the Peavey SP connected to your computer's SCSI bus.
 5. Save your configuration if you like and/or hit OK.
 6. Check to see that your Configuration settings are confirmed without error and hit Get Sample. Sound Forge will receive the sample very quickly.
 7. When the transfer is complete, the sample will appear in the active sample window.
-

See also

[Using the Sampler Tool](#)

File Menu

New

Open

Close

Open Workspace

Save Workspace

Save

Save As

Save All

Delete

Properties

Send

Exit

New

The *New* command creates a new sound data window.

Sample Rate

Select a standard sample rate from the drop-down list or type in a custom rate.

Sample Size

Check the button to select whether the new data will be 8-bit or 16-bit.

Channels

Check the button to select whether the new data will be mono or stereo.

Maximum Editing Time

Shows the maximum time currently available on the hard drive for editing a file of the specified type.

Note that files open in Sound Forge are edited in 16-bit format and are only converted to 8-bit or compressed audio when closing the file.

Using the *New* toolbar button (or holding the *Shift* key down while selecting *New* from the menu) will skip the *New* dialog and use the previously selected *Mono/Stereo*, *8-bit/16-bit*, and *Sample Rate* settings.

Shortcut

Control+N

Drag and drop a selection to the Sound Forge desktop to create a new file from the selection.

Selecting a portion of the Sound Forge desktop will create a new file with the most recently used attributes.

See also

[File Menu](#)

[Creating a New Window](#)

[Drag and Drop Operations](#)

Open

Use this command to load a sound file into Sound Forge in Windows 95 or NT 4.0. If you are running Windows 3.x, go [here](#).

Look In

Lists available drives and directories.

File Name

Type the *File name* you want to open or a *. and an extension to see a list of files with the corresponding extension in the current directory.

Files of Type

Shows the type of files displayed. Choose a type from the drop-down list. If you select All types (*.*), Sound Forge will try to auto-detect the file format.

If you know that the file format is unsupported, select *Raw File (*.*)*. Hitting *OK*, will take you to the *Open raw data* file dialog where you'll be able to specify format parameters.

MRU Folders

Lists the most recently used directories. Use to speed the opening of regularly used files in different folders.

Open as Read-only

When checked, the sound file will be opened but you will not be able to alter the sound data in the file. 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.

When you open a sound file, Sound Forge defaults to whichever edit mode you last selected (either *Open as read-only* or *Operate directly on the sound file*). You can change modes in the *Open* dialog.

Operate Directly on the Sound File

When checked, the sound file will be opened for editing but with no temporary file created. This greatly speeds the opening process but doesn't give you the security of a backup file. In most cases this is not a problem since all changes can be undone (if you have enough disk space), but if, for some reason, Sound Forge terminated improperly the file would remain in its edited state.

Whenever you open a file in Sound Forge, a backup copy is created so that the original file is not affected until you elect to save your work. When you open a file as read-only or direct, Sound Forge will not create a backup copy of the file, which makes opening files quicker and uses less disk space.

AutoPlay

When checked, a file selected in the window will automatically begin playback without opening it in Sound Forge.

Play

Auditions the sound file without opening it in Sound Forge. If playing a file from a slow network or CD-ROM, some skipping might occur. This feature is only available for .WAV files.

Shortcut

Control+O

Control+F12

Control+Alt+F2

See also

[File Menu](#)

Opening an Existing File
Open raw data file

Open (Windows 3.x and NT 3.51)

Use this command to load a sound file into Sound Forge in Windows 3.x. If you are running Windows 95 or NT 4.0, go [here](#).

Drives

Lists available drives. Select a new drive from this drop-down list.

Directories

Lists available directories. Select a new directory from this drop-down list.

List Files of Type

Select the type of file you want to open. If you select *All Types*, Sound Forge will automatically identify the format of a sound file if it can.

File Name

Type the file name you want to open or a *. and an extension to see a list of files with the corresponding extension in the current directory.

MRU Folders

Lists recently used folders. Use to speed the opening of regularly used files.

Read-only

When checked, the sound file will be opened but you will not be able to alter the sound data in the file. 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.

When you open a sound file, Sound Forge defaults to whichever edit mode you last selected (either *Open as read-only* or *Operate directly on the sound file*). You can change modes in the *Open* dialog.

Direct

When checked, the sound file will be opened for editing but with no temporary file created. This greatly speeds the opening process but doesn't give you the security of a backup file. In most cases this is not a problem since all changes can be undone (if you have enough disk space), but if, for some reason, Sound Forge terminated improperly the file would be remain in its edited state.

Whenever you open a file in Sound Forge, a backup copy is created so that the original file is not affected until you elect to save your work. When you open a file as read-only or direct, Sound Forge will not create a backup copy of the file, which makes opening files quicker and uses less disk space.

AutoPlay

When checked, a file selected in the window will automatically begin playback without opening it in Sound Forge.

Play

Auditions the sound file without opening it in Sound Forge. If playing a file from a slow network or CD-ROM, some skipping might occur. This feature is only available for .WAV files.

Shortcut

Control+O

Control+F12

Control+Alt+F2

See also

[File Menu](#)

[Opening an Existing File](#)

[Open raw data file](#)
[Installing the ACM](#)

Open raw data file

This dialog box is reached when you select Raw from the List of File Types in the **Open** dialog. Use this to open a file that is not stored in one of the standard sound file formats supported by Sound Forge.

Name

List of preset formats available. To create your own format from the selected parameters, press Save As.

Sample Rate (2,000-60,000 Hz)

Sample playback rate which will be used by Sound Forge when playing the file.

Sample Type

Format used to store each sample.

8 Bit PCM

Uncompressed linear format.

16 Bit PCM

Uncompressed linear format.

G.711 u-Law

Compressed format commonly used for telecommunications in the United States.

G.711 A-Law

Compressed format commonly used for telecommunications in Europe.

Format (Unsigned, signed, sign bit)

Shows the sample format that the data is stored as. Samples are most often saved in the PCM format.

Channels (Mono/Stereo)

Number of channels, or tracks, stored in the file.

Byte Order (Little or Big Endian)

Order in which the high and low bytes of a 16-bit sample are stored. Little Endian is used by Intel microprocessors, while Big Endian is used by Motorola microprocessors.

Header Bytes

Number of bytes stored in the file before the sound data.

Trailer Bytes

Number of bytes stored in the file after the sound data.

See also

[File Menu](#)

Open Video Stream

This dialog is reached when you select an .AVI file that has multiple audio or video streams. It allows you to select a particular video stream and an audio stream to open in Sound Forge. Most .AVI files have just one video stream and one audio stream, in which case this dialog doesn't appear.

Stream

The black diamond on the left side of a stream indicates that it has been selected to be open. If you have more than one audio stream or video stream, click on the left of the stream to select it for opening. Additional information about each stream is available by clicking on the + area to expand the list.

See also

[File Menu](#)

Close

Use this command to close the current sample data window. If the data has not been saved since the last edit you will be asked if you wish to save your changes.

See also

[File Menu](#)

[Saving a File](#)

[Close All](#)

Open Workspace

Use this command to open a Sound Forge workspace. A workspace contains a list of all the files open in a session along with their position and zoom ratios.

File Name

Type the file name of the workspace (.sfw) you wish to open.

See also

[File Menu](#)

Save Workspace

Use this command to save a Sound Forge workspace. The workspace will contain a list of all the files open in the current session along with their position and zoom ratios.

File Name

Type the file name of the workspace (.sfw) you wish to save the current session as.

See also

[File Menu](#)

Save

Use this command to save the current sound data. If the data is new data which was not retrieved from a previous file or was loaded from a format which is not supported for saving, you will be prompted with the Save As dialog. The Macintosh Resource (SND) format is not supported for saving. If you need to export sound files to a Macintosh, use the Macintosh AIFF format.

Shortcut

Control+S

Shift+F12

Alt+Shift+F2

See also

[File Menu](#)

[Saving A File](#)

[Save As](#)

[Sound File Formats](#)

[Embedded Summary Information](#)

Save All

Use this command to save all the currently opened files. You will be prompted for each unsaved file. If you hold the shift-key down when selecting this command, no prompts will be displayed.

See also

[File Menu](#)

Save As

Use this command to save the current sample data in a different file and/or file format. The Macintosh Resource (SND) format is not supported for saving. If you need to export sound files to a Macintosh, use the Macintosh AIFF format.

File Name

Type the name of the file in which you want to save the current sample.

Drives

Lists available drives. Select a new drive from this combo box.

Directories

Lists available directories. Select a new directory from this list box.

File Types

Select the type of file to which you would like to save.

Format

Shows the sample format in which the data will be saved. Samples are most often saved in the PCM format. However, other formats are often used for compression purposes. You can save files in a variety of compressed formats including the Microsoft ADPCM format.

When saving as a Raw file, other options are available after choosing OK in this dialog box.

Attributes

Sample Rate, sample size, and stereo/mono. Here, you can change the sample size (8 or 16 bit) that the file will be saved as. You can also convert a file between mono and stereo formats. When you convert from mono to stereo, the data will be stored in both channels. When converting from stereo to mono, the data will be mixed to a single channel.

Rgns/Playlist Information

Lists options for saving *Regions/Playlist* information. Options include saving in the .WAV file, saving to a Sound Forge external playlist file, saving to a Session 8-compatible playlist file or not saving at all.

Save Summary Information in file

Check if you want Summary information (such as author and copyright) stored in the WAV file.

Save Sampler Information in WAV file

Check if you want Sampler Information (such as note assignment and loops) saved in the WAV file.

Only data formats supported by the selected File Type are listed in the Format drop down list box. Other file types are available when saving a file as Raw data.

Shortcut

Alt+F2

F12

See also

[File Menu](#)

[Saving A File](#)

[Converting Files](#)

[Embedding Text in WAV Files](#)

[Sound File Formats](#)

[Embedded Summary Information](#)

Save As Raw Data File

This dialog box is reached when you select Raw from the Format list in the [Save](#) dialog. Use this to save an opened file into a file format that is not one of the standard sound file formats supported by Sound Forge.

Name

This list of file formats contains preset versions of raw file storage formats commonly used by other applications. You can create a new file format by using the Save As button to save the current storage parameters.

Sample Type

(8 Bit PCM, 16 Bit PCM, G.711 u-Law, G.711 A-Law)

Format used to store each sample.

Format

(Unsigned, Signed, Sign bit)

Binary format of each sample.

Channels

(Mono/Stereo)

Number of channels, or tracks, stored in the file.

Byte Order

(Little or Big Endian)

Order in which the high and low bytes of a 16-bit sample are stored. Little Endian (low, high) is used by Intel microprocessors, while Big Endian (high, low) is used by Motorola microprocessors.

See also

[File Menu](#)

[Saving A File](#)

Video Save Options

This dialog is reached when you select Video for Windows (.avi) from the *Save as type* list in the *Save As* dialog. In this dialog, you can select which video streams you wish to save and edit the title, language, and priority of the saved streams.

Streams

Check the streams that you want to save. The black diamond next to each check box indicates currently opened streams.

Additional information is available by clicking on the + area to expand the list. This information can be edited by right-clicking on a field and selecting *Edit*.

Save Audio to New Stream

If this option is checked, the audio track currently open in Sound Forge will be saved to a newly created stream.

Compression Options

This dialog is reached when you select Microsoft .AVI from the *Format* list in the *Save As* dialog and then press *OK* in the *Video Save Options* dialog.

Choose a Stream

Check the audio or video stream for which you want to set the compression formats.

Interleave Frames

When checked, selects how the audio and video streams will be interleaved. Interleaving the audio and video streams improves playback performance when using storage devices that are more efficient at providing data sequentially (such as CD-ROM disks). The *Frames* value indicates the interval between video and audio segments. The default is 1 for CD-ROMs, but can be increased for faster media.

Options

Press this button to view and/or edit the audio and video compression options for the selected stream.

Video Compression

This dialog is reached when you select Microsoft .AVI from the *Format* list in the *Save As* dialog, press OK in the *Video Save Options* dialog, and after selecting a stream pressing the *Options* button in the *Video Compression* dialog.

Compressor

Select a compression format from any of the installed .AVI compression schemes. Compression formats are available from different vendors (not from Sonic Foundry), and also come included with Windows.

If the file is already compressed, you should select *No Recompression*. This will be faster and will also avoid the loss of quality inherent in most recompression.

Compression Quality (0 to 100)

Determines the final output quality. Usually, higher quality (close to 100) means less compression and fewer visual artifacts.

Key Frames Every

Some compression formats are based on key frames (sometimes called temporal compression). A key frame is a frame which is usually less compressed. Compressed frames that follow a key frame are smaller but slower to draw. This option is only relevant to video streams.

A high number of key frames increases quality, but decreases compression. Also, if the number of key frames is very low, it will take a long time to perform decompression when drawing the video strip or moving randomly through the file.

Data Rate (Kilobytes per second)

Determines the data rate which will be required to play the compressed stream in real time. Lower data rates are less demanding on computer systems than higher rates. If the data rate is too high for a certain computer (usually because of a bottleneck such as a slow CD-ROM, network, or hard drive) the quality begins to suffer and glitching can occur.

Configure

Press this button to set more detailed configuration parameters specific to the compression scheme being used. Not all compressors provide this options.

Preview

Press this button to view a preview of the final output.

Save .RA File

This dialog is reached when you select *Real Audio (.ra)* from the *Save as type* list in the *Save* dialog. This feature is only available for Windows 95 and NT running in x86 processors.

Desired playback baud rate

Determines the data rate required to play back the file.

Stereo file conversion

Since *RealAudio* supports mono files only, when saving a stereo file you need to specify how to convert from two to one channels.

Delete

Use this command to delete a sound file from your hard drive. When running the new Windows 95 shell, this function is not available. Instead, you can delete files from the File Open dialog.

File Name

Type the file name you want to delete or a *. and an extension to see a list of files with the corresponding extension in the current directory.

Files

Lists the files in the current directory of the type specified in the File Name box.

Drives

Lists available drives. Select a new drive from this combo box.

Directories

Lists available directories. Select a new directory from this list box.

File Types

Select the type of file you want to delete. This provides an easy way of setting the file extension.

See also

[File Menu](#)

Properties

The *Properties* folders list attributes and information which is embedded in the currently active sound file. The available Properties folders are:

[General](#)
[Summary Info](#)
[Format](#)
[Video](#)
[Display](#)

See also

[File Menu](#)
[Summary Information](#)

General Properties

This folder contains technical information regarding the current sound file, including size, location, file type and attributes of the file.

Microsoft .WAV files are composed of sections called RIFF chunks. These chunks contain the sound data plus other embedded information. If there are any non-standard chunks found in the current file, the junk, padding, and additional chunks fields will be checked. These chunks might be used by other sound utilities, or might just be garbage which can cause problems like glitching in some less intelligent programs.

See also

[File Menu](#)

[Properties](#)

Summary Information Properties

This folder contains extra information that can be embedded in the sound file, about the sound file. It is only available in Microsoft .WAV files.

Title, Subject, Engineer, Comments

Commonly used fields you can quickly edit. To view and edit all other fields, press the Extended button.

Picture

Allows you to select an icon, bitmap or cursor to be attached to the current file.

Load

Sets all of the fields to the default values. The default values can be set in the **Extended Summary** dialog box.

Extended Summary

Allows you to view and edit all available summary fields.

Fields

To edit a field, you must first select it from the Fields list. The check appearing to the left of the type tells whether this field is currently enabled for saving to the WAV file. If the check is not shown, the field will not be written to saved files. To change the state of the check, click with the left mouse button in the area immediately to the left of the type or double click on the entry.

If the field for a type currently has no text associated with it the word (Empty) will appear to the right of the type in the list box. Empty fields are not saved.

Contents

Shows the text associated with the selected field. The text can be changed by typing in this box.

Default

Sets the default fields to the current file's field information. Use this to save information which will be used in a number of different files, like copyright and engineer information at your recording site.

Load

Sets the current file's summary information fields to the default settings.

Shortcut

Double click on the right-most field of the Status Bar

See also

[File Menu](#)

[Summary Information Fields](#)

Attach Picture

Use this dialog to select a bitmap (.bmp), cursor (.cur), or icon (.ico) to embed in the current sound file.

See also

[Summary Information Fields](#)

Format Properties

Use this folder to change the data format of the current data window. This includes *Sample rate*, *Sample size (bits per sample)*, and *Stereo/Mono*.

Sample Rate

The playback rate can be set from 2,000 Hz to 60,000 Hz by entering the rate you wish to use. You may also select the most common sample rates by using the drop-down list. Note that this will not resample the sound file. If the playback rate is different from the originally recorded rate, the pitch will vary unless resampling is done.

Sample Size

Select 8-or 16-bit for the sample size.

Channels

Select *Mono* or *Stereo* for the number of channels.

Cursor Position

Displays the cursor position in the sound file using the current units.

Sample Value

Displays the sample value at the current cursor position in the sound file. For 16-bit audio, this value ranges from -32768 to 32767. For 8-bit audio, it ranges from -128 to 127.

Peak Data Ratio

Ratio at which the peak data is stored. This means that when zoomed out beyond 1:512, the peak file, instead of the entire file is scanned when drawing the waveform.

Sound Data Size

Amount of storage space being used by the sound data.

Video Source Format

Frame size (height, width, bits per pixel) and compression algorithm of the original .AVI file.

Video Decompression Format

Intermediate frame size (height, width, bits per pixel) and compression algorithm used by Sound Forge before displaying the video frames. The video decompression mode can be changed from the *Performance* page in the *Preferences* folder.

Undo buffers

Number of *Undo buffers* currently existing for the sound file and the amount of hard-drive space they consume.

Redo buffers

Number of *Redo buffers* currently existing for the sound file and the amount of hard-drive space they consume.

Shortcut

Alt+Enter

See also

[File Menu](#)

[Properties](#)

Video Properties

This folder contains information specific to Microsoft .AVI files. If your sound file does not currently contain any video, you can attach a video from an .AVI file by pressing the *Attach* button.

An .AVI file can contain multiple video and/or audio streams (or tracks). For example, you can include multiple audio tracks in different languages to go along with a video. When playing back, Microsoft's Media player detects what language the operating system is set to and plays the corresponding video stream.

Stream

Displays information about the video and/or audio streams present in the .AVI file, such as name, compression format, frame rate, and language.

The black diamond to the left of the streams indicate the active streams. You can view and edit additional information (stream name, language, and priority of the active audio stream) by clicking on the + sign to expand the list. Right-clicking on a field allows you to edit it.

Streams that are checked will be preserved when the file is next saved. Streams that are not checked will be deleted when the file is next saved.

See also

[File Menu](#)

[Properties](#)

Display Properties

This folder allows you to change how the data window of the sound file appears on the Sound Forge workspace. Checking a box will display that item in the current sound file's data window. An unchecked item will not be displayed.

To set the current settings as the default display for all sound files, simply check the *Save as the default for all new windows* check box.

If *Data Window only* is enabled, only the Waveform Display will show in each Data Window, with all other controls such as rulers and scrollbars disabled.

Shortcut

[Edit Tool Selector Shortcut Menu](#)

See also

[File Menu](#)

[Properties](#)

Send

Use this command to send the current sound file via electronic mail.

This option is only available in the 32-bit version of Sound Forge.

Sends a mail message with an attached sound file. Note that you must have electronic mail (email) capabilities to use this feature. When you select the *Send* command, your email program opens with a new message and embeds the active sound file from Sound Forge in the message. You then have the option of sending the new mail message to the desired recipient.

When a sound file is embedded into a mail message, the entire contents of the sound file are inserted into the message. To do this requires that any modifications to a sound file be saved before the message can be sent. Sound Forge will prompt you to save the file if this is necessary.

Sound files are generally very large when compared to text files. An email message containing an embedded sound file can take a considerable amount of time to send and receive, especially if you are using a modem and not a high-speed network.

See also

[File Menu](#)

Exit

Use this command to exit Sound Forge. You will be asked whether you would like to save any sound files you have modified during the editing session.

See also

[File Menu](#)

Preset Save As

To save a preset, just enter the name of the preset and press OK. If you select an already existing preset, you will be asked if you want to replace the existing preset with the current preset.

The presets that come installed with Sound Forge cannot be replaced or deleted.

Temporary File Cleanup

If for some reason Sound Forge is terminated improperly when Sound Forge or, most likely, another program crashes all of the opened and unsaved sound files can be recovered excluding those opened in direct and read-only modes. Unless a file is opened in direct or read-only modes, a temporary file is created and any edits made are stored in this file. When an improper termination of the program occurs, these temporary files remain on your hard drive and can be reopened to recover any work done to the sound files before crashing. Also, the original sound files will remain unchanged until you save your work.

Crash Recovery

When starting the program, any temporary files detected in your temporary file directory will indicate that something went wrong. You have the option to delete these files if no important work has been done, or to just ignore these files. Ignoring the files leaves them intact on your hard drive.

Recover

Use this to change the ending of all file names from FORGExxx.TMP to FORGExxx.WAV. You can then open these .WAV files using the *File Open* dialog. The files will be placed in the directory specified on the *Storage* page of the *Preferences* folder.

Delete

This command deletes all the temporary files regardless of selection in the drop-down list. Use this only when you are sure that no important work is in the file.

Ignore

If you don't want to deal with the temporary files left over from an improper termination at this moment, use this command. However, it is recommended that you either rename the files and check their contents or delete them, since they are taking up valuable disk space.

Mono to Stereo

When converting a mono (single channel) file to a stereo (two channel) file, you must specify what channel or channels of the stereo file the mono data should go to. This operation is not only used in file conversions, but also when mixing, pasting, or crossfading mono material into a stereo file.

Left Channel

The mono sound data is placed only on the left channel of the stereo file.

Right Channel

The mono sound data is placed only on the right channel of the stereo file.

Both Channels

The mono sound data is placed on both the right and left channels of the stereo file.

Stereo to Mono

When converting a stereo (two channel) file to a mono (one channel) file, you must specify what channel or channels of the stereo data should go to the mono file. This operation is not only used in file conversions, but also when mixing, pasting, or crossfading stereo material into a mono file.

Left Channel

Only the left channel of the stereo file is used in the operation.

Right Channel

Only the right channel of the stereo file is used in the operation.

Mix Channels

The left and right channels are mixed together to perform the operation.

Preview Configuration

This dialog box allows you to set preview preferences for sound processing operations.

Limit Previews to (1 - 30 seconds)

Determines the length of the wave data that will be processed for previewing.

Fade out last 10 milliseconds (Of/Off)

Use this to fade the ends of the preview data to prevent clicks and pops.

Loop preview continuously (On/Off)

When on, the preview data will loop infinitely when pressing the preview button in the processing function dialog box.

Default all previews to the current setting (On/Off)

Use this to store the current preview settings to be used as defaults for all preview operations. When Off, only the current preview will be affected by any changes made in this dialog box.

See also

[Previews](#)

Attach video

This dialog box allows you to select an .AVI file from which a video stream will be attached to the current sound file.

See also

[Video Properties](#)

View Menu

[Maximize Width](#)

[Full Screen](#)

[Toolbars](#)

[Time Display](#)

[Clipboard Contents](#)

[Play Clipboard](#)

[Zoom Level|Out Full](#)

[Zoom Level|Selection](#)

[Zoom Level|Window](#)

[Zoom Time|In Full](#)

[Zoom Time|Normal](#)

[Zoom Time|Selection](#)

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[Focus to Data Window](#)

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[Playlist](#)

[Keyboard](#)

[Mixer](#)

[Time Display](#)

[Video Preview](#)

[Play Meters](#)

[Loop Tuner](#)

[Undo/Redo History](#)

[Show Loop Tuner](#)

Maximize Width

This command stretches the active data window to fit between the extents of the Sound Forge window.

Shortcut

Control+Enter

Maximize Width button on lower-right hand side of the data window.

See also

[View Menu](#)

Full Screen

This command super maximizes the Sound Forge workspace in your display. To return to the regular window display choose the *Full Screen* command again.

Toolbars

This command brings up the *Toolbars* tab in the *Preferences* folder and allows you to quickly bring up any toolbars you would like on the Sound Forge workspace.

Clipboard Contents

Use this command to display information on the current contents of the [clipboard](#). Information includes, size, sample rate, number of channels, and bits per sample. If no data is in the clipboard you will be notified with a message box.

See also

[View Menu](#)

Play Clipboard

Use this command to hear the current contents of the [clipboard](#). This command has no effect if there is no data in the clipboard.

See also

[View Menu](#)

Zoom Level|Out Full

Use this command to zoom out vertically to allow viewing of the entire amplitude range. This command also centers the centerline.

Zoom Level|Window

Use this command to zoom in or out vertically to the maximum zoom ratio that allows you to view the entire waveform on the waveform display. This command also moves the centerline if the maximum positive and negative peaks are different.

Zoom Level|Selection

Use this command to maximize a selected area both vertically and horizontally.

To zoom in and out vertically by small increments, you can use *Shift+Up* or *Shift+Down* arrow keys or the *Zoom In/Out* buttons on the lower left-hand side of the data window.

Shortcut

Control+Up arrow (when a selection is active)

Level Ruler shortcut menu

Double-click on the *Level Ruler*

Zoom Time|In Full

Use this command to maximize the horizontal magnification to one sample per pixel on the screen. When you zoom in full, the entire waveform display length will span a very small amount of time, usually around 20 milliseconds (depending on the sample rate and window size).

To zoom in and out by small increments, you can use the up and down arrow keys.

Shortcut

Control+Up arrow (when no selection is active)

Waveform Display Shortcut menu

See also

[View Menu](#)

Zoom Time|Normal

Use this command to zoom out to the default zoom ratio preference setting. If the entire sound file can fit in the window at a closer magnification, zoom normal will only zoom out to that ratio.

To change the default for the *Normal zoom ratio*, go to the *Display* page in the *Preferences* folder.

Shortcut

Control+Down arrow

Waveform Display Shortcut menu

See also

[View Menu](#)

Zoom Time|Out Full

Use this command to minimize the data magnification and if possible, fit the whole sample in the data window. To zoom in and out by small increments, you can use the up and down arrow keys.

Shortcut

Waveform Display Shortcut menu

See also

[View Menu](#)

[Magnification and Zooming](#)

Zoom Time|Selection

Use this command to maximize a selected area in the data window.

The *Zoom Selection* command will calculate the maximum zoom factor for the size of a selection area, draw the data at this zoom factor, and center the selection in the data window.

To zoom in and out in time by small increments, you can use the up and down arrow keys or the *Zoom In/Out* buttons on the lower right-hand side of the data window.

Shortcut

Control+Up arrow (when a selection is active)

Waveform Display shortcut menu

See also

[View Menu](#)

[Magnification and Zooming](#)

Focus to Data Window

Use this function to return the screen focus to the sample data window. This is does the same operation as left-clicking anywhere on the data window.

Shortcut

Alt+0

See also

[Window Menu](#)

Regions List

Use this to open or close the *Regions List* window. The *Regions List* contains all of the regions and markers that exist in the currently selected data window. To play a regions, press on the *Play* button to the left of each region. Selecting a marker or region moves the cursor or selection in the data window.

Shortcut

Alt+1 to turn on or set focus.

See also

[Window Menu](#)

[The Playlist and Regions List](#)

Playlist

Use this to open or close the *Playlist* window. The *Playlist* is used to arrange regions for playback. To re-arrange regions, drag and drop them within the *Playlist*. Pressing the *Play* button to the left of each region plays from that region onwards down the list.

Shortcut

Alt+2 to turn on or set focus.

See also

[Window Menu](#)

[The Playlist and Regions List](#)

Keyboard

Use this to open or close the *Keyboard* window. The *Keyboard* allows you to *send MIDI note on and off* commands to your sound card or external sampler.

Shortcut

Alt+3 to turn on or set focus.

See also

[Window Menu](#)

[The Sonic Foundry MIDI Keyboard](#)

Mixer

Use this to turn on or off the Mixer Window. The Mixer allows you to control the Wave (digital audio) and MIDI (synthesizer) output volume level of sound cards that support this feature. The Auxiliary fader can be used to control a number of different parameters, such as CD-ROM output level or Recording Level. Not all sound cards support mixing control.

To select a device, right-click on Wave, MIDI or Aux. You can then choose from a list of available devices from the corresponding Shortcut menu. Grayed items represent devices that are present but do not support Mixer functionality.

Most sound cards contain custom mixer applications for controlling how a sound card behaves. These custom mixers are usually much better suited to setting levels on your sound card than Sound Forge's Mixer window, but are often not as convenient.

Shortcut

Alt+4 to turn on or set focus.

See also

[Window Menu](#)

Time Display

Use this to open or close the *Time Display* window. The *Time Display* window displays the current cursor or play position.

Shortcut

Alt+5 to turn on or set focus.

See also

[Window Menu](#)

Video Preview

Use this to turn on or off the Video Preview Window.

Shortcut

Alt+6 to turn on or set focus.

See also

[Window Menu](#)

Play Meters

Use this to open or close the *Play Meters* window. The *Play Meters* display peak levels during playback.

Shortcut

Alt+7 to turn on or make active.

Undo/Redo History

Use this to open or close the *Undo/Redo History* window. This window will allow you to see all of your edit operations. The most recently performed operations will be at the top of each list.

When you perform an edit operation on your sound file, such as when doing a *Cut*, the operation name appears at the top of the *Undo* list. When you undo the *Cut*, the operation disappears from the *Undo* list and appears in the *Redo* list.

You can audition various versions of undos and redos by pressing the *Play* button next to each. However, some *Undo* or *Redo* operations cannot be auditioned.

To undo or redo up to a certain operation, double-click on the operation in the *Undo* or *Redo* list. If you *Undo* an operation at the bottom of the list, all operations that followed (which are above in the list) will also be undone (this can take some time when working on large files). *Redo* works the same way.

Shortcut

Alt+8 to turn on or make active.

Loop Tuner

Use this to turn on and off the *Loop Tuner* window. The *Loop Tuner* shows the point where the end and beginning of the loop meet.

See also

[View Menu](#)

[Sample Tools](#)

Special Menu

[Transport](#)

[Regions List](#)

[Playlist](#)

[Edit Sample](#)

[Edit Frame Rate](#)

[Edit Tempo](#)

[Center Cursor](#)

[Drop Marker](#)

[Create Region](#)

[Create Loop](#)

[Mark In](#)

[Mark Out](#)

[Toggle Selection](#)

[Undo/Redo History Clear](#)

[Undo/Redo Clear All](#)

[Rebuild Peak Data](#)

Transport

[Record](#)

[Play All](#)

[Play](#)

[Pause](#)

[Stop](#)

[Start](#)

[Rewind](#)

[Forward](#)

[End](#)

[Play Normal Mode](#)

[Play Looped Mode](#)

[Play as Sample Mode](#)

See also

[Special Menu](#)

Record

[Video tutorial](#)

Use this function to record to an existing or new data window. If you have a file open in direct mode, Sound Forge defaults to recording to a new window. To record to the direct-opened file simply use the *Window* button (explained later).

The Record and Record Remote dialogs are always destructive and contain no Undo capabilities. To prevent accidentally recording over sound data, record into a new or scratch data window and paste the takes you want to keep into the desired sound data window.

The following is a list of all controls available in the record dialog.

New

This button allows you to create a new data window for recording.

Close

Press this button to exit record mode.

Remote

Press this button to enter [Remote Recording](#) mode. This mode will hide the main Sound Forge Window and show only the Record Remote dialog box (a condensed version of the Record dialog). This allows you to easily record while using other components of your system like a CD player, mixer, or sequencer.

Go To

Pressing this button displays the Go To dialog for selecting a Start position. This option is not available in Punch In mode.

Selection

Pressing this button displays the Set Selection dialog for selection a Region. This option is only available in Punch In mode.

Window

Pressing the window button allows you to select the data window into which you will record.

Mode

Selects the current recording mode. Modes are as follows:

Automatic Retake

This mode begins recording at the specified **Start** position and replaces any sound data that already exists after that position. When recording is stopped, the **Start** position is automatically rewound to the original starting position.

Multiple Takes with Regions

This mode begins recording at the specified **Start** position and replaces any sound data that already exists after that position. When recording is stopped, the **Start** position is set to ending position of the take. Additionally, a unique Region is created in the **Regions List** that marks the take. To review a take using the **Play** button, press the **REW** button to rewind the **Start** position to the beginning of the take then press **Play**. Pressing the **FWD** button will set the **Start** position to the end of the take in preparation for the next take.

Multiple Takes (no Regions)

This mode is exactly the same as the Multiple Takes with Regions mode with the exception that no Regions are created for takes.

Punch In

This mode is used to record over the top of an existing region of sound. This allows you to record a portion of a previous take without recording a whole new file.

Start

Displays the position where recording will begin.

End

Displays the position where recording will end (this value is inclusive).

Length

Displays the length to record.

Input format

Use this to change how the Start, End, and Length values are entered and displayed as well as the display of Time Recorded and Time Left on Drive values.

Monitor (On/Off)

Select this checkbox to monitor the level of the input signal. This option can be enabled at any time. The meters light up as a function of the volume of the recording input. For best results the level should be somewhere in the yellow range with an occasional red. This button also activates the Peak and Margin values. The Peak value shows the maximum input level recorded since the last Reset was invoked. The Margin value shows the percentage the input level can be increased prior to clipping. If the Peak value reaches 100% (and the Margin value reaches 0%), clipping has occurred.

Reset

Resets the Peak and Margin values back to their initial states.

Prepare

Pressing the prepare button prepares your system for recording. This means that the record device is opened and all preparation that can be done prior to recording is completed. This allows the system to begin recording as quickly as possible after the Record button is pressed. You do not need to use the prepare button unless you are trying to begin recording immediately after pressing the Record button.

Record

Pressing the record button starts recording. If recording has been Prepared, then recording starts immediately. If recording has not been Prepared, Sound Forge attempts to start recording as quickly as possible; a small delay may occur.

Play

Pressing the play button allows you to hear a section of data over which you are going to record or which you have just recorded. This allows you to keep recording until you get the correct take.

Review pre/post-roll

These two edit boxes specify the amount of time which will be played prior to and after regions when reviewing a take.

The following commands and options affect the *Record Meters* display. The *Record Meters* options can be activated by right-clicking anywhere in this dialog or by Alt+Spacebar (Windows 95/Windows NT 4.0) or Alt+- (Windows 3.x/Windows NT 3.51)

Advanced Options

The following options are available from the Record Shortcut menu, which is reached by right-clicking anywhere on the dialog.

Reset Clip

This function resets the clip indication on the *Record Meters*. After a clip occurs, the clip indication will remain lit until the *Reset Clip* function is invoked. You can also click on the clip light or levels to clear it.

Resolution (-24, -42, -60 -78 or -90 to 0 dB)

These settings display the minimum and maximum readings for the *Record Meters*. Changing the resolution of the meters is helpful for metering sound files of varying dynamism.

Show Labels

This option turns on or off the markings on the *Record Meters*.

Hold Peaks

This option turns on or off the markings to indicate the highest peaks in the sound file.

Hold Valleys

This option turns on or off the markings to indicate the lowest peaks in the sound file.

Gap Detection (Ignore, Mark, Stop)

Sound Forge detects most gaps by noticing when it is not able to save wave data as fast as the record device produces it. However, it is still possible for very short gaps to occur because of conflicts between drivers. This sort of gap is usually quite short and either happens frequently or

not at all. If your application has zero tolerance for gaps, we recommend that you monitor your first several recording sessions closely to determine if your system has this problem.

The *Gap Detection* switch can be reached by right-clicking anywhere in this dialog or in the *System* menu. It allows you to specify what action Sound Forge should take if it detects a high probability of a gap having occurred while recording.

Blinking Status (On/Off)

This option enables or disables blinking of the status indicator. It is reached by right-clicking anywhere in the *Record* dialog.

DC Adjust (On/Off)

Checking this before recording to enable automatic adjusting for any *DC offset* produced by your audio hardware during the record process.

Calibrate

When you press this button, Sound Forge will calculate the DC offset while monitoring incoming sound data. The numbers below show the value, in sample amplitude, of the adjustment. If you change sound cards or are recording from different digital sources or at different sample rates, it is suggested that you re-calibrate the *DC Offset* before recording.

Sync

Pressing this button brings up options for syncing Sound Forge and external devices or software.

Shortcut

Control-R

See also

[Special Menu](#)

[Using Recording](#)

[Video tutorial](#)

Record Sync

This dialog allows you to set parameters for input and output synchronization during recording.

Enable MTC/SMPTE Output Synchronization **On/Off**

Turn on to allow Sound Forge to generate MTC/SMPTE synchronization while recording.

Start **On/Off**

When recording begins, Sound Forge will output SMPTE starting at this time.

Pre-roll **On/Off**

Turn on to begin the SMPTE output at a time before recording.

Enable MTC/SMPTE Input Synchronization **On/Off**

Turn on to trigger *Record* of Sound Forge from incoming MTC/SMPTE synchronization.

Start On/Off

Turn on to specify an exact time when Sound Forge will kick into *Record*.

End On/Off

Turn on to specify an exact time when Sound Forge will kick out of *Record*.

Back

Push to switch the focus back to the main *Record* dialog.

See also

[Special Menu](#)

[Using Recording](#)

Record to New

To record to a new window, choose the New button in the Record dialog. The New Window dialog will be presented allowing you to select the attributes for the new window. Holding the Shift key down when selecting the New button will bypass the New Window dialog and create a window using the last used attributes.

When in Direct mode, Sound Forge automatically creates a new file when you hit *Record*.

Shortcut

Control+Shift+R

See also

[Special Menu](#)

[Recording basics](#)

Record Remote

[Video tutorial](#)

The Record Remote dialog is invoked by pressing the Remote button in the Record dialog box. It allows you to record to a data window while using other components of your system to generate audio. Note that some options are not available in the Remote Record dialog that are available in the main Record dialog.

The System Menu (Alt+Spacebar) provides many of the options that are available in the main Record dialog such as New Window, Selection, Go To, etc.

The Record and Record Remote dialogs are always destructive and contain no Undo capabilities. To prevent accidentally recording over sound data, record into a new or scratch data window and paste the takes you want to keep into the desired sound data window.

The following is a list of all controls available in the record dialog.

Close

Press this button to exit record mode.

Un-Remote

Press this button to leave Remote Recording mode and return to Normal Recording mode.

Monitor (On/Off)

Select this checkbox to monitor the level of the input signal. This option can be enabled at any time. The meters light up as a function of the volume of the recording input. For best results the level should be somewhere in the yellow range with an occasional red. This button also activates the Peak and Margin values. The Peak value shows the maximum input level recorded since the last Reset was invoked. The Margin value shows the percentage the input level can be increased prior to clipping. If the Peak value reaches 100% (and the Margin value reaches 0%), clipping has occurred.

Prepare

Pressing the prepare button prepares your system for recording. This means that the record device is opened and all preparation that can be done prior to recording is completed. This allows the system to begin recording as quickly as possible after the Record button is pressed. You do not need to use the prepare button unless you are trying to begin recording immediately after pressing the Record button.

Record

Pressing the record button starts recording. If recording has been Prepared, then recording starts immediately. If recording has not been Prepared, Sound Forge attempts to start recording as quickly as possible; a small delay may occur.

Play

Pressing the play button allows you to hear a section of data over which you are going to record or which you have just recorded. This allows you to keep recording until you get the correct take.

RTZ

Return To Zero (RTZ) sets the Start position to the beginning of the sound data. This option is not available in Punch In mode.

REW

Rewind (REW) sets the Start position to the beginning of the most recent take. This option is not available in Punch In mode.

FWD

Forward (FWD) sets the Start position to the end of the most recent take. This option is not available in Punch In mode.

GTE

Go To End (GTE) sets the Start position to the end of the sound data. This option is not available in Punch In mode.

The following is a list of all extra System Menu (press Alt+Spacebar) commands available in the Record

Remote dialog.

New Window

This option allows you to create a new data window for recording.

Go To

This option displays the Go To dialog for selecting a Start position. This option is not available in Punch In mode.

Selection

This option displays the Set Selection dialog for selecting a Region. This option is only available in Punch In mode.

Window

This option allows you to select the data window into which you will record.

Review Pre/Post-Roll

Enables the Review Pre/Post-Roll values that are entered in the main Record dialog.

Reset Monitor

Resets the Peak and Margin values back to their initial states.

See also

[Remote recording](#)

Record Synchronization

Use this dialog to turn on and off record input and output synchronization.

When **Output Synchronization** is enabled, Sound Forge generates SMPTE/MTC to the output device selected in the MIDI/Sync Preferences. When you start recording, SMPTE output begins at the **Start** value. If **Pre-Roll** is selected, SMPTE output will begin before recording by the specified amount. For example, if the **Start** value is 00:00:30:00 and the **Pre-Roll** is 00:00:10:00, SMPTE will be generated starting at 00:00:20:00 and after 10 seconds (at 00:00:30:00) have elapsed, recording will commence.

When Input Synchronization is enabled, recording begins (is triggered) when the **Start** time is received from the current MIDI/Sync input device (see MIDI/Sync Preferences). Recording will then stop when the **End** time is reached.

Output Synchronization Start

Start time which will be generated when record begins.

Output Synchronization Pre-Roll

Time offset from the Start time at which SMPTE generation will begin.

Input Synchronization Start

Time at which recording will be begin.

Input Synchronization End

Time at which recording will end.

Tips

Make sure that MIDI Input and/or Output is enabled (under the Special menu) to allow MIDI/MTC to be received or generated by Sound Forge. Also, check to make sure that the input and output devices are specified correctly in the MIDI/Sync Preferences (File menu).

Select Record Window

Use this to select an open Data Window which will store the recorded data.

Play All

Use this function to play the entire sample file from beginning to end, regardless of cursor position, selection, or Playlist.

Shortcut

Shift+Spacebar

See also

[Special Menu](#)

Play

Playback file in current playback mode (Normal Mode, Looped Mode, or Sample Mode)

Shortcut

Spacebar

Right-click on Overview

See also

[Special Menu](#)

Pause

Stop playback and keep cursor at the stop point.

Shortcut

Shift+Spacebar

Enter

Right-click on Overview during playback

See also

[Special Menu](#)

Stop

Stop playback and return the cursor to its position prior to playback.

Shortcut

Spacebar (during playback)

See also

[Special Menu](#)

Go to Start

Moves the cursor to the start of the file.

Shortcut

Control+Home

See also

[Special Menu](#)

Rewind

Pressing this button moves the cursor backwards, take by take, in the current sound file.

Forward

Pressing this button moves the cursor forward, take by take, in the current sound file.

Go to End

Moves the cursor to the end of the file.

Shortcut

Control+End

See also

[Special Menu](#)

Play Normal Mode

Sets Playback mode to Normal Mode. When **Play** is selected while in this 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.

Shortcut

To switch between playback modes, use Control+Spacebar

See also

[Special Menu](#)

Play Looped Mode

Sets Playback mode to Looped Mode. When **Play** is selected while in this mode:

- If there is no selection, the entire sample is played in an endless loop.
- If there is a selection, the selection is played in an endless loop.

Shortcut

To switch between playback modes, use Control+Spacebar

See also

[Special Menu](#)

Play as Sample Mode

Sets Playback mode to Sample Mode. When **Play** is selected while in this mode, the file is played with the sustain and release loops repeating a set number of times. Use this to listen to a sample as it would sound when played by a sampler.

Shortcut

To switch between playback modes, use Control+Spacebar

See also

[Special Menu](#)

Regions List

[Add](#)

[Delete](#)

[Modify](#)

[Replicate](#)

[Split](#)

[Update](#)

[Clear](#)

[Markers to Regions](#)

[Open](#)

[Save As](#)

[Copy to Clipboard](#)

See also

[Special Menu](#)

[The Playlist and Regions List](#)

Add to Regions List

Use this function to add a selection to the Regions List.

Shortcut

Drag and drop selection to Regions List

Ruler Shortcut menu

See also

[Special Menu](#)

[Shortcut Menus](#)

Delete Region from Regions List

Use this function to delete a region from the Regions List.

Shortcut

In the Regions List, select a region and press Delete
Region Tag Shortcut menu

See also

[Special Menu](#)

[Shortcut Menus](#)

Edit Regions List

Use this function to modify a region in the Regions List.

Name

Name of the region, which can be up to 255 characters long.

Type **(Marker/Region)**

A Marker is a single point, a region is a start and end point, and has an associated length.

Start, End, Length, Input Format

Use to change the selection. Sound Forge automatically fills in the values depending on your selection or cursor position.

Trigger

Use if you want to trigger a region or marker using MIDI commands.

MIDI 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 region.

MIDI Note On - Play / Note Off - Stop

The region will be played when the specified note on message is received and will stop when either the full region is played or the specified note off message is received.

MIDI Note On - Queue / Note 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.

Channel

Determines the MIDI input channel for triggering.

Note

Determines the MIDI note that will trigger region playback. This value can be entered as either a MIDI note value such as C5 or as a MIDI note number such as 60.

If you have the **MIDI Input Sync/Trigger** mode enabled while using this dialog box, you can auto fill the Channel and Note values by pressing a key on your MIDI keyboard.

Shortcut

Region Tag Shortcut menu

Region List Shortcut menu

Press Enter when a region in the Regions List has focus

Double click on a region in the Regions List

See also

[Special Menu](#)

[Shortcut Menus](#)

[The Playlist and Regions List](#)

Replicate Region in Regions List

Use this function to create a new copy of the marker/region.

Shortcut

Region List Shortcut menu

See also

[Special Menu](#)

[Shortcut Menus](#)

Split Region in Regions List

Use this function to split a region into two at the current cursor position.

Shortcut

Region List Shortcut menu

See also

[Special Menu](#)

[Shortcut Menus](#)

Update Region

Use this function to change the current marker/region location to match the current cursor/selection.

Shortcut

Region Tag Shortcut menu

Region List Shortcut menu

See also

[Special Menu](#)

[Shortcut Menus](#)

Clear Regions List

Use this function to remove all markers/regions from the Regions List.

See also

[Special Menu](#)

Markers to Regions

This function will take all of the markers in the *Region List* and convert them into regions using the data between each consecutive marker as the region boundary.

See also

[Special Menu](#)

Regions List|Open

This function allows the importing of another region/playlist file into the current sound file. This handy option offers the flexibility of having multiple playlists for the same sound file.

Note: Opening a new regions/playlist will clear the present one. Make sure you have saved the current regions/playlist before continuing.

See also

[Special Menu](#)

Open Regions List and Playlist

This dialog allows you to select a Playlist file (.sfp), Session 8 file (.prm), or another Wave file (.wav) to import a Regions list and Playlist from.

See also

[Special Menu](#)

Regions List|Save As

This function saves the current regions/playlist to an external regions/playlist file. This allows the ability to create multiple regions/playlists for a single sound file.

See also

[Special Menu](#)

Save Regions List and Playlist

Use this dialog to save the Regions List and Playlist of the current sound file into a separate Playlist file (.sfp) or Session 8 file (.prm).

See also

[Special Menu](#)

Copy Regions List to Clipboard

Use this function to copy the text of the Regions List onto the clipboard for use with a text editor. Excellent for making a hard copy of the *Regions List*.

Shortcut

Region List Shortcut menu

See also

[Special Menu](#)

Playlist

[Add](#)

[Delete](#)

[Replicate](#)

[Edit](#)

[Clear](#)

[Stop Point](#)

[Pre-Roll on Playback](#)

[Copy to Clipboard](#)

[Open](#)

[Save As](#)

[Convert to File](#)

See also

[Special Menu](#)

[The Playlist and Regions List](#)

Add Section to Playlist

Use this function to add a region to the Playlist.

Shortcut

Drag and drop a region from the Regions List to the Playlist.

Playlist Shortcut menu

See also

[Special Menu](#)

Delete Section from Playlist

Use this function to delete a region from the Playlist.

Shortcut

Press the Delete key when a region has focus in the Playlist.

Playlist Shortcut menu

See also

[Special Menu](#)

Replicate Section in Playlist

Use this command to create a new copy of the region.

Shortcut

Playlist Shortcut menu

See also

[Special Menu](#)

Clear Playlist

Use this function to remove all markers/regions from the Regions List.

See also

[Special Menu](#)

Stop Point

Use this command to place a stop point on the region. Sound Forge stops playback within the Playlist whenever a stop point is reached.

Shortcut

Playlist Shortcut menu

* (asterisk) when a Playlist region has focus

See also

[Special Menu](#)

Pre-Roll on Playback

When on, playback from the Playlist begins at the specified time prior to the selected start point in the Playlist. The amount of Pre-Roll time can also be set in the [Playlist/Region List Preferences](#) folder.

For example, say you have two regions, Region 1 and Region 2 in the Playlist. If you have the Pre-Roll time set to 0.5 seconds, when you start playback of the Playlist from Region 2, the last 0.5 seconds of Region 1 will be played back before starting with Region 2. This allows you to hear the transition between the two regions.

Shortcut

Playlist Shortcut menu

/ (backwards slash) when a Playlist region has focus

See also

[Special Menu](#)

Open

This function allows the importing of another region/playlist file into the current sound file. This handy option offers the flexibility of having multiple playlists for the same sound file.

Opening a new regions/playlist will clear the present one. Make sure you have saved the current regions/playlist before continuing.

See also

[Special Menu](#)

Save As

This function saves the current regions/playlist to an external regions/playlist file. This allows the ability to create multiple regions/playlists for a single sound file.

See also

[Special Menu](#)

Copy Playlist Text to Clipboard

Use this command to copy the text of the Playlist onto the clipboard for use with a text editor.

Shortcut

Playlist Shortcut menu

See also

[Special Menu](#)

Convert to New

Use this command to create a new sound file which contains the regions arranged sequentially as specified in the *Playlist*. The new file will contain the exact sequence of regions you hear when playing from the *Playlist*.

Shortcut

Playlist Shortcut menu

See also

[Special Menu](#)

Edit Playlist Section

Use this function to add or modify a region in the Playlist.

Name

The name of the region in the Playlist that is being modified

Play Count

The number of times the Playlist region will repeat before continuing on the next region in the play list. Setting this value to 0 will cause the region to be skipped during playback.

Trigger

The method with which the Playlist region will be invoked (triggered) when you have the [MIDI Input Sync/Trigger](#) mode enabled.

Channel

If the Playlist region is being triggered by a MIDI note, this will be the receive channel for that region.

Note

If the Playlist region is being triggered by a MIDI note, this is the MIDI note number that will trigger this Playlist region. This value can be entered as either a MIDI note value such as C5 or as a MIDI note number such as 60.

If you have the **MIDI Input Sync/Trigger** mode enabled while using this dialog box, you can auto fill the Channel and Note values by pressing a key on your MIDI keyboard.

SMPTE

If the Playlist region is being triggered from a SMPTE time cue, this is the point in SMPTE time that will trigger the Playlist region.

Pre-roll playback

When checked sets the number of seconds of pre-roll from the end of the prior Playlist region that will be heard when starting the Playlist sequence from this Playlist region.

Shortcut

Playlist Shortcut menu

Double click on a region in the Playlist

Press Enter when a region in the Playlist has focus

See also

[Special Menu](#)

[The Playlist and Regions List](#)

AVI Playlist Save As

Use this dialog to specify a name and directory for the new AVI file which will be created from the current Playlist arrangement.

See also

[Special Menu](#)

[Playlist|Convert to New](#)

Edit Sample

Use this command to change sample parameters.

Sample Type (One Shot, Sustaining, Sustaining with Release)

Selecting one of these choices will determine how the sample is played back.

One Shot The sample will play normally (as it appears) with no loops.

Sustaining The sample will repeat the sustain-loop region the specified number of times.

Sustaining with Release The sample will repeat the sustain-loop region, play the region between the sustain and release loop and then repeat the release-loop region.

Loop (Sustain, Release)

Use to select the loop currently being edited.

Loop Count (1-999, infinite)

Number of times that the sustain or release loop will be repeated before continuing playback. This information may or may not affect how your sampler plays the data. By specifying a finite loop count it is easier to hear how transitions between loops and normal data will sound.

Start, End, Length, Input Format

Use to change the position and length of the loop.

MIDI Unity Note

This is the MIDI note that will cause a sampler to play the sample at the pitch (sample rate) it was originally recorded.

Fine Tune

Some samplers will utilize this information to adjust the pitch (sample rate) of the sample. Sound Forge does not fine-tune the sample itself when this option is utilized. This option is an information type setting that will be transmitted to a sampler via a sample transfer procedure. A sampler such as the K2000 can use this information for its playback of the sample. The K2000 should accurately display this information on Master/Sample/Misc. page, it appears as *Pitch Adjust*, just below *Root Key Number*.

SMPTE Offset/Format

Some sampler editors store a SMPTE Offset value in the sample. Sound Forge ignores this offset value. If you want to add SMPTE Offset information to your sample, choose a SMPTE Format from the Format list and type in an offset value. Otherwise leave the Format option to *No Offset*.

Shortcut

Ruler Shortcut menu

Loop Tag Shortcut menu

Drag and drop a selection to the ruler to create a new loop

See also

[Special Menu](#)

[Sampler Tools](#)

Edit Frame Rate

Use this command to change the *Frames per second* value used by Sound Forge for status information when using *Absolute Frames* or *Time & Frames* format. The included presets are there for your convenience. You can manually enter the frame rate below.

Set Frame Rate to:

Half NTSC video (14.985 fps)

This is one half of the standard rate for video broadcasts.

NTSC video (29.970 fps)

This is the standard rate for video broadcasts.

Red Book digital audio (75.000 fps)

When recorded to CD, the audio is divided into frames which occur at about every 8 ms. Thus, when editing a sound file that will end up in an audio CD, this format is useful for guaranteeing that the start and end of selected audio will exactly match with the final product.

SMPTE Non-drop (29.970 fps)

This is the same as NTSC video.

Video for Windows Default (15.000 fps)

Default rate used by Video for Windows .AVI files.

Frames per Second (1 to 1000)

Number of frames per second used. You can select a pre-set from the list above or manually enter a value.

See also

[Special Menu](#)

Edit Tempo

Use this command to automatically calculate the musical tempo (*Beats per minute*) from the current selection.

To use the *Edit Tempo* function, first make a selection in the waveform display. The selection made should be equal in size (or integer multiple) to the length of a beat or measure. The easiest way to tune a selection to be exactly one measure long is to play the selection looped and change the selection points until a constant downbeat is heard.

Sound Forge uses the current tempo to display measures in the *Time Ruler* and in the *Auto Region* tool.

Type (Beat/Measure)

Choose whether the current selection in the waveform display is one beat or one measure long.

Start, End, Length, Input Format

Determines the selection used to calculate the tempo.

Beats Per Measure (1 to 32)

Use to specify the number of beats in a measure when selecting a measure (e.g. 4 for most pop music out there). It can also be used to specify beats per selection length if the selection is larger than one measure.

Beats per Minute (1.0 to 1000.0)

Sound Forge calculates this tempo value depending on your selection length, selection type (*Measures or Beats*), and Beats per Measure.

Note: The default tempo (Default: 120) can be changed on the <i>Status</i> page of the <i>Preferences</i> folder.

See also

[Special Menu](#)

Center Cursor

Selecting this item will center the display so that the cursor appears in the center of the data window. This item does not actually move the cursor to a new position in the data window, it simply redraws the display so that you will now see the areas of the sound file equally on either side of the cursor. If the cursor does not seem to center when selecting this, it is because the cursor is either very close to the beginning or end of the sound file, and therefore cannot be centered.

Tip: If you have a selection, you can switch the cursor between selection points using *Home* and *End*. You can also switch the cursor between selection points by holding the *Control* key while selecting the *Go to Start* or *Go to End* buttons in the *Playbar* or *Transport* toolbar.

Shortcut

. (period)

* (asterisk)

See also

[Special Menu](#)

Drop Marker

The Drop Marker command is a way to mark positions in a sound file. You can drop a marker at the current cursor position while editing or on the fly during file playback. It is much easier to create a marker from the toolbar or by pressing 'M' on the keyboard rather than the menu (as this would be difficult to coordinate on the fly). Markers can be quickly selected from the list in the Go To dialog box located in the same menu/button bar. Also, markers are placed in the Regions List for quick playback.

There is usually a very small delay between the time when you select Drop Marker and when the marker is actually created. **Using the keyboard shortcut 'M' gives the fastest response**, and thus the highest accuracy.

Depending on the sound card you are using, there might be an offset error between what you hear and where the cursor is at. To correct this, go into the [Wave folder](#) in the Preferences Folder and use the *Position Bias* to correct for the sound card driver's offset error.

Some sound cards do not return position information to Sound Forge very efficiently. If the cursor moves in large chunks during playback, check the *Interpolate position between buffers* to allow Sound Forge to estimate where the sound card playback is occurring.

Shortcut

M

See also

[Special Menu](#)

[Using Markers](#)

Create Region

Use this function to create a new region in the Regions List from the current selection.

Shortcut

R

See also

[Special Menu](#)

[The Playlist and Regions List](#)

Create Sample Loop

Use this function to create a loop from the current selection.

Shortcut

L

See also

[Special Menu](#)

Mark In

This item is similar to the *Drop Marker* function. It will allow you to mark the in point of a new selection while the sound file is playing back. You can also use this as a method of marking the in point of a new selection while editing by moving the cursor to where you want the selection to start. This is a good way to accurately set the start of a selection. For greatest accuracy you should use the keyboard shortcut key '[' when *Marking In* during playback.

Note: If there is a discrepancy between what you hear and the cursor position on screen, use the Play position bias or *Interpolate play position for inaccurate devices* between fields options on the *Wave* page in the *Preferences* folder (Options menu).

Shortcut

[(left square bracket)

See also

[Special Menu](#)

[Using Markers](#)

[Mark Out](#)

Mark Out

Use this function to create a mark out point of a new selection. *Mark In* and *Mark Out* are used to determine a selection, and can be used during playback. For greatest accuracy you should use the keyboard shortcut key ']' when marking out during playback.

Note: If there is a discrepancy between what you hear and the cursor position on screen, use the *Play position bias* or *Interpolate play position for inaccuracies* between fields options on the *Wave* in the *Preferences* folder (Options menu).

Shortcut

] (right square bracket)

See also

[Special Menu](#)

[Using Markers](#)

[Mark In](#)

Toggle Selection

By using the S key (or this menu item), you can toggle back and forth between the last selection and last cursor position. For instance, you may be setting the cursor in various places while navigating throughout a sound file. If you then hit S, the last selected region will immediately return to the display.

Shortcut

S

See also

[Special Menu](#)

Undo/Redo History Clear

Use this function to erase the temporary files created in the undo history for the active file. These files do take up disk space, so you might want to periodically clear them when you are sure that you don't want to undo any of your previous changes. This function only affects the current sound file.

See also

[Special Menu](#)

Undo/Redo Clear All

This command will clear the undo histories for all open sound files. *Clear all* affects all of the opened sound files. After doing this you will be unable to undo changes you have made to all modified sound files.

See also

[Special Menu](#)

Rebuild Peak Data

This function recalculates the peak data file for the active data window. A peak data file is used in displaying a file's waveform without scanning the entire length of the file for the sample amplitudes. This makes screen redraws much faster.

If for some reason you believe that the waveform display is not accurately representing the sound file, you can use this function to force Sound Forge to re-scan the entire file. However, under normal circumstances you should never have to do so since Sound Forge will always update the peak file after every edit operation and when loading a file.

See also

[Special Menu](#)

Window Menu

[New Window](#)

[Cascade](#)

[Tile Horizontally](#)

[Tile Vertically](#)

[Arrange Icons](#)

[Resore All](#)

[Minimize All](#)

[Close All](#)

New Window

Use this command to create a new data window. The new window will be given a generic name. This command uses the sound parameters from when the **File New** command was last used.

Shortcut

Control+Shift+N

See also

[Window Menu](#)

Cascade

Use this command to arrange the data windows so they overlap with the title bar of each window remaining visible.

Shortcut

Shift+F5

See also

[Window Menu](#)

Tile Horizontally

Use this command to arrange the data windows side by side without them overlapping.

See also

[Window Menu](#)

Tile Vertically

Use this command to arrange the data windows top to bottom without them overlapping.

Shortcut

Shift+F4

See also

[Window Menu](#)

Arrange Icons

Use this command to arrange the minimized data icons in the Sound Forge workspace.

See also

[Window Menu](#)

Minimize All

Use this command to minimize all open data windows in the Sound Forge workspace.

See also

[Window Menu](#)

Restore All

Use this command to restore all minimized data windows in the Sound Forge workspace.

See also

[Window Menu](#)

Close All

Use this command to close all open data windows in the Sound Forge workspace.

See also

[Window Menu](#)

Go To Marker

Use this to move the cursor to a marker location.

Shortcut

Marker Tag Shortcut menu

Select Region

Use this to change the current selection to be equal to the region endpoints.

Shortcut

Region Tag Shortcut menu

Edit Marker or Region

Use this to edit a marker or region.

Shortcut

Region or Marker Tag Shortcut menu

Update Marker or Region

Use this to update the marker or region to the current cursor or selection point.

Shortcut

Region or Marker Tag Shortcut menu

Delete Marker or Region

Use this to remove a marker or region.

Shortcut

Region or Marker Tag Shortcut menu

Select Sample Loop

Use this to change the selection to be equal to the loop points.

Shortcut

Loop Tag Shortcut menu

Edit Sample Loop

Use this to edit the sample sustain or release loop.

Shortcut

Loop Tag Shortcut menu

Update Sample Loop

Use this to change a loop to the current selection.

Shortcut

Loop Tag Shortcut menu

Delete Sample Loop

Use this to delete a sample loop.

Shortcut

Loop Tag Shortcut menu

Edit Menu

[Undo](#)

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[Copy Object Link](#)

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Undo

Use this command to undo the last edit operation. For instance, if you deleted a section of data by accident, simply choose the *Undo* option from the Edit menu to put the data back.

You can also undo operations from the Undo/Redo History Window.

Tips

When working on large files, it takes Sound Forge a longer period of time to create an undo buffer. To save time and file space, you can disable the undo. This is done by selecting the [Disable Undo](#) item under the **Edit** menu or from within most processing functions.

Shortcut

Control+Z

Alt+Backspace

See also

[Edit Menu](#)

[Copy, Paste, Cut, and Undo](#)

Redo

Use this command to re-perform an undone event. For instance, if you decide that you really did want to delete that selection of data, select *Redo* and the undo will be undone.

You can also redo operations from the Undo/Redo History Window.

Shortcut

Control+Shift+Z

See also

[Edit Menu](#)

[Copy, Paste, Cut, and Undo](#)

Repeat

Use this command to repeat the last performed operation. This can be used with most processing functions. The last dialog settings will be used unless you hold down the shift key, which allows you to change the operation's dialog parameters.

Shortcut

Control+Y to repeat last operation with the last used dialog box options

Control+Shift+Y allows you to change the dialog settings before repeating

See also

[Edit Menu](#)

[Processing Shortcuts](#)

Cut

Use this command to remove selected sample data and put it onto the [clipboard](#). This command has no effect if there is no selected data. Cutting sample data replaces the previous contents of the clipboard. To hear what the cut will sound like before without altering the data, use [Preview Cut](#).

Cutting data from the beginning of a very large file will take a long time to process. This is because large amounts of data need to be moved. You can preview a cut by using the *Preview Cut/Cursor* function (Edit menu). Also, the *Playlist* can be used to make non-destructive cuts which are instantaneous.

Note: You cannot cut from a single channel in a stereo file since the two channels in a stereo file must always be equal in length. To shift a single channel in time, use the *Delay/Echo|Simple* function in the Effects menu.

Shortcut

Control+X

Control+Del

See also

[Edit Menu](#)

[Copy, Paste, Cut, and Undo](#)

Copy

Use this command to copy selected sample data onto the [clipboard](#). This command has no effect if there is no selected data.

Shortcut

Control+C

Control+Insert

See also

[Edit Menu](#)

[Copy, Paste, Cut, and Undo](#)

Copy Object Link

32 Bit Only: This option is only available in the 32 Bit version of Sound Forge.

Use this command to copy an object link for the active sound file onto the clipboard. You can then *Paste* (or *Paste Special*) the Link into another application (such as Microsoft Word or Write) that supports *Object Linking and Embedding (OLE)*. This feature is useful for embedding sound files in documents.

Object links are simply references to existing sound files on your hard disk or network server. If you embed an object link to a sound file in a document, the sound file is not actually inserted into the document, just a reference. Therefore, if you copy the document to another computer, any sound file(s) that are embedded will not be available on the other computer unless the sound file is stored on a common network server.

If you intend to share documents between computers that contain embedded sound files, it is suggested that you store all sound files on a network server and embed them *using Universal Naming Convention (UNC)* path names. For example, if your server and share name for sound files is \\myserver\sounds, then open and embed all sound files using the UNC path (i.e. \\myserver\sounds\<directory and file name >).

If you are running Windows 95, UNC path names are extremely simple to use. Instead of using a drive letter when browsing in the *File Open* dialog, select *Network Neighborhood* in the *Look In* field and browse to the appropriate network server and share containing your sound files. This method of browsing uses UNC names exclusively.

For more information on *Copy Object Link*, and OLE support in general, refer to *the Object Linking and Embedding (OLE)* chapter in the Appendix.

See also

[Edit Menu](#)

Paste

Use this command to insert a copy of the [clipboard](#) contents at the current insertion point. If there is a selection made, the Paste command deletes the selected data before inserting.

Note: This command has no effect if the clipboard is empty. 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. Pasting into a stereo file will insert data to both channels since the channels in a stereo file must always be equal in length.

Shortcut

Control+V

Shift+Insert

Drag and drop a selection to another data window + Alt.

Instead of using the *Alt* key while dropping, you can also hit the right mouse button while dragging. This toggles between *Mix*, *Crossfade*, and *Paste* drag and drop.

See also

[Edit Menu](#)

[Copy, Paste, Cut, and Undo](#)

[Drag and Drop Operations](#)

Crossfade (Paste Special)

Use this command to crossfade a copy of the [clipboard](#) contents with the sample data. The crossfade start point is either the cursor point or the start or end of the selection in the destination file.

The crossfade operation is similar to a mix, except that you can apply a linear fade to the source and destination data. It is commonly used as a transition between one sound segment to another.

When applying a crossfade, you often want to fade out (100 to 0%) the destination data and fade in the source (0 to 100%) data starting near the end of the destination file. Linear fades are applied only to the area where the destination and source files overlap following the crossfade start point. Any extra source data is appended at the end of the destination file.

Source Start/End Level (-Inf. to 0 dB)

Determines the linear fade applied to the source data before mixing with the destination. A gain equal to the Source End Level is applied to the source data when appending to the end the source file.

Destination Start/End Level (-Inf. to 0 dB)

Determines the linear fade applied to the destination data. Any destination data following the crossfade end point is not affected.

Start crossfade at (Start of Selection, End of Selection)

If there is no selection in the destination data window, the linear fade starts at the insertion point. When there is a selection in the destination data window, the linear fade can begin at the start or end of the destination selection.

Note: Crossfading data of different sample rates will cause the data on the clipboard to play at the same rate as that in the current sample with which the data is crossfaded. This command has no effect if the clipboard is empty.

When using drag and drop crossfading, make sure that the Control key is down when releasing the mouse button. When dealing with mono and stereo data, crossfading will occur on the channel which the selection is dropped to (Left, Right, or Both).

Shortcut

Control+F

Drag and drop a selection from the source to the destination Data Window while holding the Control key. Make sure that you don't start the drag with the Control key down, otherwise you will switch to **Magnify Mode**. The Control key must be down when you release (drop) the selection.

You can crossfade regions by dragging any region onto an open data window. The crossfade will occur at the current insertion point within the data window on which the data is dropped. The *Crossfade* operation is used rather than *Mix* when the *Control* key is held down while dragging. When dragging a single channel from a stereo file or a mono file onto a stereo file you can select whether the data is placed in the left or right channel by the vertical position within the data window. A small L or R will appear next to the cursor when crossfading only to a left or right channel respectively.

See also

[Edit Menu](#)

[Drag and Drop Operations](#)

Mix (Paste Special)

Use this command to mix a copy of the [clipboard](#) contents with the sample data. The mix start point is either the cursor point or the start or end of the selection in the destination Data Window.

Source Volume (-Inf. to 15 dB)

Gain applied to the source data before mixing with destination data.

Destination Volume (-Inf. to 15 dB)

Gain applied to the destination data before mixing with source data.

Invert Data (On/Off)

Use to invert the amplitude of the source or destination data.

Start Mix At (Start of Selection, End of Selection)

When there is a selection in the destination data window, you have the option of performing the mix starting at the beginning or end of the selection. When no selection is present, the data is always mixed starting at the cursor position.

Apply destination volume to overlapping area only (On/Off)

When off, the Destination Volume gain is applied to the entire destination sound file. Otherwise, the Destination Volume gain is applied to only where the destination and source data are mixed.

Pre-Post-fade destination edges (0 - 5 seconds/Off)

Use this to apply a fade to the destination data before and after the mixing region. The Pre-fade time determines the fade time before the mix start, while the Post-fade time determines the fade-time after the mix end.

For example, when mixing vocals over a music track, you can have the music track fade before and after the region where vocals are mixed, so that the vocals are more pronounced. This technique is sometimes referred to as ducking.

Note: This command has no effect if the clipboard is empty. Mixing data of different sample rates will cause the data on the clipboard to play at the same rate as that in the current sample with which the data is mixed.

Shortcut

Control+M

Drag and drop a selection from the source to the destination data window.

See also

[Edit Menu](#)

[Drag and Drop Operations](#)

[Simple Mixing](#)

[Advanced Mixing Options](#)

Overwrite (Paste Special)

Use this command to replace an area of sample data with the contents of the [clipboard](#).

The *Overwrite* command replaces data using the following rules:

- If an area is selected which is greater than the length of the clipboard, the data from the beginning of the selection is replaced for the length of the clipboard.
- If an area is selected which is less than or equal to the length of the clipboard, the data within the selection is replaced with the clipboard for the length of the selection.

The *Overwrite* command is useful when you are trying to replace silent sections of audio with background noise.

Shortcut

Control+H

See also

[Edit Menu](#)

Replicate (Paste Special)

Use this command to copy multiple copies of the [clipboard](#) to the current data window.

The *Replicate* command will copy as many copies of the clipboard as will fit to a selected area. If no selection is currently made in the data window, the function will have no effect.

Copy Partials

Selecting the Copy Partials button will fill the selected region completely, using a partial copy of the clipboard if needed.

Whole Copies

Selecting this option will use only whole copies of the clipboard, and will not fill the selected region completely if its length is not an exact multiple of the clipboard length.

Example:

1. Copy a 1 second sample onto the clipboard.
2. Insert a 5.5 second silence sample in the data wave form.
3. Select the 5.5 second area and select *Replicate*.
4. If you choose Whole Copies, 5 copies of the clipboard will be placed in the area the silence previously occupied and 0.5 seconds of silence will remain at the end.
5. If you choose Copy Partials, 5.5 copies of the clipboard will be placed in the data rather than just 5. No silence will remain.

The replicate command is useful when you are trying to replace silent sections of audio with background noise. It can also be used to introduce stuttering effects or echoes.

See also

[Edit Menu](#)

Paste to New (Paste Special)

Use this command to create a new data window which contains the contents of the clipboard.

Shortcut

Control+E

Drag and drop a selection to the Sound Forge workspace to create a new file from the selection.

See also

[Edit Menu](#)

[Drag and Drop Operations](#)

Trim/Crop

Use this command to remove all data from the sample except the selected data. This command has no effect if there is no selected data.

Trimming/Cropping sample data does not copy data onto the clipboard.

See also

[Edit Menu](#)

[Cropping data](#)

Clear/Delete

Use this command to remove selected sample data without copying it onto the [clipboard](#).

Note: You cannot clear data from a single channel in a stereo file since the two channels in a stereo file must always be equal in length. This command has no effect if there is no selected data.

Shortcut

Delete

See also

[Edit Menu](#)

Select All

Use this command to select all data in the current data window.

Shortcut

Control+A

Control+Numpad 5

Double clicking the left mouse button in the Waveform Display will select all data.

See also:

[Edit Menu](#)

Preview Cut/Cursor

Use this command to playback around a selected region. This lets you preview the result of a Cut or Clear operation without altering the file. Go to the *Previews* page in the *Preferences* folder (File menu) to set the amount of pre- and post-roll around the selection.

If there is no selection, the playback will pre- and post-roll around the cursor position.

See also

[Edit Menu](#)

Pre-roll to Cursor

This command will play the data before the cursor position for the specified pre-roll length in the preview *Preferences* folder.

If a selection is made the pre-roll will occur for whichever end of the selection is active. To toggle between ends of the selection use the *Home* and *End* keys. This function is invaluable when tweaking the end of a long selection and not needing to hear the entire thing.

Shortcut

Control+Shift-click on the *Play Normal* button

Control+Shift+K

See also

[Edit Menu](#)

Edit Tool

Use this command to use the *Edit Tool* of Sound Forge. This is the typical editing mode used for selecting and editing data.

Magnify Tool

Use this tool to zoom in to a particular region without losing your selection. While using the *Magnify Tool*, when you make a selection Sound Forge will draw a dotted rectangle around an area and will magnify the area when the mouse button is released.

There are three modes of operation for the *Magnify Tool*, and these can be chosen by toggle-clicking (refer to the Using Sound Forge section explanation of toggle-clicking) while the *Magnify Tool* is selected. These modes allow for *Zoom Time* only, *Zoom Level* only or a *Time/ Zoom Level*.

To return to the normal editing, choose the *Edit Tool* from the Edit menu.

Shortcut

While using any tool, if you hold the *Control* key down before making a selection with the mouse, you will switch to the *Magnify Tool*. This is handy for quickly magnifying a particular section of your sound file.



Data Magnify Mode

When you are in the data window this cursor shows that you can select an area to magnify. Dragging within the data window will draw a dotted rectangle which specifies the region of data to magnify.

Pencil Tool

Use this tool to edit the waveform by drawing on it. For example, if you have a glitch in the sound data, zoom in to the glitch and smoothly re-draw the waveform using the *Pencil Tool*.

While holding down the left mouse button, you can also use the arrow keys to fine tune the pencil position.

The *Pencil Tool* is only available when operating at magnification levels between 1:1 and 1:16. To return to the normal editing mode, choose the *Edit Tool* from the Edit menu.



Data Pencil Mode

When you are in the data window this cursor shows that you can draw new audio data. Dragging within the data window will draw a new line which represents the new audio data. This is an easy way to remove glitches or pops within the audio data.

Go To

Use this command to set the cursor to a particular location in the sound file.

Go To

Provides preset locations to pick from.

Position (0 to Total length)

The position where the cursor will be moved can be specified here.

Input Format

Determines the input format that will be used to enter the position.

If you have a selection, you can switch the cursor between selection points with the ``` key (reverse quote, normally above *Tab* key). You can also switch the cursor between selection points by holding the *Control* key while selecting the *Go to Start* or *Go to End* buttons in the *Playbar* or *Transport* toolbar.

Shortcut

Control+G

F5

Waveform Display shortcut menu

Double-click on the selection start field in the playbar

See also

[Edit Menu](#)

[Shortcut Menus](#)

[Using Go To](#)

Selection|Snap to Time

This command forces both edges of a selection to a whole time division as designated by the marks on the *Time Ruler* above the data window.

Shortcut

T

See also

[Edit Menu](#)

Selection|Snap Edge to Time

This command forces 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.

Shortcut

Shift+T

See also

[Edit Menu](#)

Snap to Zero

This command forces both edges of a selection to the next zero-crossing of the waveform. The *Editing* tab in the *Preferences* folder allows you to choose whether this is a negative, positive or any zero-crossing.

Shortcut

Z

See also

[Edit Menu](#)

Snap Edge to Zero

This command forces 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.

Shortcut

Shift+Z

See also

[Edit Menu](#)

Selection|Set

Use this command to change the start, end, length, or channel of the selection.

Selection

Provides preset regions to pick from.

Start (0 to Total Length)

The start point of the selection is specified here.

End (Start to End)

The end point of the selection is specified here.

Length (0 to Total Length)

The start point of the selection is specified here.

Channel (Left, Right, Both)

If the sound file is stereo, you can specify which channel of the data will be selected.

Input Format

Determines the input format that will be used to enter the *Start*, *End*, and *Length*.

Snap to Zero

Pushing this button snaps the current selection to the nearest zero-crossing.

Snap to Time

Pushing this button snaps the current selection to the nearest whole time, as defined by the major ticks on the *Time Ruler*.

Shortcut

Control+D

Waveform Display shortcut menu

Double-click on the selection length field in the playbar

See also

[Edit Menu](#)

[Shortcut Menus](#)

[Advanced Editing and Navigation](#)

Undo All

Use this command to undo all operations performed on a file. Unless *Undo* was disabled at some point, this will return your file to its original state when opened. However, if you disable undo at any time, you might not be able to return to the files original state.

When not working in direct mode, you can simply close the file without saving changes to revert to the original file. However, when working in direct mode if you decide to not save the current changes, Sound Forge does an *Undo All* operation automatically before saving.

Disable Undo/Redo

Use this command to prevent Sound Forge from creating temporary files used for undoing operations. Since creating undo files takes time and disk space, it is sometimes more convenient to have the *Disable Undo* option on, especially when working with large files.

Remember to turn the *Disable Undo* feature off when you wish to be able to undo operations.

Since Sound Forge doesn't modify the original file unless the file is opened direct or read-only, you can always recover it by re-opening it. However, saved changes are permanent.

Shortcut

Control+U

See also

[Edit Menu](#)

Tips of the Day

[Accurate play cursor position](#)
[Attach a video to a sound file](#)
[Auto-region](#)
[Context-sensitive help](#)
[Create a new file from the Playlist](#)
[Digidesign's Session 8](#)
[Direct file editing](#)
[Docking a Toolbar](#)
[Double-clicking with markers and regions](#)
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[Save multiple regions list for a file](#)
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[Undo/Redo History](#)
[Vertical Zoom](#)
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[Turn off Tip of the Day](#)

Turn off Tip of the Day

[Next](#)

To keep this window from popping up every time you run Sound Forge, go to the [General Preferences](#) page (Options menu) and uncheck *Show Tip of the Day on open*. While doing so, you will realize how many options and features Sound Forge has. You might then decide that if you learn a new trick from this Tip of the Day feature every time you run Sound Forge you'll save yourself valuable editing time.

[List all Tips of the Day](#)

Optimizing your system

[Next](#) [Prev](#)

Some suggestions concerning how to optimize your system to improve Sound Forge's performance are included in the [Optimizing Sound Forge](#) section.

[List all Tips of the Day](#)

Direct file editing[Next](#) [Prev](#)

For very fast file opens and saves, opening files in [Direct Mode](#) allows you to work on the original file instead of a backup.

[List all Tips of the Day](#)

Play meters

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The Play meters window is opened by selecting *Play Meters* from the View menu. A very fast and accurate playback meter is useful for maximizing dynamic range and achieving the hottest signal possible without distortion. The meter will indicate if clipping ever occurs. It even works while previewing effects. Right click on the meter to change its range.

[List all Tips of the Day](#)

Optimizing AVI performance

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For best AVI performance, make sure the AVI has been copied to your hard drive. CD-ROM's and network drives are often too slow for Sound Forge to keep proper audio and video synchronization. For more details, see [Optimizing Sound Forge](#).

[List all Tips of the Day](#)

Level Zoom[Next](#) [Prev](#)

To view low-level sounds, zoom vertically by using the [Level Zoom](#) spinner on the lower left-hand corner of the Data Window or press Shift+Up/Down arrow keys. Once you're zoomed in, you can drag the Level Ruler up and down to change the zero line.

[List all Tips of the Day](#)

Marker Navigation[Next](#) [Prev](#)

To move the cursor to the next marker or region, press Control+Left/Right Arrow keys. If you also hold down the Shift key, a precise selection can be made.

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Double-clicking with markers and regions [Next](#) [Prev](#)

Double-clicking on the Waveform Display between two markers selects the region between them. The same goes for double clicking inside regions and loops. Triple clicking or holding down the Control key while double clicking selects the entire file. Dragging the new selection extends the selection to the next marker, loop, or region.

[List all Tips of the Day](#)

Workspaces

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You can group files that are commonly used together by saving Workspaces (File menu). A workspace file contains all of the files currently open along with their position and size on the screen.

[List all Tips of the Day](#)

Reactive Previews

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While previewing any processing, you can change any dialog parameter during playback and Sound Forge will automatically create a new preview.

[List all Tips of the Day](#)

Toolbars

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Toolbar buttons exist for almost all functions. To view a list of all possible [toolbars](#), select View|Toolbars. If you hold the cursor over a toolbar button, a floating ToolTip displays the button's function.

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Shortcut Menus

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Shortcut menus exist for almost every square inch of space in Sound Forge. To view a [shortcut menu](#), move the cursor to any area and click the right mouse button. Press F1 while selecting any menu item for more information.

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Docking a Toolbar

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You can drag floating toolbars anywhere on the screen or dock them to any side. Double clicking on any toolbars free space switches it from being floating or docked. To keep a toolbar from docking, hold down the Control key while dragging it.

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Spinner Controls

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Instead of clicking on the upper or lower buttons on a [spinner](#), you can also drag the center button to change the control's value. When you move the mouse slowly (or hold the Control key while dragging), the value will change in small increments.

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Edit boxes[Next](#) [Prev](#)

Any control which requires you to type a number can also be changed by using the Up/Down or Page Up/Page Down arrow keys. Holding down the Control key allows for larger increments or decrements.

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Faders and trackbars[Next](#) [Prev](#)

To fine tune a [fader or trackbar](#) value, hold down both the left and right mouse button while dragging the control. You can also use the Left/ Right arrow and Page Up/Down keys. Double clicking on a control changes the value to its default state.

[List all Tips of the Day](#)

Keyboard shortcuts

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Some of the most useful [keyboard shortcuts](#) include:

Spacebar	Play/Stop
Enter	Pause
M	Drop Marker
R	Create a region
C	Center cursor on screen

[List all Tips of the Day](#)

Opening files[Next](#) [Prev](#)

You can quickly open files by dragging them from any folder onto the Sound Forge desktop.

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Drag and Drop Mix, Crossfade, and Paste [Next](#) [Prev](#)

By [dragging](#) a selection from one open file to another, you can mix, crossfade or paste. To crossfade, hold down the Control key before dropping. To paste, hold down the Alt key before dropping. If you have any markers or regions on the destination file, the cursor will snap to them when near the selection point.

[List all Tips of the Day](#)

New file from selection[Next](#) [Prev](#)

To create a new copy of the current selection, simply drag it onto an open area of the Sound Forge desktop.

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Making a selection during playback[Next](#) [Prev](#)

To make a selection during playback, hit the [key to set the start time and then the] key to set the end time and make the selection. Hitting R will then create a region.

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Magnify Mode

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To quickly switch to *Magnify Mode*, hold down the Control key before making a selection. When you let go of the mouse button, the selection will be magnified to fit the screen.

While in *Magnify Mode* and making a selection, toggle-clicking toggles between Horizontal Only, Vertical Only, and Horizontal and Vertical magnify modes.

[List all Tips of the Day](#)

Re-select the last selection[Next](#) [Prev](#)

To re-select the last selection, press the S key. Press again to unselect the selection.

[List all Tips of the Day](#)

Zero crossing

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When making selections, if *Auto Snap to Zero* is turned on (*Options* menu), the end points will snap to the next position where the wave crosses the zero line. This helps prevent glitches when cutting, pasting and performing other audio processing operations.

[List all Tips of the Day](#)

Regions List and Playlist drag and drop [Next](#) [Prev](#)

Adding a selected region to the Regions List can be done by dragging it from the Waveform Display into the Regions List. If you then want to add the region to the Playlist, drag it from the Regions list into the Playlist. (To open the Regions List, press Alt+1. To open the Playlist, press Alt+2.) You can also create a region and add it to the Playlist immediately by dragging a region directly from the Waveform Display into the Playlist.

[List all Tips of the Day](#)

Time and Level Zoom[Next](#) [Prev](#)

To quickly zoom a selection both vertically and horizontally, double click on the Levels Ruler. Double clicking again zooms out to the default zoom ratios.

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Digidesign's Session 8

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You can save files compatible with Digidesigns Session 8 by turning on Sector align data from the [File Preferences](#) (Options menu). Also, you can Import and Export Session 8 regions ([Special|Regions List|Open](#)) to maintain regions edited in both Session 8 and Sound Forge.

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Dropping markers

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You can create [markers](#) by pressing M during playback. It is also possible to drop markers while recording by pressing the drop marker button in the record dialog.

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Silky smooth fades

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The [Graphic Fade](#) dialog allows you to apply Dithering and Noise Shaping to fades to minimize 16-bit quantization noise.

[List all Tips of the Day](#)

Large Time Display

[Next](#) [Prev](#)

If you'd like to see the current cursor or play position in a large display window, open the *Time Display* window from the *View* menu.

[List all Tips of the Day](#)

Previewing a Cut

[Next](#) [Prev](#)

To [preview a cut](#) operation, make a selection that you wish to delete and Control-click on the *Play Normal* button.

[List all Tips of the Day](#)

Maximize the level of a sound file

[Next](#) [Prev](#)

The [Normalize](#) dialog allows you to specify an amount for the peak amplitude or loudness of a sound file. You can use this to make all sound files equal in loudness.

[List all Tips of the Day](#)

Previewing a Loop transition

[Next](#) [Prev](#)

To preview a transition from the end to the beginning of a loop, make a selection that you wish to loop and Control-click on the *Play Looped* button.

[List all Tips of the Day](#)

Saving a file in RealAudio format

[Next](#) [Prev](#)

The RealAudio file format allows you to broadcast speech in real time over the Internet. To save a file in the RealAudio format (.ra), simply select the RealAudio type from the *Save As* dialog.

[List all Tips of the Day](#)

Playing to the cursor[Next](#) [Prev](#)

To play up to a cursor (with no selection), Control-click the *Play Normal* button. The pre-roll time used can be changed in the [Other Preferences](#) folder (Options menu).

[List all Tips of the Day](#)

Stereo expansion[Next](#) [Prev](#)

In the [Pan/Expand](#) dialog, you can widen the image of stereo files.

[List all Tips of the Day](#)

Mid-Side recording mix

[Next](#) [Prev](#)

You can mix the Mid and Side channels of a Mid-Side recording (MS) in the [Pan/Expand](#) dialog. This converts an MS recording into left and right channels.

[List all Tips of the Day](#)

Pencil tool[Next](#) [Prev](#)

The *Pencil Tool* allows you to draw a waveform straight into the Waveform Display. This is commonly used for correcting glitches. To enable the *Pencil Tool*, zoom in greater than 1:16 and select Edit|Tool|Pencil.

[List all Tips of the Day](#)

Attach a video to a sound file[Next](#) [Prev](#)

To add a video to an open sound file, select the *Video* folder in *File|Properties* and press *Attach video*. You will then need to save the file as a Video for Windows AVI file.

[List all Tips of the Day](#)

Repeating an operation

To repeat the last operation, use Control+Y.

[List all Tips of the Day](#)

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Auto-region[Next](#) [Prev](#)

The [Auto Region](#) tool automatically creates regions for you based on sound levels or tempo.

[List all Tips of the Day](#)

Find the maximum level in a file[Next](#) [Prev](#)

To find the maximum level in a file, use the *Find largest peak option* in the [Find](#) dialog, or use the [Statistics](#) function (Tools menu).

[List all Tips of the Day](#)

Filter out inaudible frequencies[Next](#) [Prev](#)

Frequencies below 20 Hz are inaudible yet can exist in large volumes in sound file. These frequencies make your speaker and amplifier do unnecessary work and make sound editing more difficult. To remove very low frequencies, use the [Paragraphic EQ's](#) Low-Shelf filter.

[List all Tips of the Day](#)

Save multiple regions lists for a file[Next](#) [Prev](#)

You can save the Regions List and Playlist as separate files (sfl). They can then be opened at any time and with any other file. Use the *Special|Regions List|Open* and *Save As* commands to import and export lists.

[List all Tips of the Day](#)

Create a new file from the Playlist[Next](#) [Prev](#)

To save the audio (and video) sequence specified by the Playlist to a new file, use Special|Playlist|Convert to New.

[List all Tips of the Day](#)

Undo/Redo History

[Next](#) [Prev](#)

The [Undo/Redo History](#) window (View menu) shows all of the operations that can be undone or redone. Double click on any listed operation to restore the sound file to a previous state.

[List all Tips of the Day](#)

Fine tuning a control[Next](#) [Prev](#)

To fine-tune a slider, fader, or spinner, hold down both the right and left mouse button while dragging. Holding down the Control key while dragging works the same way.

[List all Tips of the Day](#)

Scrolling with the Time Ruler

[Next](#) [Prev](#)

To scroll the Waveform Display backwards or forwards in time, simply drag the Time Ruler above the waveform.

[List all Tips of the Day](#)

Audio Event Locator

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The [Audio Event Locator](#) allows you to find audio events by clicking and dragging the mouse on the Overview (the thin strip above the Time Ruler). Similar to a scrub, playback follows mouse movement across the Overview and loops when the mouse is still. Releasing the mouse stops playback and centers the cursor on the last start position in the Waveform Display.

[List all Tips of the Day](#)

Accurate play cursor position[Next](#) [Prev](#)

If the play cursor in the Waveform Display seems slightly unaligned with the wave graph, use the *Play position bias* option in the [Wave Preferences](#) folder (Options menu). This compensates for sound cards that report position information incorrectly.

[List all Tips of the Day](#)


Context-sensitive help[Next](#) [Prev](#)

To get help on any main or pop-up menu item, select the item from the menu and press F1 while the menu is still up. Pressing the *Help* button or F1 while in any dialog will take you directly to the page in the Help file describing the dialog's function.

[List all Tips of the Day](#)

Play Loop mode

[Prev](#)

To play in Loop Mode, press the *Play Looped* button . During playback, changing the cursor or selection by poking around the Waveform Display or using [Views](#) updates the play position immediately. This is very useful for tuning loops or selections and auditioning different parts of a file.

[List all Tips of the Day](#)

Options Menu

[Status Format](#)

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[Play Meters](#)

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Status Format

Use this command to set the status format. Possible formats are:

[Samples](#)

[Time](#)

[Seconds](#)

[Time & Frames](#)

[Absolute Frames](#)

[Measures & Beats](#)

[SMPTE Drop](#)

[SMPTE Non-Drop](#)

[SMPTE EBU](#)

[SMPTE Film Sync](#)

Shortcut

Playbar Selection Status Shortcut menu

See also

[Options Menu](#)

[Using Status Formats](#)

[Shortcut Menus](#)

[Status Preferences](#)

Samples

Use this command to set the status format to samples. A sample is the smallest division of digitized sound. The number of samples per second is determined by the Sample Rate.

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

Time

Use this command to set the status format to Hour:Minutes:Seconds:Frames, with the frame rate adjustable from the Edit Frame Rate dialog under the Special menu. In this mode, the hour, minutes, and seconds are in real time, independent of the frame rate.

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

Seconds

Use this command to set the status format to show seconds with six decimal place precision.

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

Time & Frames

Use this command to set the status format to Hour:Minutes:Seconds:Frames, with the frame rate adjustable from the Edit Frame Rate dialog under the Special menu. In this mode, the hour, minutes, and seconds are in real time, independent of the frame rate.

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

Absolute Frames

Use this command to set the status format to show frames and fractions of a frame.

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

Measures & Beats

Use this command to set the status format to beats.

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

SMPTE Non-Drop

Use this command to set the status format to *SMPTE Non-Drop* (30 or 29.97 fps). The standard rate for *SMPTE Non-Drop* when synchronizing to video is 29.97. However, in music applications, a rate of 30.00 is often used. You can enable using 30 instead of 29.97 under the *Options* menu on the *MIDI/Sync* page in the *Preferences* folder.

Note: When using a frame rate of 29.97, the time displayed will be different than real world time because thirty frames are still used to measure one second (see the Appendix section on SMPTE).

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

SMPTE Drop

Use this command to set the status format to *SMPTE Drop* (29.97 fps). Frames will be dropped at specific points in time to ensure that the hours and minutes displayed stay synchronized to real world time (see the Appendix section on SMPTE).

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

SMPTE EBU

Use this command to set the status format to *SMPTE EBU* (25 fps). This is the standard rate used by European television.

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

SMPTE Film Sync

Use this command to set the status format to *SMPTE Film Sync* (24 fps), which is a common slower film rate.

See also

[Options Menu](#)

[Using Status Formats](#)

[Status Formats](#)

Annotations

The names of markers, regions, and loops can be shown on the waveform display, along with dotted lines which run the entire height of the window...all with your permission, of course.

[Marker Names](#)

[Marker Lines](#)

[Region Names](#)

[Region Lines](#)

[Loop Names](#)

[Loop Lines](#)

Annotations|Marker Names

Displays marker names on the waveform display.

See also

[Options Menu](#)

Annotations|Marker Lines

Displays dotted lines on the waveform display showing where markers exist.

See also

[Options Menu](#)

Annotations|Region Names

Displays region names on the waveform display.

See also

[Options Menu](#)

Annotations|Region Lines

Displays dotted lines on the waveform display showing where regions exist.

See also

[Options Menu](#)

Annotations|Loop Names

Displays loop names on the waveform display.

See also

[Options Menu](#)

Annotations|Loop Lines

Displays dotted lines on the waveform display showing where loops exist.

See also

[Options Menu](#)

Play Meters

The following commands and options affect the *Play Meter* display. The *Play Meters* can be turned on from the View menu.

[Reset Clip](#)

[Resolution](#)

[Show Labels](#)

[Hold Peaks](#)

[Hold Valleys](#)

[Always on Top](#)

Meters|Reset Clip

This function resets the clip indication on the *Play or Record Meters*. After a clip occurs, the clip indication will remain lit until the *Reset Clip* function is invoked. You can also click on the clip light or levels to clear it.

See also

[Options Menu](#)

Meters|Resolution

These settings display the minimum and maximum readings for the *Play or Record Meters*. Changing the resolution of the meters is helpful for metering sound files of varying dynamism.

[-90 to 0 dB](#)

[-78 to 0 dB](#)

[-60 to 0 dB](#)

[-42 to 0 dB](#)

[-24 to 0 dB](#)

See also

[Options Menu](#)

Meters|-90 to 0 dB

Determines the bottom-most level that will be displayed. A value of -90 dB is the absolute minimum peak value in 16-bit audio, corresponding to a sample value of +/- 1.

Meters|-78 to 0 dB

Determines the bottom-most level that will be displayed. A value of -78 dB (0.012%) will be the smallest value displayed.

Meters|-60 to 0 dB

Determines the bottom-most level that will be displayed. A value of -60 dB (0.1 %) will be the smallest value displayed.

Meters|-42 to 0 dB

Determines the bottom-most level that will be displayed. A value of -42 dB (0.8 %) will be the smallest value displayed.

Meters|-24 to 0 dB

Determines the bottom-most level that will be displayed. A value of -24 dB (0.6 %) will be the smallest value displayed.

Meters|Show Labels

This option turns on or off the markings on the *Play or Record Meters*.

See also

[Options Menu](#)

Meters|Hold Peaks

This option turns on or off the markings to indicate the highest peaks in the sound file.

See also

[Options Menu](#)

Meters|Hold Valleys

This option turns on or off the markings to indicate the lowest peaks in the sound file.

See also

[Options Menu](#)

Meters|Always on Top

Turning this option on forces the *Play Meters* to always remain on top even when another window is dragged over them.

See also

[Options Menu](#)

Time Display

Use this command to set the Time Display window format and update speed.
Possible formats are:

[Position](#)

[Sync/Trigger Status](#)

[Playlist Position](#)

[Passive Update](#)

See also

[Options Menu](#)

Time Display|Position

Checking this option will display the current cursor position in the time display window. The format of the display will depend on the *Status Format* setting above.

See also

[Options Menu](#)

Time Display|Sync/Trigger Status

Checking this option will display the *Sync/Trigger Status* in the *Time Display* window.

See also

[Options Menu](#)

Time Display|Playlist Position

Checking this option will display the playlist position in the *Time Display* window. The format of the display will depend on the *Status Format* setting above.

See also

[Options Menu](#)

Time Display|Passive Update

Checking this option will reduce the overhead (processor time) needed to update the *Time Display* window. The smallest increments may not be exact but the major increments will stay perfectly locked. If you have a slower machine and are experiencing gaps in playback try checking this option.

See also

[Options Menu](#)

Video Options

These options affect the Video Preview window.

[Copy Current Frame](#)

[White, Black or Default Background](#)

[Animate Video Strip](#)

[Number Frames](#)

[Stretch to Window](#)

[Integral Stretch](#)

[Preserve Aspect](#)

[Passive Update](#)

Video|Copy Current Frame

Enabling this option will copy the frame corresponding to the current cursor location to the clipboard.

See also

[Options Menu](#)

Video|Default Background

This option selects the color used surrounding frames in the *Video Preview* and video strips. The default color can be chosen from the in the Options menu on the Preferences page.

See also

[Options Menu](#)

Video|Black Background

This option selects the color used surrounding frames in the *Video Preview* and video strips.

See also

[Options Menu](#)

Video|White Background

This option selects the color used surrounding frames in the *Video Preview* and video strips.

See also

[Options Menu](#)

Video|Animate Video Strip

When this option is turned on, the video strip above the waveform display will come to life at the current cursor or play position. This feature is frame accurate, meaning that when you move the cursor, the correct frame corresponding to the cursor position will be displayed instead of the static overview frame.

When off, the video strip always shows the frame that corresponds to the left edge of each image in the video strip,

See also

[Options Menu](#)

Video|Number Frames

Turn this on to show the frame number at the bottom left-hand corner if the video windows.

See also

[Options Menu](#)

Video|Stretch to Window

Turn this option on to cause each *Video Preview* frame to fit exactly into the *Video Preview* window. This mode is generally the slowest because resizing a frame by non-integral fractions requires many calculations.

See also

[Options Menu](#)

Video|Integral Stretch

When this resize mode is enabled, the *Video Preview* frame will only be stretched by integral amounts. This is usually the fastest mode.

See also

[Options Menu](#)

Video|Preserve Aspect

Check this re-size option when you want the original aspect ratio (horizontal vs. vertical size) of the video to be preserved.

See also

[Options Menu](#)

Video|Passive Update

Checking this option will reduce the overhead needed to update the *Video Display* window. When on, the *Video Display* will only be updated when the processor is idle, which in slower computers is much less frequent. This will help to keep the audio from skipping when a computer can't update video frames fast enough.

See also

[Options Menu](#)

Scroll Playback

Checking this option makes the data window scroll automatically during playback. When the cursor moves off of the current window it will quickly scroll to show another full window of data.

Shortcut

F6

See also

[Options Menu](#)

Scroll Smoothly

Checking this option changes the behavior of the playback auto-scroll. Rather than jumping a window at a time, the cursor will slowly move back to the center and the wave data will scroll past it. This allows you to always see in front of the sound data that is being played.

Shortcut

Shift+F6

See also

[Options Menu](#)

Drag and Drop Snapping

This command toggles on and off the ability to snap to objects when dragging and dropping from window to window. With this on you can snap to major time ticks on the *Time Ruler*, markers and regions, .AVI video frames and cursor positions.

See also

[Options Menu](#)

Lock Loop/Region Length

Use this to have the length of a region remain constant when changing the start or end time of a region or loop.

Shortcut

Holding the Shift key down while dragging Ruler Tabs also locks the length of a region or loop.

See also

[Options Menu](#)

[Using the Loop Tuner](#)

Paste Events

Enabling this operation will transfer any events such as regions and markers from the source to the destination sound files when pasting, mixing, or crossfading. This occurs when using drag and drop or the clipboard.

If you are using another application to paste data into other than Sound Forge, these events may cause problems. Although a well-written application should ignore these events, if you are experiencing clicks at the beginning or end of the data you should disable this option.

Auto Snap to Time

When on, selections and cursor positions are always rounded to the nearest unit of time length.

See also

[Options Menu](#)

Auto Snap to Zero

When on, selections are always moved outward to the nearest zero crossing. This is useful for avoiding clicks and pops which often result when edits are made while the waveform has a high amplitude, causing abrupt discontinuities in the waveform.

Shortcut

Control+B

See also

[Options Menu](#)

MIDI Input Sync/Trigger

When *MIDI Input Sync/Trigger* is turned on, Sound Forge can be triggered from MIDI commands received through the *MIDI Input* port. The *MIDI Input* port is specified on the [MIDI/Sync](#) page in the *Preferences* folder (Options menu).

When this option is off, the MIDI triggers, Regions List triggers, and Playlist triggers specified will not be enabled. When this option is on, dialogs which specify MIDI triggers will also accept input from the *MIDI Input* port, allowing easy entry of *MIDI Note* and *Controller* values.

See also

[Options Menu](#)

[MIDI and SMPTE Synchronization](#)

MIDI Output Sync

Selecting this item will enable Sound Forge to send MIDI time code out the MIDI *Output* port. The MIDI *Output* port is specified on the [MIDI/Sync](#) page of the *Preferences* folder (Options menu). A SMPTE output offset can also be specified in the MIDI/Sync folder which tells Sound Forge what time to start outputting SMPTE from.

See also

[Options Menu](#)

[MIDI and SMPTE Synchronization](#)

Pre-Queue for SMPTE/MTC

Selecting this item will make Sound Forge open the wave device and pre-load data of the next region to be played in the *Playlist*. This is a good measure to take to insure that audio will begin playing the moment the designated SMPTE time is detected by Sound Forge when synchronizing to MTC. This keeps the time between triggering and playback to a minimum.

This option is automatically disabled when any other audio command is used such as <i>Play, Stop, or Record</i> .

See also

[Options Menu](#)

MIDI Triggers

[Video tutorial](#)

MIDI Triggers are used to control several Sound Forge functions using MIDI commands from external devices such as a MIDI keyboard or sequencer. Triggers can be either MIDI Note-On/Off or Controller commands. For Sound Forge to trigger, the [MIDI Input Sync/Trigger](#) option (Options menu) must be enabled.

Name

The name of the user defined *MIDI Trigger* setup. You can save the current configuration using the Save as button.

Event window

The Event window displays the available trigger operations in Sound Forge and, if enabled, triggering information. Selecting an event allows you to edit its trigger information.

Event

The operation which will be performed by Sound Forge when a trigger is received.

Trigger

The specific note or controller that will evoke the event it is assigned to.

Channel

The MIDI channel the trigger is associated that will evoke the event.

Trigger (None, Note, Controller)

This is where you enter the above information upon selecting an Event in the Event window.

None

Used to select no MIDI trigger for the selected event.

Note

Determines the selected event will be triggered by a MIDI note.

Controller

Determines the selected event will be a MIDI controller.

Channel (1-16)

The channel the controller or trigger is assigned to.

Controller/Note (0-127)

The MIDI controller being assigned, if a controller is selected. The MIDI note if note is selected.

Value (0-127)

The value the controller must reach in order to evoke the event.

Enable MIDI Input Sync/Trigger (for training triggers) (On/Off)

If you have the *MIDI Input Sync/Trigger* mode enabled while in the *MIDI Triggers* dialog, you can auto fill the *Channel*, *Note*, and *Controller* values by pressing a key or controller on your *MIDI Keyboard*.

Note: MIDI Triggers are different from triggers in the Playlist and Regions List. When using triggers in the Playlist, Regions List, or MIDI Triggers, be aware that they can interact with each other to create unexpected results. Sound Forge 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 have the *MIDI Input Sync/Trigger* mode enabled while in the *MIDI Triggers* dialog box, you can auto fill the Channel, Note, and Controller values by pressing a key or controller on your MIDI keyboard.

See also

[Options Menu](#)

Preferences

Many user-configurable options are available in the *Preferences* dialog. To select an item, simply click on the folder tab or use the *Tab* key to place the focus on the folder name and use the right or left arrow key to navigate through the folders. Options available in the *Preference* folder include:

[General](#)

[Display](#)

[Editing](#)

[File](#)

[MIDI and Sync](#)

[Performance](#)

[Regions List and Playlist](#)

[Previews](#)

[Status](#)

[Toolbars](#)

[Wave Devices](#)

[Other Preferences](#)

See also

[Options Menu](#)

General Preferences

Use to select general user options.

Show a Tip of the Day on Startup **Default: Checked**

When this option is enabled, a handy *Tip of the Day* will appear every time you run Sound Forge.

Always Ask for Region, Marker, and Loop Names **Default: Checked**

When creating regions, markers, or loop points and this option is turned off, the *Edit Region* or *Loop* dialog will not automatically appear. A default name based on location will be used.

Open Default Workspace on Startup **Default: Checked**

If this option is enabled, Sound Forge will automatically open any files that were open when you last exited the program.

Open Drag and Drop Sound Files in Direct Mode **Default: Unchecked**

Any files opened by dragging them onto the Sound Forge desktop (from outside of the program) will be opened in direct mode when this option is enabled.

Open Command Line Sound Files in Direct Mode **Default: Unchecked**

When launching Sound Forge from the command line (DOS prompt), if this option is turned on any files in the command will be opened in direct mode. For example, if Sound Forge is being launched by another application (such as a multi-track editor), when this option is turned on Sound Forge will edit directly on the original file.

Default to Slow Scroll When Drag Selecting **Default: Unchecked**

In some very fast computers, automatic scrolling while selecting is too fast to use accurately. When this options is turned on, drag-selecting will cause a slow scroll.

Note: If you click the right mouse button while selecting, you can be toggle slow scroll on or off.

Warn When Mix/Crossfade Rates Mismatch **Default: Checked**

Enabling this option tells Sound Forge to warn you with a dialog prior to mixing or crossfading data which has different sample rates. Mixing or Crossfading data of different sample rates may produce unintended sounding results.

Always Open Full-screen **Default: Unchecked**

Enabling this option forces Sound Forge to always open in full-screen mode. If this option is disabled Sound Forge opens in the size and position it was in when it was last closed.

Confirm on Close **Default: Unchecked**

Enabling this option forces Sound Forge to prompt you with a confirmation message box prior to exiting Sound Forge.

If you are a system administrator you may want to enable this option for users who are new to Windows, and who may double-click on the close box by accident.

Compatible Draw Mode (for Broken Video Drivers) **Default: Unchecked**

Because Sound Forge's drawing routines are highly optimized, they increase the chance of causing little-known video card problems to arise. Some video cards have bugs in their drivers which can make your system lock up when Sound Forge tries to draw a waveform. If you experience a system lockup frequently in Sound Forge, it is likely that your video card is causing this problem.

Compatible draw mode uses a different method of drawing the waveform which, although not as smooth, puts less stress on the video card. With some video cards, this mode actually increases the draw speed. However, more flashing can occur.

This problem is known to happen with ATI's Mach64 card when using High Color (16-bit) in Windows 95.

Compatible Scroll Mode **Default: Unchecked**

When this mode is enabled, Sound Forge will use a less optimized method of scrolling the Waveform Display. In some instances, this can reduce interference problems between your audio and video card during Smooth Scroll playback.

Show a Textured Background on the Workspace Default: Checked

If this option is turned on, a stucco texture will be used for Sound Forge's application background.

Show Free Storage Space on Status Bar Default: Checked

When this options is turned on, the total amount of free disk space available on your specified temporary drive is displayed on the status bar.

Show Shuttle Controls on Data Window Transport Default: Unchecked

When enabled, Rewind and Forward buttons will appear on each Data Window's Transport.

Auto-power MIDI Keyboard Window Default: Checked

Enabling this option tells Sound Forge to open up the MIDI device assigned to the *MIDI Keyboard* (if it is not already open) when you press a key on the *MIDI Keyboard*. You may want to disable this option if you are using the same MIDI output device for MIDI synchronization or for your sequencer. If this option is disabled you need to press the *On* button on the Keyboard prior to using it to send notes.

Ignore Device Capabilities in Mixer Window Default: Unchecked

Normally, when Sound Forge loads audio devices found in the system, it checks to see if the drivers support volume control. Those devices which report that they do not support volume control are grayed in the mixer devices list. However, some devices do not report their capabilities correctly. If this option is enabled Sound Forge will ignore the device capabilities and attempt to set the levels for any device you select in the *Mixer* window.

Show OLE Drag Source on Data Window Transport Default: Checked

When checked, the *OLE Drag Source* control is displayed on the right side of the status bar in all data windows. This provides a convenient method of creating Sound Forge audio objects that can be embedded into other documents. If you do not use this feature and want to reclaim the space for the status bar, simply uncheck this option.

Multiple select file open (Windows 3.x and NT) Default: Checked

When this option is checked you can select multiple files in the File Open dialog and Sound Forge will open all of them.

When running Windows NT, if this option is turned on you will not be able to use long file names. This is a bug in the common dialogs in Windows NT. If you want to use long file names, turn this option off.

3D Dialog and display effects (Windows 3.x and NT) Default: Checked

This option is used to enable or disable the 3D look of dialogs.

Recently Used File List Default: Enabled (2 to 9)

This option allows you to set the number of recently used files which will appear in the File menu. If you do not wish to have any files listed in the File menu, uncheck the check box for this option.

See also

[Options Menu](#)

[Preferences](#)

Display Preferences

Use to set your display preferences.

Max zoom ratio on open **Default: 1:1,024**

Specifies the maximum zoom magnification when loading a new sample file. This allows you to control the maximum data that will be shown in a data window when a file is loaded from the hard drive. Higher values show more data on initial file loading, which takes more time.

Wait cursor zoom ratio threshold **Default: 1:32**

Specifies the amount of zoom above which an hourglass cursor will be displayed between data window repaints. If the mouse cursor seems to flicker on and off a lot when redrawing wave data, you may want to increase this value.

Show Axis in Waveform Display **Default: On**

Specifies whether the sample data window displays a set of horizontal axes indicating the volume of the sample. Lines are drawn at volume levels of 0, 50, and 100% of the maximum value.

Color Preference for

The color preferences section allows you set a custom color for a variety of graphics within Sound Forge. Each graphic has a default color or you can set a custom color by checking the **Custom** box and setting the component Red, Green, and Blue of each color.

Play Cursor

Specifies the color of the cursor which is drawn while playing a file.

Regions: Marker

Specifies the color used to display Marker tabs on the Data Window ruler.

Regions: Region

Specifies the color used to display Region tabs on the Data Window ruler.

Sample Loop: Sustaining

Specifies the color used to display Sustain loop tabs on the Data Window ruler.

Sample Loop: Release

Specifies the color used to display Release loop tabs on the Data Window ruler.

Wave Axis: Center

Specifies the colors used to draw the center line of the Wave Axis.

Wave Axis: Half

Specifies the color used to draw the line at half volume for the Wave Axis.

Wave Background

Specifies the colors used to draw the background of each Data Window.

Wave Data

Specifies the color used to draw the wave data in each Data Window.

See also

[Options Menu](#)

[Preferences](#)

Editing Preferences

Use this to set preferences related to editing and undo operations.

Disable Triple-clicking to Select All Data **Default: Off**

Turn this option on if you don't want Sound Forge to select all sound data when you triple-click. You might want to turn this off if triple-clicks are constantly being falsely detected when you make a selection and then try to do a drag operation. Otherwise, lowering Window's double-click threshold time might help.

Snap to Zero-crossing Slope **Default: Negative Slope**

Specifies how zero-crossings are detected.

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.

It is usually best to use either the *Positive* or *Negative Slope* setting for zero-crossings, so that noticeable pops and clicks are not generated by cutting data when using *Snap to Zero*.

Zero Cross Scan Time (0 to 1000 samples) **Default: 1000**

Determines the maximum time that Sound Forge will search for the next zero-crossing.

Zero Cross Level Threshold (0 to 32,767) **Default: 0**

Levels below this value will be considered a zero-crossing. This is useful if the file you are using has a DC offset, although a better idea would be to remove the DC offset.

Disable Auto-snapping Below 1:4 Zoom Ratios **Default: Off**

When this option is turned on, auto-snapping will not occur when you are zoomed in at 1:1 or 1:2. This is useful if you commonly zoom in full to tweak selection points manually yet still want to use auto-snapping when zoomed out.

Limit Number of Undo Buffers To (0 to 999) **Default: 100**

If this is checked, it determines the maximum number of *Undo* operations that will be saved before they start to get deleted. Having many *Undo* operations available will use a lot of hard-disk space.

Limit Number of Redo Buffers To (0 to 999) **Default: 100**

If this is checked, it determines the maximum number of *Redo* operations that will be saved before they start to get deleted. Having many *Redo* operations available will use a lot of hard-disk space.

Default All Windows to Undo Disabled **Default: Unchecked**

Enabling this option will keep Sound Forge from creating undo buffers for any operation which modifies a sound file. You are still able to turn the undo creation on by using the *Disable Undo* command (Edit menu) for any window, but by default no undo buffers will be created.

Drag & Drop Auto Rise Delay (0.1 to 2.0 seconds) **Default: 0.7**

Determines the time necessary before a window underneath the cursor becomes active during drag and drop operations.

See also

[Options Menu](#)

[Preferences](#)

File Preferences

Use this to set preferences related to opening and saving files.

Warn if Save Changes the Sound Data Default: **Checked**

When you save a sound file to a different format, such as doing 16 to 8-bit conversions or compressing audio, you will not hear the conversion's effect on the sound until you re-open the file. If this option is enabled, you will get a message warning of this.

Automatically Re-open if Save Changes the Sound Data Default: **Unchecked**

If this option is on, when you save a file to a different format Sound Forge will save the file and then re-open it. This will happen when you change bit size, channels, or compression format and it will allow you to listen to any changes in sound quality.

Sector Align Data for Digidesign's Session 8 .WAV Files Default: **Unchecked**

This option is only necessary if you are using Sound Forge to modify files for use with Digidesign's Session 8. Enabling this option forces Sound Forge to align the data portion of .WAV files on a sector boundary, which is needed when using .WAV files with the Session 8.

Save .VOC Files Using Newest Format (16-bit capable) Default: **Unchecked**

Prior to Creative Labs' boards supporting the playback of 16-bit data, VOC files had no way of storing 16-bit audio data. With the addition of 16-bit playback sound boards to the Sound Blaster line of products, a newer version of the VOC file was created. With this option disabled Sound Forge will save all 8-bit files in the old format and all 16-bit files in the new format. With this option enabled it will save all files in the new format. Only enable this option if you have software which requires the new format for 8-bit data.

Open and Save with Sound Forge 3.0 Compatible Regions Default: **Off**

Sound Forge 3.0 had a minor bug when opening and saving regions in which the total length was off by one sample. Since we both opened and saved the files inconsistently with the published standard, it didn't affect anything. Sound Forge 4.0 fixes this problem, but in doing so this means that file created with Sound Forge 3.0 will have their region length display 1 sample smaller than before.

If this bothers you, turn this option on when opening a file saved in Sound Forge 3.0, turn it off, and then save the file. From there on after, the region lengths will not be off by one in Sound Forge 4.0 (but will be in 3.0!)

Default SDS Channel When Saving (1 to 16) Default: **1**

Specifies the channel used in SDS packets when saving audio as an SDS file.

Default SDS Patch When Saving (0 to 127) Default: **0**

Specifies the patch used in SDS packets when saving audio as an SDS file.

Default Dialogic .VOX Sample Rate for Open Default: **8,000 Hz**

Specifies the default sample rate used when opening Dialogic .VOX sound files in Sound Forge. This option is necessary because Dialogic .VOX sound files do not contain this information. Typical settings for this option are 6,000 Hz and 8,000 Hz.

Associate Sound File Extensions

Pressing the *Associate Sound File Extensions* button from the *File* tab in the *Preferences* folder opens the *Sound File Associations* dialog. This dialog provides a method of associating audio files with Sound Forge. When a file is associated with Sound Forge, you can simply double-click on a sound file in the *File Manager* or the *Windows 95 Explorer* and it will open for editing.

See also

[Options Menu](#)

[Preferences](#)

MIDI and Sync Preferences

Use to select MIDI and synchronization user options.

Output

Specifies the MIDI output device for synchronization.

This device is used to send SMPTE/MTC when *MIDI Output Sync* (Options menu) option is enabled.

Input

Specifies the MIDI input device for synchronization and triggering when the *MIDI Input Sync/Trigger* (Options menu) is enabled.

This device is the device through which Sound Forge will receive all MIDI triggering and synchronization input. This includes *SMPTE/MTC*, *MIDI triggers*, and *Regions/Playlist triggers*.

Use 30 Frames Per Second for SMPTE Non-Drop **Default: Checked**

Specifies whether all *SMPTE Non-Drop* calculations performed by Sound Forge use a frame rate of 30 frames per second or 29.97 frames per second. You will only need to uncheck this option if you intend to work with video which expects true 29.97 fps *SMPTE Non-Drop*. Refer to the Appendix for a discussion of SMPTE issues.

Bound Record Time on SMPTE Record Sync **Default: Checked**

When this option is enabled, Sound Forge will not allow recording beyond the specified end time. This ensures that your record length is exact regardless of any inaccurate time code.

Use Internal Timer for SMPTE Generation **Default: Checked**

Specifies whether Sound Forge should 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.

Internal Timer Resolution (2 - 32 milliseconds) **Default: 4**

Specifies the timer accuracy used for generating SMPTE when the *Use Internal Timer for SMPTE generation* option is enabled. The lower the resolution value, the more accurate the SMPTE generation will be. However lower values may also decrease system performance.

Use Free-wheel for SMPTE Loss **Default: Off**

Enabling this option makes Sound Forge stop playback if the incoming MIDI time code signal stops being received. Otherwise, when the time code stops, Sound Forge will continue to play until the user stops playback manually.

Free-wheel Time (0.0 to 20.0 seconds) **Default: 1.000**

When the time code stops being received, Sound Forge will continue to play for this amount of time before stopping. During this time, if time code is found again, playback will continue.

Free-wheel Slack (0.05 to 1.0 seconds) **Default: 0.200**

This parameter controls how fast Sound Forge should expect time code updates before going into *Free-wheel* mode. If you have a fast machine, this value can be set to a lower value if you want Sound Forge to stop playback immediately when time code is interrupted.

Enable SMPTE Playback Offset **Default: 00:00:00:00**

When outputting MIDI time code during playback, you can specify a SMPTE offset that will be added to the time displayed in Sound Forge's play counter. For example, if you want Sound Forge to generate MIDI time code starting at 01:00:00:00, instead of inserting 1 hour of silence to your sound file, you can specify that amount in this field.

When using Record Sync, you'll often want to set this value to the Input Synchronization start time. Sound Forge's ruler and play counter will not display this offset.

See also

[Options Menu](#)

[Preferences](#)

Performance Preferences

Use this to set preferences related to opening and saving files.

Temporary Storage Folder **Default: C:\FORGE40**

The specified directory is used to store temporary files used by Sound Forge.

Sound Forge stores its temporary data in a temporary directory on your hard drive. Using temporary file space allows you to edit very large files and keeps Sound Forge from eating up large portions of RAM on your computer. Your temporary directory must have enough space to handle the total size of all files you plan to edit at one time along with space for any clipboard data and undo buffers.

If you change the *Temporary storage* folder, you will have to close Sound Forge for the change to take effect.

Free Storage Space in Selected Folder

Shows the amount of available disk space in the selected folder.

Video Decompression Mode **Default: 8 bits always**

Determines the bits per pixel (color resolution) which will be used by Sound Forge when it decompresses video frames before displaying. Note that when a high number of bits is used, the performance demands on your system are higher. The available modes are:

8 bits always

Sound Forge will always decompress the video to 8 bits per pixel (256 color).

Best for display

Sound Forge will decompress the video to the best format that matches your system's color capabilities.

Full decompress always

Sound Forge will always decompress the video to its original bits per pixel when displaying. This is useful if you want to ensure that the video frames you see contain all the color information stored in the file. Also, if you wanted to paste frames to the clipboard, this ensures maximum resolution.

Peak Ratio Default For New Sound Files

Default: 1:256

When drawing the waveform display, Sound Forge will use a peak file instead of scanning all of the data when the zoom ratio is above this level. Lower values allow for faster scrolling at those ratios, but increases the peak file's size. For example, setting this value to 1:128 instead of 1:256 will make the peak file two times larger but will make scrolling at 1:128 faster and scrolling at 1:65536 or higher ratios slower.

To calculate the size of the resulting peak files, you 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.

Wait Cursor Zoom Ratio Threshold

Default: 1:512

Specifies the amount of zoom above which an hourglass cursor will be displayed between data window repaints. If the mouse cursor seems to flicker on and off a lot when redrawing wave data, you may want to increase this value. This value is ignored during playback.

Show the Position of the Playback Cursor

Default: Checked

When this option is enabled, the *Position* field in the status bar will show the cursor position during playback. Disable this option if you have a very slow computer or video card.

Show the Record Counter While Recording

Default: Checked

When this option is enabled, the record time will show in the *Record* dialog while recording. Disable this option if you have a very slow computer or video card.

See also

[Options Menu](#)

[Preferences](#)

Regions List and Playlist Preferences

Use to select user options dealing with the *Regions List* and *Playlist*.

Regions List Display Format

Default: Start and End

Specifies how the regions are displayed within the Regions List.

Start and End

Regions are displayed with the start and end time of each region.

Start and Length

Regions are displayed with the start time and length of each region.

Playlist Display Format

Default: Start and End

Specifies how the regions are displayed within the *Playlist*.

Start Only

Regions are displayed with only the start time of each region.

Start and End

Regions are displayed with the start and end time of each region.

Start and Length

Regions are displayed with the start time and length of each region.

Sort the Regions List Alphabetically

Default: Checked

Specifies whether regions are displayed in the *Regions List* in the order they were added or are sorted alphabetically.

Playlist Pre-roll (0.0 to 300.0 seconds)

Default: 2.000 seconds

When playing entries in the *Playlist*, this value specifies the amount of pre-roll which will be used. This allows you to easily hear the transition from one region to another without having to play all the way through the first region.

See also

[Options Menu](#)

[Preferences](#)

Preview Preferences

Preview preferences apply to all processing, effects, and tools which have the ability to play a preview. You can also modify preview options within any dialog which has preview by clicking on the system box of the dialog and selecting the *Preview Configuration* option.

Limit Previews To (1 to 30 seconds) Default: 4 Seconds

Specifies the length of audio which will be used when generating a preview. Keeping this value small lowers the amount of time needed to generate a preview when tuning effects or process values.

Pre-roll (1 to 30 seconds) Default: Unchecked

When turned on, this determines 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.

Post-roll (1 to 30 seconds) Default: Unchecked

When turned on, this determines 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.

Fade Out Last 10 Milliseconds (snap/pop eliminator) Default: Checked

Enabling this option tells Sound Forge to fade out the last 10 milliseconds of a preview effect so that large pops are not heard at the end of a preview buffer. With this option disabled, a preview could end with a large data value which can cause an audible and annoying pop.

Loop Preview Continuously Default: Checked

Enabling this option causes preview buffers to loop infinitely rather than playing just a single time. This can be useful when listening to the difference between the original sound and the sound after an effect or process has been applied.

Reactive Previewing Default: Checked

Reactive Previewing is a preview mode in which any control changes in a dialog cause an automatic re-calculation and playback of the preview buffer. This allows for immediate aural feedback on the affects of a change.

This option is very useful when using a fast computer, limiting previews to a short time, and not using very slow processing effects. You can temporarily disable *Reactive Previewing* in a dialog by holding down the *Shift* key while making parameter changes.

See also

[Options Menu](#)

[Preferences](#)

Status Preferences

Use to select display status options.

Default SMPTE Format

Default: Non-Drop 30 FPS

Specifies the SMPTE type used when selecting a SMPTE format option from the *Status Bar* shortcut menu.

Non-Drop 30 FPS

Calculates status values using a 30 frames per second SMPTE code.

Drop 29.97 FPS

Calculates status values using a 29.97 frames per second SMPTE code. This is accomplished by dropping two frames every minute except for minutes 00, 10, 20, 30, 40, and 50.

EBU 25 FPS

Calculates status values using a 25 frames per second SMPTE code. This format is the *European Broadcast Union* time code.

Film Sync 24 FPS

Calculates status values using a 24 frames per second SMPTE code. This format is for standard 24 frames per second film media.

Default Time Format

Default: Time

Specifies the time format (time or seconds) used when selecting the *Time* option from the Status Bar shortcut menu.

Default Frames Per Second (1 to 1,000) Default: 15.000

Specifies 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 shown by Sound Forge 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.

Frame rates can be any value between 1.0 and 1,000.0 frames per second.

Default Beats Per Measure (1 to 32) Default: 4

Specifies the number of beats in each measure for displaying in measures and beats. For example 2/4 time would have two beats per measure.

Default Beats Per Minute (1 to 1,000) Default: 120.000

Specifies the number of beats per minute, i.e. the tempo of a song for displaying lengths. For example, a tempo of 120 is 120 beats per minute.

RMS Level Scan Time (5 to 2,000 ms) Default: 50

Specifies the amount of sound data surrounding the cursor used to calculate the RMS level in the Levels toolbar.

Peak Level Scan Time (5 to 2,000 ms) Default: 50

Specified the amount of sound data surrounding the cursor used when searching for a peak level to display in the *Levels* toolbar.

Display Format for SMPTE Code Default: 00:00:00.00

Specifies how SMPTE values will be formatted for display by Sound Forge. Select one of the available preferences from the drop-down list.

See also

[Options Menu](#)

[Preferences](#)

Toolbars Preferences

Use to select which toolbars are displayed on the main screen. A check before the toolbar means that the toolbar is turned on. The following toolbars are available:

Standard Contains standard file, and edit operations.	Default: Checked
Transport Contains record and play operations.	Default: Checked
Navigation Contains operations for navigating within the data window. Some of the operations included are zooming, marking, moving the cursor, and moving to relevant points.	Default: Unchecked
Views Contains operations for storing and retrieving data window views.	Default: Unchecked
Status/Selection Contains operations to set the display <i>Status Format</i> as well as the Snap To operations.	Default: Unchecked
Regions/Playlist Contains the <i>Regions/Playlist</i> buttons as well as synchronization commands and synchronization status boxes.	Default: Checked
Process Contains all processing operations.	Default: Unchecked
Effects Contains all effect operations.	Default: Unchecked
Tools Contains all tool operations.	Default: Unchecked
Levels Displays the left and right channel levels at the cursor in a variety of different formats.	Default: Unchecked
Show Tooltips The <i>Show Tooltips</i> check box turns on or off the help information box displayed next to the mouse when it is held over certain items.	Default: Checked

See also

[Options Menu](#)

[Preferences](#)

Wave Devices Preferences

Use this folder to specify what wave devices Sound Forge should use for recording and playback.

Note: The capabilities of these devices may limit you from playing some files.

Playback

Specifies the device to be used when playing sound data.

Selecting a mapper device such as the *Wave Mapper* or *Microsoft Sound Mapper* allows Windows to select an appropriate device to use for the current sound data.

Record

Specifies the device to be used when recording sound data.

Selecting a mapper device such as the *Wave Mapper* or *Sound Mapper* allows Windows to select an appropriate device to use for the current sound data.

Interpolate Position for Inaccurate Devices Default: Checked

Specifies whether Sound Forge should interpolate the position of the play position pointer during playback. If this option is not checked, Sound Forge relies on the device driver for the sound card to provide the correct position of the pointer during playback.

Many sound card drivers do not report their position accurately. If the play position pointer does not move smoothly you should enable this option. You may also want to contact your sound card manufacturer for updated drivers.

If your sound card driver does not have accurate position reporting, your *Drop Marker* and *Mark In/Out* positions will also be off when dropping markers while playing.

Position Bias (-64 to 64)

Default: 0

Specifies an offset value which Sound Forge should add to the position values returned by the sound card wave driver during recording or playback.

Many sound card drivers do not report their position accurately. When you play sounds back, the position of the play pointer may seem to lag or precede the actual sound you are hearing. If this is the case, you may want to adjust this value. Most sound card drivers that report incorrect position give a value ahead of the actual sound.

To remove this bias, start with smaller values and work toward larger ones in increments of 4 (a value of 1 is equal to 512 bytes) until the play position pointer is accurate. Recommended settings are 4, 8, 16, and 32. These settings correspond to the typical bias values for these drivers of -2048, -4096, -8192 and -16384 (negative multiples of 2048) bytes.

If your sound card driver does not have accurate position reporting, your *Drop Marker* and *Mark In/Out* positions will also be off when dropping markers while playing.

An inaccurate playback cursor is due to poorly written sound card drivers. If you experience this problem, call your sound card manufacturer and request that they get their act together.

Total Buffer Size (kilobytes)

Default: 512

Select the amount of RAM you would like to use for buffering sounds during play and record. Using a large value can optimize performance at high sample rates but may also cause a slight delay starting playback.

Preload Size (kilobytes)

Default: 0

Determines the total amount of wave data that will be queued to the sound driver before playback starts.

See also

[Options Menu](#)

[Preferences](#)

Other Preferences

Use to select miscellaneous user options.

Play Looped Adjust Pre-roll (0.0 to 10.0 seconds) Default: 1.500 sec

Specifies the number of seconds that will be played before the loop end point when playing in *Loop Preview* mode. To play a loop in *Loop Preview* mode, control-click on the *Play Looped* button. This is useful when loop-tuning a long loop.

Audio Event Locator

The following settings control how the *Audio Event Locator* plays audio. To use the *Audio Event Locator*, click on the overview (above the waveform display) and drag the mouse. Playback is looped around the position until the mouse button is released.

Pre-roll (0.0 to 10.0 seconds)

Default: 0.100 Second

Specifies the location where playback will begin prior to where the cursor is placed in the overview.

Loop Time (0.01 to 30. 0 seconds)

Default: 1.000 Second

Determines the amount of time which will be looped during playback when using the *Audio Event Locator*.

Cut Preview Configuration

When using the *Preview Cut* function in the Edit menu, the *Pre-roll* and *Post-roll* times specify the amount of audio data that is played prior to and after the data which is to be cut.

Pre-roll (0.0 to 30.0 seconds)

Default: 1.500 Second

Specifies the amount of data played prior to a cut region.

Post-roll (0.0 to 30.0 seconds)

Default: 1.500 Second

Specifies the amount of data played after a cut region.

See also

[Options Menu](#)

[Preferences](#)

File Associations

Pressing the *Associate Sound File Extensions* button from the *File* tab in the *Preferences* folder opens the *Sound File Associations* dialog. This dialog provides a method of associating audio files with Sound Forge. When a file is associated with Sound Forge, you can simply double-click on a sound file in the *File Manager* or the *Windows 95 Explorer* and it will open for editing.

File Types to Associate

The list displayed contains all sound file formats that Sound Forge supports. If a sound file extension is associated with an application, then the name of the application is displayed on the right. If a sound file is associated with Sound Forge, then a check mark is placed in the box to the left of the item. To associate an item with Sound Forge, simply check the item and select *OK*.

File Association Details

When a sound file extension is associated with Sound Forge, Windows will open Sound Forge to edit any files with the extension when selected. For example, if you double-click on a file with an extension that is associated with Sound Forge in the *File Manager* or the *Windows 95 Explorer*, Sound Forge will automatically run and the sound file will be opened for editing.

Select All

Pushing this button will select all sound files to be associated with Sound Forge.

Select None

Pushing this button will de-select all sound files so no files are associated with Sound Forge.

See also

[Options Menu](#)

[File Preferences](#)

[Preferences](#)

Storage Directory

The selected directory will be used to store temporary files used by Sound Forge. It is reached from the *Performance Preferences* tab.

See also

[Options Menu](#)

[Performance Preferences
Preferences](#)

