Sound Forge XP 4.0 Help Contents

Sonic Foundry

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Shortcuts

You want shortcuts? We've got shortcuts...

Keyboard Shortcuts Shortcut Menus Mouse Shortcuts

See also

Tips of the Day

Keyboard Shortcuts

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

<u>General Keyboard Shortcuts</u> <u>Edit Commands</u> <u>Misc. View, Option and Special Commands</u> <u>Cursor Movement</u> <u>Selecting Data</u> <u>Navigation and Playback</u>

See also Shortcuts

General Keyboard Shortcuts

File Commands

Press	То
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 XP

Window Management Commands

Press	То
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 Video Preview window.
Alt+F5	Restore the Sound Forge XP application window.
Alt+F10	Maximize the Sound Forge XP 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 horizontally.
Shift+F5	Cascade the data windows.

See also

Edit Commands

Press	То
Control+A	Select all data in the active window.
Control+C	Copy the selected data onto the clipboard.
Control+D	See the Set Selection dialog box.
Control+E	Paste the clipboard contents into a new data window.
Control+F	Crossfade data from the clipboard with the active window.
Control+G	Go to a specified spot in the waveform.
Control+M	Mix data from the clipboard with the active window.
Control+T	Trim (crop) to the current selection.
Control+U	Toggle Undo buffer creation for the active window.
Control+V	Paste data from the clipboard into the active window.
Control+X	Move (cut) the selected data onto the clipboard.
Control+Y	Repeat last process, effect or tool.
Control+Z	Undo the last action.
Control+Shift+Z	Redo the last undone action
Delete	Clear (delete) the selected data; nothing placed on the clipboard.

See also

Misc. View, Option and Special Commands

Press	То
S	Toggle current selection on and off.
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+R	Record new data into a data window.
Escape	Stop or cancel the current action (including playback).
F1	Open the Sound Forge XP Help file to the Contents page.
F10	Make the menu bar active.
F6	Toggle playback scrolling on and off
Shift+F10	Display a shortcut menu.

See also

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 or Markers Exist in the File:Control+ Left ArrowThe previous region or marker boundary.Control+ Right ArrowThe next region or marker boundary.

See also

Selecting Data

Press

Shift+Right Arrow Shift+Left Arrow Shift+Left Arrow Shift+End Shift+Home Control+Shift+End Control+Shift+Home Shift+Page Up 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 previous to 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 or Markers Exist in the File:

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

See also

Navigation and Playback

Press Up Arrow Down Arrow Shift+Up Arrow Shift+Down Arrow Control+Up Arrow Control+Down Arrow Control+Shift+Up Arrow Control+Shift+Down Arrow Numpad 5 Tab Shift+Tab [(open bracket)] (open bracket)] (open bracket)] (open bracket) G or . (period) Spacebar Shift+Spacebar Control+Spacebar Enter Escape	To Increase time magnification (zoom in closer to data). Decrease time magnification (zoom out farther from data). Increase level magnification Decrease level magnification Zoom Selection if a selection exists, Zoom In Full if no selection. Zoom Normal (zooms to default zoom ratio set in Preferences). Pan data window up if zoomed in vertically. Pan data window down if zoomed in vertically. Switch cursor to opposite end of selection Cycle stereo selection from left channel to right channel to both channels Cycle stereo selection from both channels to right channel to left channel. Set Mark In at the current cursor position. Set Mark Out at the current cursor position. Center the cursor in the waveform display. Play or Stop the current data window in default mode. Play All or Pause the current data window. Switch playmode between Normal and Looped. Pause playback; keep the cursor at the current position.
•	

See also

Regions List Shortcuts

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

See also

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 XP...chances are that a Shortcut Menu will appear.

Sound Forge XP Desktop Waveform Display Playbar Selection Status Regions List Time Ruler Display Properties Level Ruler Status Bar Marker Tag Record Dialog Summary Information Picture Video Strip Edit Tool Selector

See also

Sound Forge XP Desktop Shortcut Menu

Right-click the desktop workspace of Sound Forge XP to see this shortcut menu. Use its option to manage sound files on your workspace. You can open recently used files from the *Recent Files* option. You can also quickly reach the *Preferences* dialog from this menu.

See also

Shortcut Menus

Waveform Display Shortcut Menu

Right-click the waveform display of a sound file to see this shortcut menu. Use it to quickly reach the following commands and dialogs: *Zoom In Full/Normal/Out Full, Paste, Crossfade, Mix, Go To, Selection, Scroll Playback, Properties.*

See also

Shortcut Menus

Playbar Selection Status Shortcut Menu

Right-click any of the playbar's selection status fields to see this shortcut menu. 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, Update, Copy onto Clipboard.*

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* dialog for adding regions and markers. It also lets you change the status format.

Display Properties Shortcut Menu

This shortcut menu is reached by right-clicking on the *Edit Tool* selector at the upper left corner of the data window. It allows you to set the *Status, Time Ruler, Time Zoom, Overview, Level Ruler, Video, Level Zoom, Scrollbar, and Data Window Only* options.

See <u>Display Properties</u> for a complete description of these options.

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

Marker Tag Shortcut Menu

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

<u>Go To</u> <u>Delete</u> <u>Edit</u> <u>Update</u>

See also Shortcut Menus

Go To Marker

Choose this option to move the cursor to the marker's position.

Delete Marker

Choose this option to delete the current marker.

Edit Marker

Use the *Edit Region/Marker* dialog to change the name, type, or location of the marker.

Update Marker

If you've made changes to the file, choose this option to update the marker.

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, such as 32,000. Note that if you do not *Resample* the data, the pitch will change also. To resample, choose <u>Resample</u> from the Process menu.

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

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 clipResolutionShow LabelsHold PeaksHold ValleysReview Pre/Post RollGo toSelectionWindowNew WindowAggressive Update

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.

<u>-90 to 0 dB</u> -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 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 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 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 Meters|Show Labels

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

See also

Record Meters|Hold Peaks

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

See also

Record Meters|Hold Valleys

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

See also

Blinking Status

Enables the blinking record status underneath the record meters.

See also

Record|Review Pre/Post Roll

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

See also

Record|Go to

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

See also

Record|Selection

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

See also

Record|Window

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

See also

Record|New Window

Sets Record to write data to a new sound file.

See also

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

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 file's embedded summary information.

See also

Summary Information Shortcut Menu

Video Strip Shortcut Menu

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

<u>Video Strip|Copy</u> <u>Video Strip|Go To</u> <u>Video Strip|Animate</u> <u>Video Strip|Number Frames</u>

See also Shortcut Menus

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.

See also

Video Strip|Go To

Moves the cursor to the beginning of the frame.

See also

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|Number Frames

Displays frame number on each frame.

See also

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 side of the data window which allows you to switch between the Edit and Magnify 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|Time Ruler

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

See also

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|Overview

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

See also

Edit Tool Selector|Video

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

See also

Edit Tool Selector|Level Ruler

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

See also

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|Scrollbar

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

See also

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

Mouse Shortcuts

Select All Zoom Both Time and Level Magnify Mode Return Control Value to Default Main Status Bar Selection Status Bar Play Normal Button Slow and Fast Selection Scroll Toggle Ruler scrolling speed control

See also

Shortcuts

Select All

Double-click on the waveform display to select the entire sound file.

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 XP to *Magnify* mode, which zooms to fit the selected area.

See also

Return Control Value to Default

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

Main Status Bar

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

See also

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

Play Normal Button

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

See also

Slow and Fast Selection Scroll Toggle

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

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 XP for 32 Bit Windows

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

Sound Forge XP for 32 Bit Windows is a **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 XP.

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

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

When running Sound Forge XP 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 XP for 32 Bit Windows when running under Windows NT:

Ϋ́ Not all sound cards have drivers available for Windows NT. Contact your manufacturer for availability of Windows NT drivers.

What's new in version 4.0

Below are just a few of the new features in Sound Forge XP 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 XP, 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* option (Special menu) makes it easy to select from all the previously existing states.

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?

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 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 or crossfaded. Remember, hold down 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.

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, RealMedia, and Active Streaming Format)

Sound Forge XP 4.0 now supports many common Internet audio (and video) file formats. We've added Progressive Networks' <u>RealMedia</u> support and the new Active Streaming Format_ (ASF) used by Microsoft's <u>NetShow</u> On-Demand. ASF files are designed for both audio and video streaming over the Internet. In addition, support has been added for the NeXT/Sun .au format for working with Java scripts.

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 XP 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

Smooth/Enhance

Quickly add or remove high frequency content.

DTMF dial tone generator

Generate touch-tone triggers in Sound Forge XP.

Graphic EQ

It's faster and more accurate.

Chorus and Flange

Improvements have been made to apply smoother pitch modulation.

Time Compress/Expand

Enhanced Algorithms allow you to stretch and squish farther than before without artifacts.

Record

New Meters

Higher accuracy meters allow for finer adjustment of input levels.

Markers and Regions

XP now allows Markers and Regions!

Markers and Regions allow for quick editing and navigation and aid in authoring interactive internet content through RealMedia and Microsoft NetShow.

Markers to Regions function

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

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 and Regions annotations

The name of markers and regions 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 region.

Shift+double-click a marker to move the marker to the current cursor position.

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

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

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

And more...

- Ÿ 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.
- Ÿ You can drag selection edges with the mouse.
- Ÿ When playing in Looped Mode, playback always follows selection and cursor changes.
- Ÿ Support for the InterVoice file format for telephony applications.
- Ϋ́ Envelope controls now multi-point editing as well as the ability to display the sound data in the envelope control.
- Ϋ́ A free disk space display has been added to the status bar. You can turn this on or off in the Preferences section.
- Ϋ́ 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.
- Ϋ́ 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 XP is loaded with them.

Troubleshooting

Use this before calling the Tech Support line.

Tip of the Day

Displays a new Sound Forge XP tip.

The Sonic Foundry Home Page

Takes you to our little plot in cyberspace.

About Sound Forge XP

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

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.sonicfoundry.com). Here you will find downloadable demos and updates, technical information, and news of what's new with the folks here 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 feedback@sonicfoundry.com). We'd love to hear how Sound Forge XP 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: 4/17/97

Sound Forge XP Main Screen Basics

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

<u>Using the Mouse</u> <u>The Main Screen</u> <u>Toolbars</u> <u>The Data Window</u> <u>ToolTips</u> <u>Getting On-line Help</u>

Using the Mouse

Although using the mouse in Sound Forge XP is not required, it will make your editing sessions easier. Once you become familiar with Sound Forge XP 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 documentation lets you know whether you need to right-click or left-click on an item to execute a specific function. When there is no specification, assume we mean left-clicking. Right-clicking is often used to reach Shortcut Menus.
- **Double-clicking** Double-clicking means clicking 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 means clicking the right mouse button while holding the left mouse button to cycle through various options. This is a great shortcut for procedures such as drag-and-drop and changing the *Magnify Tool* mode of operation.
- **Shift-clicking** Shift-clicking is holding down the *Shift* 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 *Control* key on the keyboard while clicking the mouse. Control-click to modify the operation of a normal click.
- **Dragging** Holding down the mouse button while you move the mouse pointer is known as dragging. Drag 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.

The Main Screen

When you start Sound Forge XP, you see the main screen, or workspace, where you will do all your editing. When you first open Sound Forge XP, no Data Windows are open and you will need to either open an existing sound file or create a new one. See <u>Opening an Existing File</u> and <u>Creating a New</u> <u>Window</u> for more information.

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Click an area of the screen illustrated below for a description.

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 XP Workspace

This is the background area behind the data windows. You can drag sections here to create new data windows.

Toolbars

You can use three toolbars in Sound Forge XP: the Standard, Transport, and Status toolbars. The toolbars contain buttons which are used to quickly execute commands.

- Standard
 Provides quick access to many commonly used Sound Forge XP File and Edit menu options.
- TransportProvides the audio transport buttons: Record, Play All, Play Normal, Pause, Stop,
Go to Start, Rewind, Forward, and Go to End.
- **Status** Contains operations to set the display *Status Format* as well as the Snap To operations.

You can drag and drop these toolbars anywhere on the screen, resize them, and remove them. When you drag a toolbar 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, click 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 choose *Toolbars* from 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.

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Click an area of the Data Window to see a description.

Title Bar

Shows the file name or, if present, the title stored in the *Summary* 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 *Level Ruler* 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 *Time Ruler* shortcut menu. Drag to scroll the data window.

Edit Tool Selector

Use this button to toggle between the *Edit, Magnify, and Pencil Tools*. Note: The *Pencil Tool* 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: *Go to Start, Go to End, Stop, Play Normal,* and *Play Looped*. 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 *Go To* dialog. Double-click on either of the other two fields to reach the *Set Selection* dialog. Right-click to reach the *Status Format* shortcut menu.

Waveform Display

Shows a graphical representation of the sound file. The horizontal axis represents time (marked in the *Time Ruler*) and the vertical axis represents amplitude (marked in the *Level Ruler*). Right-click anywhere on the waveform display to reach the *Waveform Display* 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.

Time Zoom Resolution

(Also called *Zoom Ratio*) 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 XP 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 XP 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:

- Υ 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.
- Ÿ Press the *Help* button or the *F1* function key in a dialog. This will take you straight to the help information on the dialog.
- Ÿ 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 XP, 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 XP, 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 XP 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. Press the *Stop* button to stop 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 XP 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 XP.

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 XP allows you to embed descriptive titles as well as copyright information and other text fields for any .WAV file.

Sound files in Sound Forge XP 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 right).

Using the Transport versus the Data Window Playbar

There are a number of ways to play your sound files within Sound Forge XP. 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 Data Window Playbar



Click a button to see a description.

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



Click a button on the *Transport* toolbar to see a description.

Play Normal

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.

Play Looped

Plays the currently selected data in a loop: from beginning to end, then to the beginning again.

Record

Brings up the *Record* dialog which allows you to record to either a new or existing window.

Shortcut: Control+R

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: Shift+Spacebar

Play

Plays the file using the current play mode. To set the play mode select one of the play buttons on the playbar as described above. The last button used becomes the active play mode.

Shortcut: Spacebar

Pause

Works like the *Stop* button in that it stops play of the current file. However when selecting *Pause* the cursor is placed at the current play position rather than back to where it was when play was selected.

Shortcut: Enter during playback.

Stop

Stops play and positions the cursor back to where it was prior to selecting play.

Shortcut: Spacebar 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

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

If you're working with the TUTOR1 file, you can 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

If you're working with TUTOR1, 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. 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."

Saving a File

To save a data window, you first need to make it the active window. For this example, 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 XP displays the *Save As* dialog. If the file had been opened or previously saved by Sound Forge XP 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 XP 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 shows you the basic editing operations of Sound Forge XP.

Common Edit Operations Making a selection Copy, Paste, Cut, Undo, Redo Cropping Mixing Status Formats Magnification and Zooming

Common Edit Operations

You can find the most common editing options on the Edit menu and the Standard toolbar. Most of these options use 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.
Сору	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 can use the TUTOR1.WAV file. If it is not currently open please open the file now. (See <u>Opening an Existing File</u> for instructions.)

Making a Selection

- 1. If TUTOR1 is not the active window, activate it by clicking on its title bar.
- 2. 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*.
- 3. Select the word "Wow" with the mouse by clicking and dragging. 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 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 XP!

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 recut 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.

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.

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.

- 1. 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.
- 2. Once you have the selection, select *Trim/Crop* from the Edit menu or the Standard toolbar. After cropping you will have only "Wow, sound editing just gets easier" left in the window.
- 3. At this point let's close the "Wow, sound editing..." window. Either click 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.
- 4. 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 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. We are going to mix this sound with TUTOR1; the "Wow, sound editing..." window.
- 2. 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 XP keeps track of the format type for each individual window.
- 3. To make the windows easier to view during the mix operations you may want to maximize Sound Forge XP by clicking the Maximize Window button in the upper right corner and selecting the Tile Vertically item from the Window menu. This will arrange the windows vertically fully utilizing the Sound Forge XP workspace and make things easier to see.
- 4. 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.
- 6. 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.
- 7. Select the Mix command from the Paste Special option in the Edit menu or the Standard toolbar. The Mix dialog now appears. Keep both levels at 0 dB and select OK.
- 8. 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.
- 9. Select the Undo Mix command from the Edit menu to put TUTOR1 back to its original state.
- 10. 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 choose Play for the result. Notice how the drum hit sound and the "Wow" sound overlap each other.

Status Formats

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

Display lengths and positions 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. Then choose an option to set the status format for the current data window.

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.

- 1. First, select the Samples format from the Status Format menu under the Options menu.
- 2. 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.
- 3. 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.
- 4. 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 XP 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.

<u>Time Ruler Zooming</u> <u>Level Ruler Zooming</u> <u>Using Zoom Selection and Zoom Out</u> <u>Magnify Tool</u>

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 *Zoom* commands from the *Data Window* shortcut menu.

When Sound Forge XP 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, select the area you want to magnify, and then choose *Zoom Time*|*Selection* from the View menu or the *Data Window* shortcut menu. Sound Forge XP then calculates the minimum zoom ratio which will allow the full selection to fit on the screen and then centers the selection within the waveform display.

This will not affect the zoom ratio of the *Level Axis*. Only the *Zoom Selection* command in the *Level Ruler* 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 you choose the *Magnify Tool*, 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 <u>Using the Mouse</u>). 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 XP:

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 Drag And Drop Operations Editing Stereo Files

Making a selection using the Set Selection dialog

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

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

You can perform many commonly used functions in Sound Forge XP a number of 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 XP 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. See <u>Shortcuts</u> for more information.

Extending a Selection the Mouse

After selecting a section of sound data, you may 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 XP allows you to update the selection by grabbing a selection edge and moving it.

- 1. Once you have established a selection, place the mouse over the selection start or end point. The cursor changes to a bi-directional arrow.
- 2. 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.

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. Press *Home* to move to the start and *End* to move to the end. 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
Shift+Right arrow	the next screen pixel.
Shift+Left arrow	the previous screen pixel.
Shift+End	the last sample visible in the waveform display.
Shift+Home	the first sample visible in the waveform display.
Press	To select
Shift+Page Up	10% of the current view previous to the cursor position.
Shift+Page Down	10% of the current view past the cursor position.
Shift+Control+Right arrow	2% of the current view past the cursor position (see below).
Shift+Control+Left Arrow	2% of the current view previous to the cursor position (see below).
Shift+Numpad +	the next sample.
Shift+Numpad -	the previous sample.
Shift+Control+Numpad +	10 samples past the current position.
Shift+Control+Numpad -	10 samples previous to the current position.

In Windows 95, *Shift Numpad* + will not work unless the *Numlock* indicator on the keyboard is lit.

As you can see from the above lists, Sound Forge XP 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 <u>Shortcuts</u>.

Example using keyboard shortcuts

Let's take a quick look at how to use some of these keys.

- 1. Open the file TUTOR1.WAV.
- 2. Make sure that you're using the *Edit Tool* by selecting it in *Tool* under the Edit menu.
- 3. Select the word "Wow" with a generous amount of space on each side of the word. When you press the *Play* button you should hear a little bit of silence, "Wow", and a little more silence.
- 4. 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.

- 5. 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 release the mouse button.
- 6. 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 manner until you've got the selection just right.
- 7. 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

- 1. 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.
- 2. 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.

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:

- Ÿ Use the Go To item in the Edit menu
- Ÿ Right-click on the waveform display and select Go To from the shortcut menu
- Ÿ Double-click on the left-most selection status field
- Ÿ Press Control+G

The *Go To* dialog is much like the *Set Selection* dialog. You can type in the location where you want to go or use one of the predetermined locations from the list.

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. See also <u>Creating a New Window with Drag and Drop</u>.

- 1. Open two sound files, such as TUTOR1.WAV and TUTOR2.WAV, and select *Tile Vertically* from the Window menu just to make them easier to work with. TUTOR2 is the source window and TUTOR1 is 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.

For example, highlight all the data in TUTOR2 by making it the active window and dragging to select in the waveform display. You can also choose Select All from the Edit menu to select the complete waveform. 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 of the TUTOR1 data window. This is where you want the mix, paste, or crossfade to occur. 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).
- 4. **Mix**: Let go of the left mouse button to see the *Mix* dialog. You see the Mix 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.

Crossfade: To perform a crossfade instead of a mix, hold the *Control* key down while letting go of the mouse button. You then see the crossfade block. This X-shaped block shows you where one sound begins to fade out and the other to fade in. The cursor position marks the beginning of the process. By dragging to the left and right, you can adjust the length of the crossfade. Once you have determined where you would like the crossfade to begin, let go of the mouse button. You should hear the end of TUTOR1 fade out as the beginning of TUTOR2 fades in.

Paste: 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.

5. Press the *Play* button to hear the results.

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 among 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.

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 XP workspace. A new data window is automatically created containing the data from the selection.

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 XP 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.

- 1. 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.
- 2. 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.
- 3. 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.
- I Cursor when selecting both channels of a stereo file or when in a mono file.
- I^L Cursor when selecting the left channel of a stereo file.
- I 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 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 multitrack 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 in 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.

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 XP 4.0 was installed (in most cases this will be C:\Forge40). To convert the file from mono to stereo, rightclick 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 see 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.

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:

- Ÿ 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.
- Ÿ *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. Select a 16 to 8-bit conversion option:

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.

Dither 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 <u>Resample</u> for more information) 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 XP. 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.

- 1. If you don't have TUTOR1.WAV open, open it up now. From the File menu, select the Save As command.
- 2. On the Save As dialog, there are three controls used for file conversion:

Save as type. This lists all the available output file types. 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.

Format. 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.

Attributes. 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 XP automatically converts any file you save to an acceptable format.

Embedding Summary Information in WAV Files

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

Summary Information

When editing .WAV 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 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 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, see General Properties.

Default

If you select the *Default* button, 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 file format 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 XP 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 XP 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 XP will only prompt you to save if such data exists.

Creating RealMedia and ASF (NetShow) files

Sound Forge supports two different formats for creating files that are optimized for streamed playback over (usually) slow networks such as the Internet. Both Progressive Network's RealMedia (.RM,.RA), which includes RealAudio and RealVideo and Microsoft's Active Streaming Format (.ASF) files can be created in Sound Forge XP. To play these files, you must have the appropriate player application installed in your computer.

Using Sound Forge XP to create RealMedia files Using Sound Forge XP to author NetShow content

Using Sound Forge XP to Create RealMedia Files

Sound Forge XP gives you the ability to save sound files in RealMedia formats (RealAudio and RealVideo) along with editing tools that help maximize the audio's fidelity. This section first explains how to save files in RealAudio format, then covers saving files in RealVideo format. All of the tips for converting files to RealAudio also apply for converting AVI files to RealVideo. For more information on authoring RealMedia content, refer to Progressive Networks' RealMedia home page (www.real.com).

Preparing RealAudio Files

Preparing RealAudio files is a three-step process:

<u>Recording the audio</u> <u>Preparing a sound file for RealAudio encoding</u> <u>Setting the RealAudio text fields</u>

Saving RealVideo Files

Saving an AVI File in RealVideo Format

See also RealMedia Save Options

Recording the audio

To record a sound file in Sound Forge XP:

- 1. Click the *Record* button. The Record window appears displaying the settings from the last clip recorded.
- 2. Use the defaults or click the *New* button to change the *Sample Rate*, *Sample Size* or number of channels of the source file.
- 3. Check the *Monitor* box and play the clip to be encoded. Check for clipping. You want the levels to be mostly in the green area, a little in the yellow and only occasionally in the red. If Sound Forge XP displays a red Clip warning above the *Monitor*, you should:
 - Ϋ́ Stop the input source.
 - Ϋ́ Click the *Reset* button.
 - Ÿ Lower the volume of the input source either by turning down the output of your sound source or by turning down the input volume on your sound card's mixer window. In general, you don't want any of your levels too high or too low.
 - Ÿ Check the levels again. It is better to be conservative with the starting levels.
- 4. Check the *DC Adjust* box. Sound Forge XP can automatically calculate how much offset is necessary. With your sound source connected but no sound playing, click the *Calibrate* button. If you always use the same input device, you only have to perform this step once. Otherwise, you should re-calibrate each time you switch input devices.
- 5. Click the *Record* button on the Record window to start the record process.
- 6. When the file is recorded completely or when you click the *Stop* button, a graphical display of the wave file appears.
- 7. Click the *Close* button to close the Record window.

Preparing a sound file for RealAudio encoding

To prepare a recorded file for RealAudio encoding:

When you have finished recording a file, it is recommended that you perform the following processes on the file before saving the file in a RealAudio format.

Note: Sound Forge XP allows you to save preset settings for each of these processes. So once you set them up the first time, you can always use the same settings.

Normalize

This process ensures that the loudest part of the file is at the maximum recording level.

- 1. Press Ctrl-A to select the entire file.
- 2. Click *Normalize* from the Process menu. The Normalize window appears.
- 3. Click the *Peak Level* box.
- 4. Move the Normalize to slider to -.50 dB (94.41%).
- 5. Click the OK button. Sound Forge XP normalizes the file.

Compress

If you have not already audio compressed the clip, you can do so now. RealAudio files benefit from compression because it increases the overall volume and intelligibility of the sound. The suggested settings use very light compression.

- 1. Press *Ctrl-A* to select the entire file.
- 2. Click Dynamics from the Effects menu and select Graphic. The Graphic Dynamics window appears.
- 3. Check the Sync Stereo Gain box.
- 4. Check the Auto Gain Compensate box.
- 5. Move the Threshold slider to -10.00 dB.
- 6. Move the Ratio slider to 2.0:1.
- 7. Set the *Attack* box to 1.0 milliseconds.
- 8. Set the *Release* box to 300.0 milliseconds.
- 9. Click the OK button. Sound Forge XP compresses the file.

Setting RealAudio text fields

To set the RealAudio text fields:

- 1. Click *Properties* on the File menu. The Properties window appears.
- 2. Click the *Summary* tab.
- 3. Enter the Title, Engineer, and Copyright information. This information will appear in the Title, Author, and Copyright text boxes on the RealAudio Player.
- 4. Click the OK button.
Saving a File in RealAudio Format

To save a file in RealAudio format:

- 1. Select Save As from the File menu. The Save As dialog appears.
- 2. Enter the file name of the file you are saving.
- 3. Select *RealMedia* (*.*rm*;*.*ra*) as the *Save as type*. The default extension for this type of file is *.rm*. To save using the *.ra* extension, type *.ra* at the end of the file name.
- 4. Select *Save*. The *RealMedia Save Options* dialog appears. The options in this dialog are covered in the below section, *RealMedia Save Options*.
- 5. Select the *Audio compression* type from the drop-down list. A brief description of the highlighted compression type is displayed below this list.
- 6. If the file you recorded was stereo and you selected a mono compression type, select the *Stereo to mono conversion*. It is recommend that you use the *Both Channels* option.
- 7. Select *OK*. After the file has been saved, a message appears asking whether you want to listen to the saved file through the RealPlayer. This message verifies that you have saved the file in RealAudio format. You can listen to the file by clicking the Yes button if you have the RealPlayer installed.

See

RealMedia Save Options

Saving an AVI File in RealVideo Format

To save an AVI file in the RealVideo Format:

- 1. Select Save As from the File menu. The Save As dialog appears
- 2. Enter the file name of the file you are saving.
- 3. Select *RealMedia* (*.*rm*;*.*ra*) as the Save as type.
- 4. Select *Save*. The *RealMedia Save Options* dialog appears. The options in this dialog are covered in the below section, *RealMedia Save Options*.
- 5. If you wish to include the audio in the new RealVideo file, make sure the *Save Audio Stream* option is checked. Then, select the *Audio compression* type from the drop-down list. A brief description of the highlighted compression type is displayed below this list.
- 6. If you wish to include the video in the new RealVideo file, make sure the *Save Video Stream* option is checked. Then, select the *Video compression* type from the drop-down list. For a brief description of the two compression types, see the below section, *RealMedia Save Options*.
- 7. The final step before saving the file is to specify the *Desired maximum data rate* in <u>total</u> kilobits per second (Kbps). This option is only available if you are saving a video stream. Refer to the documentation for *Desired Maximum Data Rate* in the *RealMedia Save Options* section for more information.
- 8. Select *OK*. After the file has been saved, a message appears asking whether you want to listen to the saved file through the RealPlayer. This message verifies that you have saved the file in RealVideo format. You can view the file by clicking the *Yes* button if you have the RealPlayer installed.

See RealMedia Save Options

Using Sound Forge XP to create ASF files

This tutorial shows how to create Active Streaming Format (ASF) files using Sound Forge XP. ASF is a multimedia file format used for streaming information across the Internet using Microsoft NetShow. With ASF files, you can deliver multimedia content, audio or video, open Web pages, or deliver scripting commands to the client computer.

Sound Forge XP includes support for saving WAV and AVI files and their associated markers and script commands in an ASF file. ASF files can be played off of your local drive, streamed off of an HTTP server, or optimally stored and streamed off of a NetShow server.

NetShow is a scaleable solution and ASF facilitates authoring of streaming content for a variety of bitrates. This tutorial will cover creating low bit-rate and relatively high bit-rate video-based ASF content, although audio-only content is also an option.

NetShow Requirements NetShow Tutorial

NetShow Requirements

In order to play ASF files you must install the *Microsoft NetShow Player* and *Tools*. The Tools install the encode codec for the VDOnet VDOWave video codec. Both of these components are available on the Sound Forge XP CD-ROM and can also be downloaded free of charge from the NetShow web page.

To install the client, select *Microsoft NetShow Player* from the list in the main setup program on the Sound Forge XP CD-ROM. This will run Microsoft's setup program for the player. Optionally, you can run NSOPLAY.EXE directly from the \EXTRAS\NETSHOW2.0\X86 folder on the Sound Forge XP CD-ROM.

To install the ASF authoring and conversion tools, select *Microsoft NetShow Tools* from the list in the main setup program on the Sound Forge XP CD-ROM. This will run Microsoft's setup program for the NetShow Tools. Optionally, you can run NSOTOOLS.EXE directly from the \EXTRAS\NETSHOW2.0\X86 folder on the Sound Forge XP CD-ROM.

Further documentation for this software is available on the NetShow web page. At this time, the components shipped on the Sound Forge XP CD-ROM are for the x86 and Pentium platforms only. See Microsoft's web page for availability of components for other platforms.

NetShow Tutorial

NetShow is a scaleable solution and ASF facilitates authoring of streaming content for a variety of bitrates. This tutorial will cover creating low bit-rate and relatively high bit-rate video-based ASF content-although audio only content is certainly an option.

Converting a High Bit-Rate AVI to ASF Authoring a Low Bit-Rate ASF Sample HTML Using an ASF File Calculating Appropriate Bit Rates for the Internet What Do I With My ASF Files

See also ASF Save Options

Converting a High Bit-Rate AVI file to ASF format

This section of the tutorial demonstrates how you can create high bit-rate video content and convert it to ASF. SAMPLE1.AVI is a short video clip that uses the Intel Indeo Interactive VCM video codec installed with the NetShow Player. The Intel codec is well suited for mid to high bit-rate video. The audio track in this file is PCM format.

1. From the File menu, open SAMPLE1.AVI from the \SAMPLES\NETSHOW folder on the Sound Forge XP CD-ROM. From the File menu select *Properties*. Select the *Format* page and set the audio *Sample Size* to 16 bits. Make sure that *Channels* is set to mono. Select *OK*.

Note that editing and playback performance of an AVI file will suffer greatly unless you first copy it to your hard drive. It is strongly recommended that you always copy AVI files that you are editing onto your hard disk.

- 2. From the Process menu select the *Resample* command. The *Resample* dialog will appear. Set the new sample rate to 8,000 Hz and select *OK*.
- 3. From the Process menu select the Normalize command and select OK.
- 4. Now convert the edited AVI file to an ASF file by saving it on your hard drive. Select the *Save As* command from the File menu. Select *Active Streaming Format* (*.*asf*) from the *Save as type* drop-down list.
- 5. The NetShow Player installs the Lernout & Hauspie CELP 4.8kbit/s ACM codec (see previous section), which is excellent for low bit-rate voice audio. Select this codec from the *Format* drop-down list. Select *Save*. (Note that it is not a requirement of ASF to convert the audio from PCM. We perform the conversion as part of this tutorial to demonstrate how to take advantage of the installed codecs.)
- 6. The ASF Save Options dialog presents the settings for the Active Streaming Format file. Select OK to select all the default settings.
- 7. Next, the *Video Save Options* dialog appears. It allows you to view the audio and video properties of the AVI file. Select *OK*.
- 8. The *Compression Options* dialog presents the available video compression options. Press *OK* to select the Indeo codec.
- 9. When the ASF file has been created Sound Forge XP will prompt you to play it. Select Yes. This will start the NetShow Player.
- 10. From the File menu in the NetShow Player select *Properties*. Note the following *Properties* pages:
- •The General page contains the general information contained within the AVI file.
- •The Details page shows that the ASF file has a bit-rate of 875,000 bps.
- •The Codecs page displays information about the codecs being used with the file.

Authoring a Low Bit-Rate ASF File

This section of the tutorial creates a low bit-rate video suitable for playback on the Internet at 28.8 Kbps. SAMPLE2.AVI is a smaller version of SAMPLE1.AVI. It uses the Intel Indeo Interactive codec and has been re-sized to 160 x 120 at 6 frames per second with one key frame every 36 frames. The VDOnet VDOWave codec and the Intel Indeo Interactive codec require that the video frame size be a multiple of 16 (i.e., 160 x 120, 320 x 240, etc.). The audio track in this file is PCM. The steps below describe how to use the VDOnet VDOWave video codec, which is an exceptional codec for low bit-rate content.

1. From the File menu, open SAMPLE2.AVI from the \SAMPLES\NETSHOW folder on the Sound Forge XP CD-ROM. You will be prompted with a message from Sound Forge XP asking if you want to disable the video strip. Select *No*.

Note that editing and playback performance of an AVI file will suffer greatly unless you first copy it to your hard drive. It is strongly recommended that you always copy AVI files that you are editing onto your hard disk.

- 2. From the File menu select *Properties*. Select the *Format* page and set the audio *Sample Size* to 16 bits and verify that *Channels* is set to mono. Select *OK*.
- 3. From the Process menu select the *Resample* command. In the *Resample* dialog, set the *New Sample Rate* to 8,000 and select *OK*.
- 4. From the Process menu select the Normalize command and select OK.
- 5. Now add Markers and ASF Script Commands. From the Special menu select *Region List/Add*. Add the following three Markers:

Туре	Name	Time
Marker	ASF CAPTION: John is here to	00:00:01.000
Marker	This is the middle of my ASF	00:00:09.000
Marker	ASF CAPTION: John is not here	00:00:10.000

Note that you can drag and drop the markers along the time line once they have been created and during playback. Markers that start with the preface ASF are processed as stored as script commands within the ASF file. For more information, see Sound Forge XP online Help.

- 6. Now save the edited AVI file to an ASF file. Select the *Save As* command from the File menu. In the *Save As* dialog, Select *Active Streaming Format (.asf)* from the *Save as type* drop-down list.
- The NetShow Player installs the Lernout and Hauspie CELP 4.8kbit/s ACM codec, which is excellent for low bit-rate voice oriented audio. Select this codec from the *Format* drop-down list. Select *Save*. The *ASF Save Options* dialog presents the settings for the ASF file.
- 8. Set the *Maximum lead time* setting to 5,000 and select *OK*. Note that 5,000 is the lead time setting that should be use when authoring low bit-rate video ASF files.
- 9. Select OK to close the Video Save Options dialog.
- 10. The *Compression Options* dialog presents the available video compression options. Select the *Options* button, and in the *Video Compression* dialog select the *VDOnet VDOWave* codec from the

Compressor drop-down list.

- 11. Select the *Configure* button to bring up the *Encoder Configuration* dialog. This dialog allows you to set the specific bit-rate for the video stream. Drag the slider to the left until it is at the 14 Kbit/sec setting. Select *OK*.
- 12. Back at the *Video Compression* dialog verify that the check box next to the *Key Frame Every* field is enabled and set the *Frames* edit field to 60. This will set a key frame every 10 frames. It is necessary to reduce the number of key frames to attain the desired low bit-rate. Select *OK*.
- 13. Back in the *Compression Options* dialog verify that the *Interleave Every* check box is clear. Select *OK* to start building the ASF file.
- 14. When the ASF file has been created Sound Forge XP will prompt you to play it. Select Yes. This will start the NetShow Player.
- 15. From the File menu in the NetShow Player select *Properties*. Note the following *Properties* pages:
- The *Details* page shows that the ASF file has a bit-rate of 19,332 bps. Achieving the exact bit-rate for a video stream will vary depending on the content. Complex video may require fewer frames per second or less frequent key frames. Frame rate, key frames, and video frame size are all variables that affect bit-rate.
- The *Codecs* page displays information about the codecs being used with the file.

Adding an ASF File to a Web Page

On the Sound Forge XP CD-ROM in the \SAMPLES\NETSHOW folder there is an HTML file containing an example of how to use the NetShow Active Control to add an ASF file to a web page. This sample will be used with the low bit-rate video ASF file, SAMPLE2.ASF. Note that the HTML sample requires Microsoft Internet Explorer v3.0 or better.

- 1. Copy SAMPLE.HTM to the folder on your hard drive that contains SAMPLE2.ASF.
- 2. Modify the *FILENAME* parameter within the NSOPlay object to reference your local SAMPLE2.ASF file (a fully qualified path is required).
- 3. Open the edited SAMPLE.HTM file within your browser. There are additional instructions on the sample page.

Calculating Appropriate Bit-Rates for the Internet

When authoring Internet file formats, it is important that the content not exceed the bit-rates of your targeted modem connection. The reason is that modem connections to the Internet are often less than 14.4 Kbps or 28.8 Kbps. It is important not to consume all the bandwidth available to the browser, control protocols, etc. The following guidelines suggest a combined (audio and video) bit-rate with error corrections slightly below the targeted Internet connection:

Internet Connection	Authored Bit-Rate (audio + video)	Total Bit-Rate w. Error Correction
28.8 Kbps	18,500 bps	20,813 bps
14.4 Kbps	10,000 bps	11,250 bps

For example, in the *Authoring a Low Bit-Rate File* section above, the audio was set to 4.8 Kbps and the video was set to 14 Kbps. With the addition of error correction to the stream, the total ASF file bandwidth as displayed in the NetShow Player is 19,332 bps.

What Do I Do with My ASF Files?

ASF files are specifically designed for network streaming and are authored for specific network bandwidths. For network streaming, ASF files can be stored on an HTTP server or optimally off of a Microsoft NetShow server. The NetShow Player can play back ASF content as an embedded Active control within an OCX container such as Microsoft Internet Explorer 3.0 or Microsoft Visual Basic. The NetShow Player can also run in standalone mode, such as when an ASF file is launched as a link within any Internet browser. ASX files (ASF Redirector Files) are meta files that pass the ASF file link from the browser to the NetShow Player so it can start streaming.

For more information, see the NetShow web site. The entire Microsoft NetShow product can be downloaded free of charge from www.microsoft.com/netshow.

Recording

Sound Forge XP 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 Remote recording

Recording basics

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, see <u>Troubleshooting</u> 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 XP 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

Choose one of the following modes of recording.

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

The Multiple Takes mode allows multiple takes to be recorded. 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 XP 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 XP 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

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).

Checking Record Levels

Sound Forge XP 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. See <u>Troubleshooting</u> 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 XP begins to listen to your recording device. This is just like recording except that Sound Forge XP 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 XP 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).

Remote Recording

By selecting the *Remote* button on the Record dialog you enter into *Remote Recording* mode. *Remote Recording* mode hides the main Sound Forge XP 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, or CD audio. 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 menus include options for applying an effect on the sound data, displaying information, or even synthesizing new sounds.

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 Processing shortcuts

Applying Simple Processes and Effects

To apply an effect, first select a section of data on which you wish to operate.

If you don't have a selection when you perform an effect which requires one, Sound Forge XP 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

- 1. If you used the Reverse effect, select the *Undo Reverse* item from the Edit menu to put TUTOR1 back to its original state.
- 2. 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.
- 3. 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.
- 4. 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.

These examples show you how easy it is to apply effects to files. Sound Forge XP has many 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 <u>Effects</u>, <u>Process</u>, or <u>Tools</u> menus.

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*, and *Pitch Bend*.

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.

- 1. 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.
- 2. Select the entire left channel by double-clicking 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.
- 3. Now select the *Reverse* item from the Process menu. You will notice that only the left channel data is reversed.
- 4. Press the *Go to Start* button on the playbar to clear the selection and place the cursor at the start of the file.
- 5. 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.

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 in all function dialogs.

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 XP's different controls used to enter function parameters. Here's a list of controls used in Sound Forge XP:

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

To see examples of these common controls, display the Dynamics dialog box. Choose Dynamics from the Effects menu. The fader controls the *Threshold level* and the trackbar adjusts the *Ratio*.



Fader:

Trackbar:



To change the parameter values, just left-click the box (or "thumb") that slides along the fader or trackbar. Drag left and right or up and down to increase or decrease the value.

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 *Dynamics* dialog, you can edit the *Attack time* and *Release time* parameters 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

Sine 💌

For an example of this control, display the *Pitch Bend* dialog box. Choose *Pitch Bend* from the Effects menu. 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 *Pitch Bend* dialog uses a drop-down list to select the channels of a stereo source.

Push Button

<u>H</u>i!

To use a push button, left-click on it or press the *Spacebar* while it is selected. In the *Pitch Bend* dialog, the *OK, Cancel, Help,* and *Reset* Envelope 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 *Simple Synthesis* dialog is a group of radio buttons.

Check Box

🔽 Create <u>u</u>ndo

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. The *Create Undo* and *Show Wave* controls on the *Pitch Bend* dialog are check boxes.

Envelope Graph

Use envelope graphs 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 Envelope* 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 *Graphic Fade* 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.

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 XP, 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 XP. Holding down the *Shift* key while selecting *Repeat* takes you to the last function's dialog.

Sound Processing Techniques

This section covers basic concepts related to some of Sound Forge XP's processing functions used for modifying (or creating) sound.

Delay/Echo Reverb Chorus Flange Noise Gate Compression Changing Time Duration Synthesis Graphic EQ

Delay/Echo

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

- 1. Open TUTOR1.WAV and select *Delay/Echo* from the Effects menu. Enter 0.4 in the Delay Time field and choose OK. When you play the sound, you hear a second copy of the sound 0.4 seconds after the original. Changing the *Delay time* determines the time between the original and echoed sound.
- 2. If you want an echo effect that contains more than a single copy of the original sound, check the *Multiple delays/echoes* box. The *Decay time* determines how long it takes for these echoes to fade out.
- 3. 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.

Reverberation

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 thousands of echoes combined into a smoothly decaying reverberation.

- 1. Open TUTOR1.WAV and select *Reverb* from the Effects menu. Select the *Bright hall* button, choose OK, and play the sound. Notice how the imagined space where the recording took place changes from a small studio to a large, rich hall.
- 2. Now, choose the *Metal tank* button, choose OK, and play the sound. This option uses a different *Reverberation mode*. Notice how its tone quality is very different very metallic.

Chorus

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.

- 1. With TUTOR1.WAV open, select the *Chorus* item from the Effects menu.
- 2. Choose the *Light* effect and play the file to hear an example of chorusing. Then compare this to the *Echo-vibrato* effect, which has a richer, fuller quality.
Flange

Use the *Flange* function to create the sweeping effects often heard in 60's guitar recordings and technosounds of today.

- 1. To hear what flanging sounds like, open TUTOR1.WAV and select *Flange* from the Effects menu.
- 2. Choose the *Slow* button and play the file. Then choose *Warble* to compare the flange effects.

Noise Gate

When recording a sound, you may also record 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.

- 1. For example, open TUTOR1.WAV. Move the cursor to a location between "Wow" and "Sound" and select *Insert Silence* from the Process menu.
- 2. In the Silence Length field, specify 1 second of silence to be inserted at the cursor.
- 3. Create a new file. In the new file, choose *Synthesis* from the Tools menu to generate four seconds of noise at -60 dB amplitude.
- 4. Choose *Noise* from the Waveform shape list, enter 2000 in the *Frequency* field, and drag the *Amplitude* slider until the value is approximately -60.
- 5. Copy all of the noise on to the clipboard and mix it to the beginning of TUTOR1. To mix, choose *Paste Special* from the Edit menu and select the *Mix* option. 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 choose *Statistics* from the Tools menu. Take note of the *Maximum sample value* percentage. If you followed the preceding steps, it should be close to -60 dB.
- 7. Select *Noise Gate* from the Effects menu. In the Noise Gate dialog, 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.
- 8. 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

Compressing a sound lowers its dynamic range. 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.

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

- 1. Open the file TUTMUSIC.WAV. From the File menu, choose Save As, and then save the file with a new name, such as *Music2*.
- 2. With the new file active, select *Dynamics* from the Effects menu.
- 3. Drag the *Threshold level* control until you see the value -18 dB. Drag the *Ratio* control until the compression ratio is 2:1. Select *OK* to see the results of compression in the sound. If you can see both files at once and are zoomed out, you'll notice how the levels of the new file are more constant than the original. Listening should reveal the same thing.
- 4. In this example, 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.
- 5. 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.

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

Changing Time Duration

Another Sound Forge XP 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.

- 1. Open SAXRIFF.WAV and select *Time Compress/Expand* from the Process menu.
- 2. This process is very simple to use. In the dialog, you can specify the final duration of a sound, from 50% to 500% of the original length. You can select a mode of time compression to match the type of material you're processing. The *Solo instruments mode* does a good job with this sound file. Select it from the *Mode* list. 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 *Time Compress/Expand* dialog offers a number of compression modes for various kinds of source material. 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* and *Solo Instrument* modes. For complex music, you must sacrifice some time localization for the ability to reproduce long bass notes.

If you have a drum recording with bass drum, snare, and cymbals, you will find the *Drums, unpitched (minimum echo)* does an impeccable job. However, once you start adding ringing low toms and congas, the *Drums (better for toms)* mode will work better. Again, experimentation is the key.

Synthesis

Basic synthesis

Use the Synthesis tool to generate a simple waveform of a given shape, pitch, and length.

- 1. 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*.
- 2. Select *Synthesis* from the Tools menu.
- 3. Select the *Sine* option from the *Waveform shape* list. Enter 3 in the *Length* field and 261.52 in the *Frequency* field. Choose OK to generate a sine wave in the window that is 3 second long and has a pitch of 261.52 Hz (Middle C). Press the *Play* button to hear your reference tone.

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.

- 1. To create these tones, choose *DTMF/MF Tones* from the Tools menu.
- 2. Enter a phone number in the *Dial string* field. 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 would dial the number you specified.

Filtering

<u>Filtering Intro</u> <u>Graphic EQ</u>

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.

Graphic Equalizer

The *Graphic Equalizer* 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.

- 1. Open TUTMUSIC.WAV and select *Graphic EQ* from the Process menu.
- 2. To remove some of the high frequencies from the music, drag the 1K fader to the bottom. Repeat the process for all the faders to the right of 1K. Click *OK* to remove the high frequencies from the music file. Press the *Play* button to hear how the music is much duller.

The Regions List

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

<u>Uses for the Regions List</u> <u>Creating Regions</u> <u>Editing Regions</u> <u>Using Markers</u>

Why use the Regions List?

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.

Creating Regions

Creating a Region Using Menu Commands

As with most features in Sound Forge XP, 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. 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).

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
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. Also, right-clicking on the *Regions List* allows you to choose from various commands in the *Regions List* 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 XP will automatically create a marker at the cursor position. If you have a selection, Sound Forge XP 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 XP 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.

Optimizing Sound Forge XP

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

Hard-Drive Use Editing in Direct Mode Fast File Saving Playback Cursor and Record Counter Meters Passive Updating for Video and Time Displays Audio and Video Synchronization Setting up Wave Devices Background Processing Crash Recovery

Hard Drive Use

Sound Forge XP 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 XP to edit sound files of almost unlimited size. Sound Forge XP 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 XP is hard-drive intensive, the faster your disk access, the faster Sound Forge XP 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 discontiguous. When files are discontiguous, 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.

Windows 95 Default Configuration

Choosing system configuration settings other than the default Windows 95 settings can result in reduced performance from Sound Forge XP, 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 32bit. 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.

Editing in Direct Mode

Whenever you open a file in Sound Forge XP, 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 XP 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 XP will scan the file to create a peak file. This allows Sound Forge XP 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 XP 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 XP. This allows Sound Forge XP 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 XP terminates improperly, the file would remain in its edited state.

Fast File Saving

When creating a new file Sound Forge XP creates a temporary file that contains all of your new sound data. When you are ready to save the file, Sound Forge XP 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 XP 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 *Record Meters* in Sound Forge XP take a small amount of processing overhead during recording. If you hear gaps in the sound while recording 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 lowers 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 forces 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 XP 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 XP 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:

- Ŷ 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.
- Ÿ Features like *Insert Silence* 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 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.

Setting up Wave Devices

The Sound Mapper/Wave Mapper

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

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

You may have noticed that Sound Forge XP 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 XP is working. Using other applications while Sound Forge XP is working can cause the processing time to increase substantially.

Crash Recovery

If for some reason Sound Forge XP is terminated improperly – when Sound Forge XP 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 XP.

Sound Forge XP 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 XP. The following topics are covered:

Introduction to Object Linking and Embedding (OLE) Using OLE with Sound Forge XP 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 XP. 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 XP, you can *embed* digital audio files used by Sound Forge XP 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 XP at any time by activating the embedded object (usually by double clicking on it with the mouse).

Using OLE with Sound Forge XP

Sound Forge XP is an OLE *server*. What this means is Sound Forge XP 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 XP are called *Sound Forge XP 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 XP 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 XP 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 XP to an OLE container.
- 2.Using the Copy Object Link command in the Edit menu of Sound Forge XP 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 XP Audio object from the list.

Drag and Drop

The most convenient method of embedding a Sound Forge XP Audio object into a document is to Drag a link from the OLE Drag Source control in Sound Forge XP 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 XP 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 XP 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 XP and open a sound file (TUTOR1.WAV in your Sound Forge XP directory will suffice).
- 3.Ensure that at least a portion of WordPad or Word 7.0 is visible while Sound Forge XP 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 XP 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 XP.

Copy Object Link

Another method for creating a Sound Forge XP Audio object is to invoke the Copy Object Link command under the Edit menu in Sound Forge XP. 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 XP Audio object into WordPad (Windows 95) or Write (Windows NT) using the Copy Object Link command:

- 1.Run Sound Forge XP and open a sound file (TUTOR1.WAV in your Sound Forge XP directory will suffice).
- 2.Select the Copy Object Link command under the Edit menu of Sound Forge XP. If the sound file needs to be saved, you will be prompted to do so.
- 3.At this point, a Sound Forge XP 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 XP Audio object will now be embedded in your document.

Insert Object

The last method for creating a Sound Forge XP Audio object is to use the Insert Object command from another application. This method provides a convenient way to get a new Sound Forge XP Audio object embedded into a document without having to first create it in Sound Forge XP and copy it into the document.

Follow these steps to embed a Sound Forge XP 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 XP 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 XP 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.
- After creating any desired sound data, simply save the modifications and close Sound Forge XP. If you did everything correctly, a Sound Forge XP Audio object will now be embedded in your document.

After the Sound Forge XP Audio object has been created and embedded into the document, you can edit it at any time using Sound Forge XP by double clicking on the object in your document.

Tips, Limitations, etc.

As stated earlier, Sound Forge XP 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 XP.

Links can be broken, but not mended

The most important rule to bare in mind when using OLE with Sound Forge XP is that links can be broken, but not mended. When you create a Sound Forge XP Audio object, it stores only a link (which is simply a directory path) to the sound file. Sound Forge XP 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 XP uses Object Links, moving, renaming or deleting the sound file associated with a Sound Forge XP Audio object will break the link. Attempting to edit the object in your document will fail because Sound Forge XP will not be able to locate the correct sound file.

To fix a broken link, simply delete the Sound Forge XP 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 XP 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 XP 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 XP 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 XP 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 XP 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 XP 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 XP using UNC path names rather than a drive letter. You can then create the Sound Forge XP 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 XP is not an OLE Container

Although Sound Forge XP 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 XP.

File Menu

New Open Close Save Save As Save All 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 XP 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 XP desktop to create a new file from the selection. Selecting a portion of the Sound Forge XP desktop will create a new file with the most recently used attributes.

See also

<u>File Menu</u> <u>Creating a New Window</u> <u>Drag and Drop Operations</u>

Open

Use this command to load a sound file into Sound Forge XP in Windows 95. If you are running Windows 3.x, go <u>here</u>.

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 XP will try to auto-detect the file format.

If you know that the file format is unsupported, select *Raw File* (*.*). Choose *OK* to see 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 *Summary* information for the file, but these changes must be saved to a new file.

When you open a sound file, Sound Forge XP 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 XP terminated improperly the file would remain in its edited state.

Whenever you open a file in Sound Forge XP, 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 XP 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 XP.

Play

Auditions the sound file without opening it in Sound Forge XP. If playing a file from a slow network or CD-ROM, some skipping might occur. This feature is only available for .WAV files.

More

Select a file and then choose the More button to see the general properties of the file.

Shortcut

Control+O

Control+F12

Control+Alt+F2

See also

<u>File Menu</u> <u>Opening an Existing File</u>

Open (Windows 3.x)

Use this command to load a sound file into Sound Forge XP in Windows 3.x. If you are running Windows 95, go <u>here</u>.

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 XP 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 *Summary* information for the file, but these changes must be saved to a new file.

When you open a sound file, Sound Forge XP 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 XP terminated improperly the file would be remain in its edited state.

Whenever you open a file in Sound Forge XP, 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 XP 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 XP.

Play

Auditions the sound file without opening it in Sound Forge XP. 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

<u>File Menu</u> <u>Opening an Existing File</u> <u>Installing the ACM</u>

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 XP.

Sample Rate

(2,000-96,000 Hz)

Sample playback rate which will be used by Sound Forge XP 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.

(Unsigned, signed, sign bit)

Shows the sample format that the data is stored as. Samples are most often saved in the PCM format.

Channels

Format

(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 XP. 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

<u>File Menu</u> Saving a File

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.

Shortcuts

Control+S Shift+F12 Alt+Shift+F2

See also

<u>File Menu</u> <u>Saving A File</u> <u>Save As</u> <u>Sound File Formats</u> <u>Embedded Summary Information</u> <u>RealMedia Save Options</u> <u>ASF Save Options</u>

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.

Save in

Choose the appropriate drive and folder for the file you want to save.

File name

Type the name of the file in which you want to save the current sample.

Save as type

Select the type of file to which you would like to save, such as Wave (Microsoft *.wav).

MRU Folders

Lists the most recently used directories. Use to speed the opening of regularly used files in different folders.

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.

Save Regions and Markers in .WAV files

Check this box to save any regions and markers in the .WAV file.

Save Summary Information in file

Check if you want Summary information (such as author and copyright) stored 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

<u>File Menu</u> <u>Saving A File</u> <u>Converting Files</u> <u>Embedding Text in WAV Files</u> <u>Sound File Formats</u> <u>Embedded Summary Information</u> <u>RealMedia Save Options</u> <u>ASF Save Options</u>

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 XP.

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.

(Unsigned, Signed, Sign bit)

Sample Type

(8-Bit PCM, 16-Bit PCM, G.711 u-Law, G.711 A-Law)

Format used to store each sample.

Format

Binary format of each sample.

(Mono/Stereo)

Number of channels, or tracks, stored in the file.

Byte Order

Channels

(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 XP 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 takes 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.

RealMedia Save Options

This dialog is reached when you select *RealMedia* (*.*rm*;*.*ra*) from the *Save as type* list in the *Save as* dialog. This feature is only available for Windows 95 and NT running on x86 processors.

Enable Selective Record Capabilities

When checked, RealMedia-compatible players can allow users to save (record) RealMedia streams to their hard drives. Use this option only if it is acceptable for users to create copies of the RealMedia file. The RealPlayer Plus from Progressive Networks supports Selective Record capabilities.

Enable PerfectPlay Capabilities

When this option is enabled, users connected with standard modems can select to download a higher quality version or the audio and/or video that is normally only available over ISDN or LAN connections. The higher bandwidth RealMedia stream is then played back from the user's hard disk after the download process is complete. The RealPlayer Plus from Progressive Networks supports PerfectPlay capabilities.

Create Event File (RAE) from Marker and Region Labels

When this option is enabled, events can be embedded from the *Regions List* (View menu) into the RealMedia stream.

To embed an event, create a marker or region with a label starting with one of the below prefixes followed by a colon.

•RM URL: Specifies a URL (web address) to open.

•RM Title: Sets the Title field in the RealPlayer.

•RM Author: Sets the *Author* field in the RealPlayer

•RM Copyright: Sets the Copyright field in the RealPlayer

For example, if you create a marker with the label RM URL:http://www.sonicfoundry.com, the RealPlayer would open the Sonic Foundry home page at the marker's location in the file during playback. You can just as easily open specific web pages. For example, RM

URL:http://www.sonicfoundry.com/realaudio.html would open the realaudio.html page on Sonic Foundry's web site.

Progressive Networks publishes extensive information on RealMedia events on their web site (www.real.com). However, Sound Forge XP is capable of generating all event files automatically rather than requiring several tools as described in Progressive Networks' documentation.

Save Audio Stream

When checked, the file's audio stream will be converted and saved into the RealMedia file.

Audio Compression

This option selects the RealAudio compressor that is used when saving the file. If you need to save RealAudio files with Bandwidth Negotiation capabilities, you can automate the process using Sonic Foundry's Batch Converter Plug-In (available for Sound Forge 4.0 only).

When a compression type is selected, a brief description of it is displayed below the list box. The frequency response parameter of the description indicates how wide, in Hertz, the bandwidth of the audio in the RealMedia file will be. The bandwidth of a normal human's hearing is about 20,000 Hz or 20 kHz.

Stereo to Mono Conversion

If the source audio file is stereo and you choose to convert to a mono RealAudio format, you will need to decide what channel(s) to use for the mono signal. Selecting *Both channels* will evenly mix the signal together. If you are unsure of this option, always choose *Both Channels*.

Save Video Stream

When checked, the file's video stream will be converted and saved into the RealMedia file.

Video Compression

Currently there are two types of RealVideo compression: RealVideo (Standard) and RealVideo (Fractal). Each type is optimized for different types of video and transfer rates. Detailed information on

the two types is available on the RealMedia home page (www.real.com).

Video Compression Quality (1 to 100)

This option allows you to compromise between quality of compression and lowest possible data transfer rate. Setting this level at 100 provides the best quality video with the greatest required bandwidth to stream. A setting of 1 gives you the lowest quality video but requires the least bandwidth to stream effectively.

Frames per second (0.050 to 1000)

This setting allows you to change the frame rate of the video that is being compressed. A lower frame rate, such as three or five frames per second, will greatly lower the bandwidth requirements for streaming the video. Just type the desired frames per second into the edit box, or choose one of the presets from the drop-down list.

Desired maximum data rate (1.0 to 500.0 Kbps)

This setting is available only if you are saving a video stream. The data rate specifies the desired <u>total</u> kilobits per second (Kbps) for the RealMedia file (including audio and video). To increase the bandwidth available for streaming the video, you can either lower the data rate for the audio, or increase the desired maximum data rate. Note that you must allow some room for inconsistent transfer rates. For example, a 28.8 modem cannot sustain 28.8 Kbps. Instead, you should target your maximum data rate for between 20.0 and 22.0 Kbps. The allowed data rates are displayed next to the *Audio/Video compression* options. After the file has been saved, statistics about the actual data rate will be displayed. Use this information to adjust parameters as necessary.

See also

<u>File Menu</u> <u>Using Sound Forge XP to create RealAudio files</u>

ASF Save Options

This dialog is reached when you select *Active Streaming Format (.asf)* from the *Save as type* list in the *Save As* dialog. This feature is only available for Windows 95 and NT platforms.

Maximum Lead Time (milliseconds)

Defines the lead time, if any, that the NetShow On-Demand player should buffer before starting playback. The default is 1,000 milliseconds. The lead time can affect the bit-rate that is calculated for the ASF file. Some video codecs may require that the lead time setting be increased to achieve the desired low bit-rate for the ASF stream. The VDOnet codec, for example, works best with a setting of 5000.

Marker and Region Labels

Specifies how Marker and Region labels should be used when saving the ASF file. Not only can markers specify locations in the ASF file for seeking purposes, but the labels can also identify commands for the NetShow Player.

ASF <scriptcmd>:</scriptcmd>	Specifies a user-defined script command event
ASF URL:	Specifies an URL (web address) to open
ASF Filename:	Specifies an .ASF file to chain to at the specified point
ASF Title:	Sets the Title field in the NetShow On-Demand player
ASF Author:	Sets the Author field in the NetShow On-Demand player
ASF Copyright:	Sets the Copyright field in the NetShow On-Demand player

For example, if you create a marker with the label ASF URL:http://www.sfoundry.com, the NetShow On-Demand player would open the Sonic Foundry home page during playback at the marker's location in the file. You can just as easily specify web pages, images, etc. For example, ASF URL:http://www.sfoundry.com/netshow.html would open the netshow.html page on Sonic Foundry's web site.

In addition, the example ASF CAPTION: What a fine day specifies a user defined script command CAPTION with the parameter "What a fine day". The NetShow On-Demand client will fire a Visual Basic event into the browser or container of the NetShow control. The event type can be processed as type "CAPTION" and the description parameter of the event would be the "What a fine day" text that could then be printed for the user, perhaps synchronized with an audio or video segment.

Microsoft publishes information about the NetShow control and ASF commands on the NetShow web site (http://www.microsoft.com/netshow). There are also extensive samples on the web site and in the SDK. However, Sound Forge XP is capable of generating all commands automatically rather than requiring several tools as described in Microsoft's documentation.

Enable Error Correction

(8 + 1 parity)

Enables error correction for low bandwidth files (not greater than 150 KB per second). The error correction information is used by the NetShow Player to recover from bad transfers that are common on low bandwidth connections (especially modems using standard phone lines). For high bandwidth content (greater than 150 KB per second), error correction is automatically disabled since the reliability of the connections required for these files is substantially better.

Enable Wavespan for Audio Only Streams

This option is used to enable a tradeoff between more efficient handling of larger audio objects and better interleaving of client processing time with smaller audio objects. For most audio codecs, this option should remain unchecked. However, some newer codecs may require this option to be checked for proper playback.

Change Video Compression Options

This option is only available when saving files that contain video. When you check this option, Sound Forge XP will display a video *Compression Options* dialog (see above section on *Compression Options*) after clicking *OK*. The *Compression Options* dialog allows you to redefine the compression algorithm (codec) used for the video when saving the ASF file.

Frames per second (0.05 to 1000)

This setting allows you to change the frame rate of the video that is being compressed. A lower frame rate, such as three or five frames per second, will greatly lower the bandwidth requirements for streaming the video. Just type the desired frames per second into the edit box, or choose one of the presets from the drop-down list.

Packet Size

(512 to 57,344 bytes)

This option is only available when saving files that contain video. If a specific packet size is desired for the ASF file, you can specify the requirements here. Changing the packet size can affect how efficiently ASF packets are transmitted and processed. Unless required, it is recommended that the default value be used.

Wavespan

(400 to 5,000 milliseconds)

This option is only available when saving files that contain video. If a specific wavespan is desired for the ASF file, you can specify the requirements here. Wavespan is the minimum length (in milliseconds) of audio objects that are manipulated by NetShow. This value is a tradeoff between more efficient handling of larger audio objects and better interleaving of client processing time with smaller audio objects. Most users will not need to change this option.

See also

<u>File Menu</u> <u>Using Sound Forge XP to Author NetShow Content</u>

Properties

The *Properties* folders list attributes and information which is embedded in the currently active sound file. The available Properties folders are:

<u>General</u> <u>Summary Info</u> <u>Format</u> <u>Video</u> <u>Display</u>

See also

<u>File Menu</u> <u>Summary Information</u>

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 .WAV, .AVI, .RA, and .ASF files.

Title, Subject, Engineer, Copyright, Comments

Commonly used fields you can guickly 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 XP 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 <u>File Menu</u> Properties

Display Properties

This folder allows you to change how the data window of the sound file appears on the Sound Forge XP 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.

Shortcut Edit Tool Soloctor Shor

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 XP.

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 XP 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 XP 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 XP. You will be asked whether you would like to save any sound files you have modified during the editing session.

See also

File Menu

Temporary File Cleanup

If for some reason Sound Forge XP is terminated improperly – when Sound Forge XP 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.

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

Edit Menu

<u>Undo</u> Redo Repeat Cut Copy Copy Object Link Paste Paste Special|Crossfade Paste Special Mix Paste Special|Overwrite Paste Special|Replicate Paste Special Paste to New Trim/Crop Clear/Delete Select All Tool|Edit Tool|Magnify Tool Pencil Go To Selection Undo All Disable Undo/Redo

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 by choosing Undo/Redo History from the Special menu.

Tips

When working on large files, it takes Sound Forge XP a longer period of time to create an undo buffer. To save time and file space, you can disable the undo by selecting the <u>Disable Undo</u> item under the **Edit** menu or from within most processing functions.

Shortcuts 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 by choosing Undo/Redo History from the Special menu.

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 <u>clipboard</u>. This command has no effect if there is no selected data. Cutting sample data replaces the previous contents of the clipboard.

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.

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* function in the Effects menu.

Shortcuts Control+X Control+Del

See also

Edit Menu Copy, Paste, Cut, and Undo

Сору

Use this command to copy selected sample data onto the <u>clipboard</u>. This command has no effect if there is no selected data.

Shortcuts

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 XP.

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, see <u>Object Linking and</u> <u>Embedding (OLE)</u>.

See also Edit Menu

Paste

Use this command to insert a copy of the <u>clipboard</u> contents at the current insertion point. If you have selected data, the Paste command deletes it 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.

Shortcuts

Control+V Shift+Insert Drag and drop a selection to another data window + Alt.

See also

Edit Menu Copy, Paste, Cut, and Undo

Crossfade (Paste Special)

Use this command to crossfade a copy of the <u>clipboard</u> 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 apply a linear fade to the source and destination data. It is commonly used as a transition between one sound segment to another.

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.

See also

Edit Menu Drag and Drop Operations

Mix (Paste Special)

Use this command to mix a copy of the <u>clipboard</u> 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)

(-Inf. to 15 dB)

Gain applied to the source data before mixing with destination data.

Destination Volume

Gain applied to the destination data before mixing with source data.

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.

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.

Shortcuts

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
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:

- Ÿ 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.
- Ÿ 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 <u>clipboard</u> 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.

Shortcuts Control+E

Drag and drop a selection to the Sound Forge XP 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 <u>clipboard</u>.

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.

Shortcuts

Control+A Control+Numpad 5 Double clicking the left mouse button in the Waveform Display will select all data.

See also: Edit Menu

Edit Tool

Use this command to use the *Edit Tool* of Sound Forge XP. 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 XP 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 XP 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.

Q

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.

A

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.

Shortcuts

Control+G F5 *Waveform Display* shortcut menu Double-click on the selection start field in the playbar

See also

<u>Edit Menu</u> <u>Shortcut Menus</u> <u>Using Go To</u>

Selection

Use this command to change the start, end, length, or channel of the selection on the Set Selection dialog.

Selection

Provides samples 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.

Play

Choose the Play button to play the selected sample.

Play looped

Check this box to play the selected sample in a loop.

Shortcuts

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 XP does an *Undo All* operation automatically before saving.

Disable Undo/Redo

Use this command to prevent Sound Forge XP 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 XP 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

View Menu

Maximize Width Full Screen Toolbars Clipboard Contents Clipboard Play Zoom Level|Out Full Zoom Level|Window Zoom Level|Selection Zoom Time|In Full Zoom Time|Out Full Zoom Time|Out Full Zoom Time|Selection Focus to Data Window Regions List Video Preview

Maximize Width

This command stretches the active data window to fit between the extents of the Sound Forge XP window.

Shortcuts

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 XP 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 XP workspace.

Clipboard Contents

Use this command to display information on the current contents of the <u>clipboard</u>. 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 <u>clipboard</u>. 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|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.

Shortcuts

Control+Up arrow (when a selection is active) *Level Ruler* shortcut menu Double-click on the *Level Ruler*

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 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.

Shortcuts

Control+Up arrow (when no selection is active) Waveform Display Shortcut menu

See also <u>View Menu</u>

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.

Shortcuts 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 <u>View Menu</u>

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.

Shortcuts

Control+Up arrow (when a selection is active) *Waveform Display* shortcut menu

See also

<u>View Menu</u> 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 <u>Window Menu</u>

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 <u>View Menu</u> <u>The Regions List</u>

Video Preview

Use this to turn on or off the Video Preview Window.

Shortcut

Alt+6 to turn on or set focus.

See also

View Menu

Effects Menu

Chorus Delay/Echo Distortion Dynamics Flange Noise Gate Pitch Bend Reverb

Chorus

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.

Choose the level of chorus effect you want to use, from light (the lowest amount of chorus effect) to echovibrato (the highest amount of chorus effect).

See also <u>Effects Menu</u> <u>Sound Processing and Other Tools</u> <u>Using Chorus</u>

Delay/Echo

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

Distortion

The *Distortion* effect can simulate the overloading of an amplifier or mangle a waveform into submission. Choose the level of distortion you want to use, ranging from fuzz to hard clip.

See also

Effects Menu Sound Processing and Other Tools

Dynamics

Compression affects 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.

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.

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 *Dynamics* functions. This makes it easier to monitor audible frequencies while compressing.

See also

Effects Menu Sound Processing and Other Tools Using Compression

Flange

The flanging or wah-wah 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.

Choose the type of flange effect you want to use, ranging from slow to warble.

Flange effects are best heard with long, sustaining sounds so that the entire sweep can be heard.

See also <u>Effects Menu</u> <u>Sound Processing and Other Tools</u> <u>Using Flange</u>

Noise Gate

A noise gate removes signals below a set threshold. It is used to remove noise from silent breaks in a sample.

Threshold

(-Inf. to 0 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 <u>Statistics</u> 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

Pitch Bend is used to draw an envelope which corresponds to increasing or decreasing the pitch of a sample 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.

Pitch Bend is accomplished in the same manner as changing the playback speed on a tape deck. Therefore, the length of the file will be changed.

See also

Effects Menu Sound Processing and Other Tools

Reverb

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.

Choose the amount of reverberation you want to use, ranging from that appropriate for a medium room (least amount of reverberation) to a metal tank (highest amount of reverberation).

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
<u>Effects Menu</u>
<u>Sound Processing and Other Tools</u>
<u>Using Reverb</u>

Process Menu

Convert to 8-Bit DC Offset Fade Graphic Fade In Fade Out Graphic EQ **Insert Silence** Invert/Flip Mute Normalize <u>Pan</u> Pan Left to Right Pan Right to Left Resample Reverse Smooth/Enhance Swap Channels Time Compress/Expand Volume

Convert to 8 Bit

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

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

Fade Graphic

Use a fade to vary the volume of a sound file over time. 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.

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.

Reset Envelope

Press this button to return the fade envelope to its default shape

In

Use this command to linearly fade a selection from a volume of 0 dB to a volume of -Inf

Out

Use this command to linearly fade a selection from a volume of -Inf. to a volume of 0 dB.

See also

Fade In

Use this command to linearly fade a selection from a volume of 0 dB to a volume of -Inf

Fade Out

Use this command to linearly fade a selection from a volume of -Inf. to a volume of 0 dB.

Graphic EQ

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.

Reset Bands

Press this button to return the band faders to their default settings.

See also

Process Menu Sound Processing and Other Tools Using Filtering

Insert Silence

Use this command to insert a section of silence into a sample.

Insert at Start of File Inserts the silence at the beginning of the file. End of File

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 <u>Delay/Echo</u> function to delay one channel forwards or backwards in time.

See also

Invert/Flip

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

Mute

Use this command to set a selection to a volume of 0 (silence).

See also

Normalize

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 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.

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

Pan

Choose a Pan option to have Sound Forge XP automatically pan left to right or right to left.

Choose Graphic to draw a dynamic pan 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.

Output Gain

The gain applied to the signal after processing.

Show Wave

Check this 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.

Reset Envelope

Press this button to return the pan envelope to its default shape.

Left to Right

Select this option to create a smooth panning effect from the left channel to the right. If the original source data is mono, Sound Forge XP automatically converts it to stereo.

Right to Left

Select this option to create a smooth panning effect from the right channel to the left. If the original source data is mono, Sound Forge XP automatically converts it to stereo.

See also
<u>Process Menu</u>
Sound Processing and Other Tools

Pan Left to Right

Select this option to create a smooth panning effect from the left channel to the right. If the original source data is mono, Sound Forge XP automatically converts it to stereo.

Pan Right to Left

Select this option to create a smooth panning effect from the right channel to the left. If the original source data is mono, Sound Forge XP automatically converts it to stereo.

Resample

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.

Interpolation Accuracy

(1 - 4)

The Interpolation Accuracy parameter determines the complexity of the method used during the resampling process. The audible difference between the different values can be subtle without the use of test tones and is most apparent in high frequencies.

In general, a setting of 1 is more than adequate for general purpose audio. Settings of 2 and 3 are good for high-end audio. Setting this parameter to 4 forces some heavy duty number crunching (i.e. slow processing) but produces very near perfect results, good for audiophiles and audio researchers.

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
<u>Process Menu</u>
Sound Processing and Other Tools

Reverse

Use this command to reverse a selection.

Smooth/Enhance

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

Swap Channels

Select this option to exchange the contents of the right and left channels of a stereo recording.

Time Compress/Expand

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, such as music and speech, but do not assume that one mode will not work well with a certain type of sound. Depending on the source sound, different modes can vary the resulting quality drastically, so experimenting with all modes is suggested.

Check the bottom of the dialog box for a summary of your choices. 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, multiinstrument 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

(hr:mn:sc:fr)

Desired length of the selection.

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

<u>Process Menu</u> <u>Sound Processing and Other Tools</u> <u>Changing Pitch and Time Duration</u>

Volume

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

Special Menu

Transport Regions List Edit Frame Rate Edit Tempo Center Cursor Drop Marker Create Region Mark In Mark Out Toggle Selection Undo/Redo History Clear Undo/Redo Clear All Rebuild Peak Data

Transport

Record Play All Play Pause Stop Go to Start Rewind Forward Go to End Play Normal Mode Play Looped Mode

See also

Record

Use this function to record to an existing or new data window. If you have a file open in direct mode, Sound Forge XP 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.

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 XP 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 selecting a sample. 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 (automatically rewind)

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

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. 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.

Punch In (record a specific length)

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 XP 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 reviewing a take.

Record Meters

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) or Alt+- (Windows 3.x). Choose *Record Meters* from the pop-up menu that appears. For direct access to the *Record Meters* options, right-click anywhere in the 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.

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.

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.

See also

<u>Special Menu</u>

Using Recording

Play All

Use this function to play the entire sample file from beginning to end, regardless of cursor position or selection.

Shortcut Shift+Spacebar

See also

Play

Playback file in current playback mode (Normal or Looped).

Shortcuts Spacebar Right-click on Overview

See also

Pause

Stop playback and keep cursor at the stop point.

Shortcuts

Shift+Spacebar Enter Right-click on Overview during playback

See also

Stop

Stop playback and return the cursor to its position prior to playback.

Shortcut

Spacebar (during playback)

See also

Go to Start

Moves the cursor to the start of the file.

Shortcut Control+Home

See also

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

Play Normal Mode

Sets Playback mode to Normal Mode. When **Play** is selected while in this mode:

- \ddot{Y} If there is no selection, playback occurs from the cursor to end of file.
- \ddot{Y} 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:

- Ÿ If there is no selection, the entire sample is played in an endless loop.
- \ddot{Y} 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
Regions List

Add Delete Edit Replicate Update Clear Copy to Clipboard

See also

<u>Special Menu</u> <u>The Regions List</u>

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.

Туре

(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.

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 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

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

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

Edit Frame Rate

Use this command to change the *Frames per second* value used by Sound Forge XP 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:

Custom

Choose Custom and then enter the frames per second you want to use.

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 XP uses the current tempo to display measures in the *Time Ruler* tool.

Туре

(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 XP calculates this tempo value depending on your selection length, selection type (*Measures or Beats*), and Beats per Measure.

Play

Choose this button to play the file with the current tempo settings.

Note: The default tempo (Default: 120) can be changed on the *Status* page of the *Preferences* 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.

Shortcuts

. (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 <u>Wave folder</u> 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 Regions List

Mark In

Use this option 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 the 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 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 the 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

<u>Special Menu</u> <u>Mark In</u>

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 XP to re-scan the entire file. However, under normal circumstances you should never have to do so since Sound Forge XP 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 Minimize All Restore 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 <u>Window Menu</u>

Tile Horizontally

Use this command to arrange the data windows side by side without them overlapping.

See also

Tile Vertically

Use this command to arrange the data windows top to bottom without them overlapping.

Shortcut Shift+F4

See also

Arrange Icons

Use this command to arrange the minimized data icons in the Sound Forge XP workspace.

See also

Minimize All

Use this command to minimize all open data windows in the Sound Forge XP workspace.

See also

Restore All

Use this command to restore all minimized data windows in the Sound Forge XP workspace.

See also

Close All

Use this command to close all open data windows in the Sound Forge XP workspace.

See also

Options Menu

Status Format Annotations Video Options Scroll Playback Drag and Drop Snapping Lock Region Length Paste Events Preferences

Status Format

Use this command to set the status format. Possible formats are:

Samples Time Seconds Time & Frames Absolute Frames Measures & Beats SMPTE Non-Drop SMPTE 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

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

Seconds

Use this command to set the status format to show seconds with six decimal place precision.

See also

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

Absolute Frames

Use this command to set the status format to show frames and fractions of a frame.

See also

Measures & Beats

Use this command to set the status format to beats.

See also
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.

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 <u>SMPTE, the</u> <u>ultimate confusion</u> for more information.

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 <u>SMPTE</u>, the ultimate confusion for more information.

See also

<u>Options Menu</u> <u>Using Status Formats</u> <u>Status Formats</u>

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 <u>SMPTE, the ultimate confusion</u> for more information.

See also

<u>Options Menu</u> <u>Using Status Formats</u> <u>Status Formats</u>

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 <u>SMPTE, the ultimate confusion</u> for more information.

See also

Options Menu

Using Status Formats Status Formats

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

Annotations

The names of markers and regions 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

Annotations|Marker Names

Displays marker names on the waveform display.

See also

Annotations|Marker Lines

Displays dotted lines on the waveform display showing where markers exist.

See also

Annotations|Region Names

Displays region names on the waveform display.

See also

Annotations|Region Lines

Displays dotted lines on the waveform display showing where regions exist.

See also

Video Options

These options affect the Video Preview window.

<u>Copy Current Frame</u> <u>White, Black or Default Background</u> <u>Animate Video Strip</u> <u>Number Frames</u> <u>Stretch to Window</u> <u>Integral Stretch</u> <u>Preserve Aspect</u> <u>Passive Update</u>

Video|Copy Current Frame

Enabling this option will copy the frame corresponding to the current cursor location to the clipboard.

See also

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

Video|White Background

This option selects the color used surrounding frames in the Video Preview and video strips.

See also

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

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

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

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

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* and cursor positions.

See also

Lock Region Length

Use this to have the length of a region remain constant when changing the start or end time of a region.

Shortcut

Holding the Shift key down while dragging Ruler Tabs also locks the length of a region.

See also

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.

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:

<u>General</u> <u>Display</u> <u>Editing</u> <u>File</u> <u>Performance</u> <u>Status</u> <u>Toolbars</u> <u>Wave</u>

See also

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 XP.

Always Ask for Region and Marker 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 Drag and Drop Sound Files in Direct Mode Default: Unchecked

Any files opened by dragging them onto the Sound Forge XP 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 XP 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 XP is being launched by another application (such as a multi-track editor), when this option is turned on Sound Forge XP 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 XP 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 XP to always open in full-screen mode. If this option is disabled Sound Forge XP 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 XP to prompt you with a confirmation message box prior to exiting Sound Forge XP.

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 XP'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 XP tries to draw a waveform. If you experience a system lockup frequently in Sound Forge XP, 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 draw mode

Default: Unchecked

When this mode is enabled, Sound Forge XP 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.

Ignore fact chunk when opening compressed WAV files Default: Unchecked

Check this option if you want to ignore the fact chunk when you're opening compressed .WAV files.

Show all ACM convert formats in Save As Default: Checked

When enabled, Sound Forge will enumerate all compression formats available for each ACM compressor. However, this enumeration process can be slow on slow machines. For simple ACM codecs (Microsoft ADPCM, IMA ADPCM, etc.), this option is unnecessary (they have only one compression type per input) and could be disabled to speed up the selection of ACM formats.

Show Free Storage Space on Status Bar Default: Checked

When this option 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.

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 XP 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.

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.

Normal zoom ratio

Default: 1:4,096

Specifies the normal zoom magnification you want to use. Higher values show more data on initial file loading, which takes more time.

Default sound file window height Default: 12

Specifies the height of the sound file window, from small (1 line) to big (25 lines).

Default video strip height Default: 4

Specifies the height of the video strip, from small (1 line) to big (25 lines).

Color Preference for

The color preferences section allows you set a custom color for a variety of graphics within Sound Forge XP. 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.

From the drop-down list, choose an item, such as the play cursor, for which you want to set a color.

See also

Options Menu

Editing Preferences

Use this to set preferences related to editing and undo operations.

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 XP 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.

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.

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 Default: Unchecked

if Save Changes the Sound Data

If this option is on, when you save a file to a different format Sound Forge XP 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

Default: Unchecked

for Digidesign's Session 8 .WAV Files

This option is only necessary if you are using Sound Forge XP to modify files for use with Digidesign's Session 8. Enabling this option forces Sound Forge XP 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 Default: Unchecked (16-bit capable)

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 XP 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

Default: Off

with Sound Forge 3.0 Compatible Regions

Sound Forge XP 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 XP 4.0 fixes this problem, but in doing so this means that file created with Sound Forge XP 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 XP 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 XP 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 XP. 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 XP. When a file is associated with Sound Forge XP, 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

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 XP.

Sound Forge XP 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 XP 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 XP 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 XP 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

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 XP 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.

Display Format for SMPTE Code Default: 00:00:00.00

Specifies how SMPTE values will be formatted for display by Sound Forge XP. Select one of the available preferences from the drop-down list.

See also

Options Menu

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

Default: Checked

Contains standard file, and edit operations.

Transport

Default: Checked

Contains record and play operations.

Status

Default: Unchecked

Contains operations to set the display Status Format as well as the Snap To operations.

Show Tooltips

Default: Checked The Show Tooltips check box turns on or off the help information box displayed next to the mouse when it is held over certain items.

See also

Options Menu

Wave Preferences

Use this folder to specify what wave devices Sound Forge XP should use for recording and playback.

Note: The capabilities of wave 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 XP should interpolate the position of the play position pointer during playback. If this option is not checked, Sound Forge XP 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 XP 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.

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

Tools Menu

DTMF/MF Tones Statistics Synthesis

DTMF/MF Tones

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)

(0.001 to 2 seconds)

(0.001 to 5 seconds)

Output length of individual tones.

Break Length

Length of the silence between tones.

Pause Length

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
Statistics

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 XP 4.0 is slightly different than in Sound Forge XP 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

Synthesis

Use Synthesis to generate a simple waveform of a given shape, pitch, and length.

Amplitude

Peak level of the waveform to be generated. When generating *Noise*, the amplitude is affected by the cutoff frequency selected.

(-Inf. to 0 dB)

Waveform Shape (Sine, Square, Saw, Triangle, Noise, Absolute Sine) Shape of a single period of the waveform.

Length (0

(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

<u>Tools Menu</u> <u>Sound Processing and Other Tools</u> <u>Using Synthesis</u>

Appendix

Sound Formats Summary Information Fields Sound Forge XP and the Audio Compression Manager Installing the Audio Compression Manager SMPTE Time Code Windows 3.1 and Windows 95 specific info Using CSOUND, MTU, IRCAM, BICSF and EBICSF Files Object Linking and Embedding (OLE) Optimizing Sound Forge XP

Sound File Formats

The file formats recognized by Sound Forge XP 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.

Active Streaming Format

This is the new Active Streaming Format (ASF) used by Microsoft's NetShow On-Demand. ASF files are designed for both audio and video streaming over the Internet.

Ad Lib Sample

(.SMP)

(.ASF)

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)

(.VOC)

This format is used with Covox software. It is an 8-bit mono uncompressed file format.

Creative Labs VOC File

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 XP will unpack prior to importing the file. The .VOC format also supports information for silence, looping, and varying sample rates. When Sound Forge XP imports a .VOC file it uses the first sample rate found and ignores such information as looping and silence. Sound Forge XP will save .VOC files as unpacked data with none of the additional information such as looping or silence. Sound Forge XP 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 XP 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. (.PAT)

Gravis Patch File d

This format is used by the Gravis UltraSound Card. Sound Forge XP 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 XP 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 XP 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 XP 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 XP reads and writes the most common data formats for these files, 8-bit linear, 16-bit linear, and u-Law.

RealMedia

(.RM/.RA)

These are compressed formats designed for real-time audio and video streaming over the Internet.

(.SMP)

SampleVision File 🙂

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

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)

(.SND)

This is a Microsoft digital video file format. It supports multiple channel audio, but Sound Forge XP 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)

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 XP 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 onedigit 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 XP."

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."

DISP Sound Scheme Title

Identifies the name of the sound scheme, such as "doorbell."

Sound Forge XP and the Microsoft Audio Compression Manager

Sound Forge XP 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 XP 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 XP to use compression and filtering algorithms provided by other companies.

Sound Forge XP fully supports audio compression and filtering through the ACM. This enables you to use any ACM compatible compression and filter driver with Sound Forge XP. The best part of this support is you don't have to learn anything new to use it! Sound Forge XP 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 XP, 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 XP inconvenient.

In Sound Forge XP, 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 XP 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* dropdown list of the *Save As* dialog. Once a file has been saved as compressed, Sound Forge XP 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 *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 XP 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 BBB's and other places for use with Sound Forge XP. If any ACM filters are installed in your computer, you will see *the ACM Filter* menu item under the Tools menu.

Sound Forge XP 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 XP 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 XP 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*.

Let's 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 don't 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 it's 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 Group's 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 XP 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 XP is the perfect tool for authoring compressed sound files for the *ACM* and many software authors are using Sound Forge XP for exactly this. Sound Forge XP 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 XP will usually sound better than those recorded with audio compression.

Notes on Sound Forge XP's sound file compression support:

Ÿ The Open dialog allows you to preview compressed .WAV files if you have an appropriate ACM driver is installed. However, you must have your *Playback* device set to the *Sound Mapper* for this to work (*Preferences*|*Wave* in the Options menu).

Ÿ 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 XP 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 you've 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 XP 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.

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 know 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 know 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 XP deals with the issue

For SMPTE 24, SMPTE 25, and SMPTE 30 Drop time codes, Sound Forge XP 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 XP 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 XP 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 xx:00:00:00	10 xx:10:00:00	20 xx:20:00:00	30 xx:30:00:00	40 xx:40:00:00	50 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

Using CSOUND, MTU, IRCAM, BICSF and EBICSF Files

Although Sound Forge XP 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 XP, 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 XP).

Opening and Saving files

BICSF and EBICSF files

When reading BICSF or EBICSF Sound Forge XP 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 XP 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 XP Raw File Type dialog box and then save presets labeled accordingly. Sound Forge XP's *Raw File* import will allows you to read these files as long as they are mono or stereo and are stored in 16-bit linear format. Sound Forge XP 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 XP, 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.
- 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 XP

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 XP. 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.

- 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? <u>ANSWER</u>
- 3. Why do some effects take so long to process a file? ANSWER
- 4. When I start Sound Forge XP my Play and/or Record buttons/options are grayed out. Why can I not play or record? <u>ANSWER</u>
- 5. Why can I not get Sound Forge XP to respond to levels from my Microphone or my CD Player? <u>ANSWER</u>
- 6. I have sound files that don't want to play on my system. Sound Forge XP loads them fine, why will it not play them? <u>ANSWER</u>
- 7. Why will Sound Forge XP not open my compressed files like Microsoft ADPCM? ANSWER
- 8. Why is it when I play my sound files the play pointer that Sound Forge XP displays seems to be off from what I am hearing? <u>ANSWER</u>
- 9. When I record or play back files I hear what sounds like small clicks in the sound. ANSWER
- 10. When I record (or play) data at high sample rates, I cannot get Sound Forge XP to stop by pressing the Stop button. <u>ANSWER</u>
- 11. 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. <u>ANSWER</u>
- 12. 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 XP? <u>ANSWER</u>
- 13. When playing Sound Files with rates above 44,100 Hz on my Turtle Beach Multisound or Tahiti, my system crashes. <u>ANSWER</u>
- 14. When playing Sound files at 8,000 Hz on my Turtle Beach Multisound, the play pointer moves but I do not hear any output. <u>ANSWER</u>
- 15. When playing Sound Files at rates other than 11, 22, or 44 kHz the files sound wrong (the pitch is wrong). <u>ANSWER</u>
- 16. Microsoft ADPCM files created using Sound Forge XP crash when I play them in Media Player on my Turtle Beach sound card. Why? <u>ANSWER</u>
- 17. When I play a sound file, I sometimes hear a click at the beginning or end of the selection. <u>ANSWER</u>
- 18. While drawing the Waveform Display, my system locks up completely. ANSWER
- 19. 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. <u>ANSWER</u>

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.

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 it's operating system to keep track of the types of files to which it has access. Sound Forge XP 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 <u>Troubleshooting</u>

Q: Why do some effects take so long to process a file?

A: Some processing functions used by Sound Forge XP are very math intensive. Examples include EQ 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 XP must then shift the rest of the file over.

See also

Q: When I start Sound Forge XP my Play and/or Record buttons/options are grayed out. Why can I not play or record?

A: When Sound Forge XP 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 XP does not recognize this, you need to check a couple of things. First check the <u>Wave folder</u> 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 XP. By default Sound Forge XP uses the Wave Mapper (may also be called the Sound Mapper) device which should automatically pick an appropriate device for use by Sound Forge XP. If either the Playback or Record device is set to (None) then Sound Forge XP 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

Q: Why can I not get Sound Forge XP 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.

Q: I have sound files that do not want to play on my system. Sound Forge XP loads them fine, why will it not play them?

A: Sound Forge XP 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.

Q: Why will Sound Forge XP not open my compressed files like Microsoft ADPCM?

A: Sound Forge XP 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 XP and The Microsoft Audio Compression Manager for more information on how to install and use the ACM with Sound Forge XP.

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 XP displays seems to be off from what I am hearing?

A: For Sound Forge XP 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 XP provides a way to compensate for this by setting various options in the <u>Wave Preferences</u> folder (File menu). Refer to the Wave Preferences folder in the Reference section for more information.

Q: When I record or play back files I hear what sounds like small clicks in the sound.

A: When Sound Forge XP 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 XP 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 XP is moving sound data, all of which are covered in the Using Sound Forge XP section called <u>Optimizing Sound Forge XP</u>.

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 XP to stop by pressing the Stop button.

- A: Sound Forge XP 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 XP 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 XP. For more information on configuring your system, see the section on <u>Optimizing Sound Forge XP</u>.

See also

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 data in the WAV file it will be interpreted as audio data.

If you are experiencing this problem simply, use the <u>Save As</u> command in Sound Forge XP to save a new WAV file with the *Save Summary Information* and *Save Regions* check boxes **unchecked**. This tells Sound Forge XP to save only the minimum amount of information needed for a WAV file. Also saving a file with *Sector aligned data for Digidesign's Session 8 (File Preferences)* turned on can cause a glitch at the start of the sound file in other software.

See also

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 XP?

- 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 XP'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

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.

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.

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.

Q: Microsoft ADPCM files created using Sound Forge XP 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 XP on these cards.

This is a bug in the Turtle Beach drivers. Contact Turtle Beach for updated drivers.

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 XP 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 (IRQ's) 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 XP, 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 XP'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
- 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

Glossary

ADPCM <u>A-Law</u> Aliasing **Amplitude Modulation** Attack Audio Compression Manager (ACM) Audio Event Locator Bit Bit Depth Byte Channel Converter <u>Chorus</u> Clipboard Clipping CODEC Crossfade Loop Cutoff-frequency Data Window DC Offset **Decibel Destructive Editing Device Driver Digital Signal Processing** Direct Edit Mode Drag and drop **Dynamic Range** Equalization (EQ) File Associations File-Format **Frequency Modulation** Frequency Spectrum **Insertion Point** MCI Media Player Noise-shaping Non-Destructive Editing Normalize Nyquist Frequency <u>Overview</u> Peak Data File Playbar **Quantization Quantization Noise Reactive Preview** Resample Root Mean Square Ruler, Level Ruler, Time Ruler Tags Sample Sampled Sound Sampler Sample Rate Sample Size

Shortcut Menu Sign Bit Signed Status Bar Tempo Waveform Waveform Display u-Law Unsigned Zero Crossing Zipper Noise

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, Microsoft's ADPCM algorithm is *not* compatible with the International Multimedia Association's (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 XP 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 XP 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 XP 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

Chorusing is an effect created by combining a signal with a modulating delayed copy of itself. This effect creates the illusion of multiple sources creating the same sound.

Clipboard

The clipboard is where sample data is saved when you cut it from a sample so that you may past it or mix it into another sample. Sound Forge XP 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 XP 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 XP 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 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 XP 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 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 XP, 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 XP 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 XP 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 XP. 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 XP 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 XP. 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 XP 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 XP, 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).

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 XP is a destructive editor.

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 XP 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.

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 XP 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.

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 XP 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 dB's.

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

The word sample is used in many different (and often confusing) ways when talking about digital sound. Here are some of the different meanings:

- X 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.
- X 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.
- Ÿ 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 XP 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 Shortcut 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 XP 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, two's 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.

Status Bar

The Status Bar is the bar at the bottom of the Sound Forge XP 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.

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 XP's fades are accomplished using 64-bit arithmetic, thereby creating no audible zipper noise.

Tips of the Day

Accurate play cursor position Attach a video to a sound file Context-sensitive help **Digidesign's Session 8 Direct file editing** Docking a Toolbar Double-clicking with markers and regions Drag and Drop Mix, Crossfade, and Paste Dropping markers Edit boxes Faders and trackbars Find the maximum level in a file Fine tuning a control Keyboard shortcuts Magnify Mode Making a selection during playback Marker Navigation Maximize the level of a sound file New file from selection **Opening files** Optimizing your system Pencil tool Play Loop mode Playing to the cursor Re-select the last selection Repeating an operation Saving a file in RealMedia format Scrolling with the Time Ruler Shortcut Menus Silky smooth fades **Spinner Controls** Time and Level Zoom **Toolbars** Turn off Tip of the Day Vertical Zoom

Turn off Tip of the Day

Next

To keep this window from popping up every time you run Sound Forge XP, go to the <u>General Preferences</u> 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 XP 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 XP 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 XP's performance are included in the Optimizing Sound Forge XP
 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.
 Direct Mode

List all Tips of the Day
Level Zoom

<u>Next</u> <u>Prev</u>

To view low-level sounds, zoom vertically by using the <u>Level Zoom</u> 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.

 Marker Navigation
 Next
 Prev

 To move the cursor to the next maker or region, press Control+Left/Right Arrow keys. If you also hold down the Shift key, a precise selection can be made.

Double-clicking with markers and regions <u>Next</u> <u>Prev</u>

Double-clicking on the Waveform Display between two markers selects the region between them. The same goes for double clicking inside regions. 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.

 Toolbars
 Next
 Prev

 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.

Shortcut Menus

<u>Next</u> <u>Prev</u>

Shortcut menus exist for almost every square inch of space in Sound Forge XP. To view a <u>shortcut menu</u>, move the cursor to any area and click the right mouse button. Press F1 while selecting any menu item for more information.

Docking a Toolbar

<u>Next</u> <u>Prev</u>

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.

Spinner Controls

<u>Next</u> <u>Prev</u>

Instead of clicking on the upper or lower buttons on a <u>spinner</u>, 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.

Edit boxesNextPrevAny 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.

Faders and trackbars

<u>Next</u> <u>Prev</u>

To fine tune a <u>fader or trackbar</u> value, as when you choose Graphic EQ from the Process menu, 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.

Keyboard shortcutsNextPrevSome of the most useful keyboard shortcutsinclude:SpacebarPlay/StopEnterPauseCCenter cursor on screen

 Opening files
 Next
 Prev

 You can quickly open files by dragging them from any folder onto the Sound Forge XP desktop. If you have Open Drag and Drop Sound files in Direct Mode selected from the General page of Preferences, the files will be opened even faster.

 Next
 Prev

Drag and Drop Mix, Crossfade, and Paste <u>Next</u> <u>Prev</u> By <u>dragging</u> 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.

New file from selectionNextPrevTo create a new copy of the current selection, simply drag it onto an open area of the Sound Forge XP
desktop.

 Making a selection during playback
 Next
 Prev

 To make a selection during playback, press the [key to set the start time and then the] key to set the end time and make the selection.

Magnify Mode

<u>Next</u> <u>Prev</u>

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, clicking toggles between Horizontal Only, Vertical Only, and Horizontal and Vertical magnify modes.

Re-select the last selectionNextPrevTo re-select the last selection, press the S key.Press again to clear the selection.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.

 Digidesign's Session 8
 Next
 Prev

 You can save files compatible with Digidesign's Session 8 by checking the Sector align data option on the File page of the Preferences properties sheet (Options menu).

 Dropping markers
 Next
 Prev

 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.

Silky smooth fadesNextPrevChoose Fade from the Process menu, and then choose the Graphic option to apply Dithering and NoiseShaping to fades to minimize 16-bit quantization noise.

Maximize the level of a sound fileNextPrevChoose Normalizefrom the Process menu 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.

 Saving a file in RealMedia format
 Next
 Prev

 The RealMedia file format allows you to broadcast audio and video in real time over the Internet. To save a file in the RealMedia format (.rm), choose Save As from the File menu, and then select RealMedia as

 the file type.

Playing to the cursorNextPrevTo play up to a cursor (with no selection), Control-click the Play Normal button.List all Tips of the Day

Pencil tool

<u>Next</u> <u>Prev</u>

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.

Repeating an operation To repeat the last operation, use Control+Y. List all Tips of the Day

<u>Prev</u> <u>Next</u>

 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.

Find the maximum level in a fileNextPrevTo find the maximum level in a file, choose theStatisticsoption from the Tools menu.List all Tips of the Day

 Fine tuning a control
 NextPrev

 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.

 Scrolling with the Time Ruler
 Next
 Prev

 To scroll the Waveform Display backwards or forwards in time, drag the Time Ruler above the waveform.
 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 on the <u>Wave</u> page of the Preferences properties sheet (Options menu). This compensates for sound cards that report position information incorrectly.

Context-sensitive help

<u>Next</u> <u>Prev</u>

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.

Play Loop mode

<u>Prev</u>

To play in Loop Mode, press the *Play Looped* button **P**. During playback, changing the cursor or selection by poking around the Waveform Display updates the play position immediately. This is very useful for tuning loops or selections and auditioning different parts of a file.