

Adobe® Audition™

User Guide



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Adobe® Audition™ User Guide for Windows®.

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Introduction

Welcome to Adobe Audition, the complete multitrack digital audio recorder, editor, and mixer for Windows 98, ME, 2000, and XP. The combination of Adobe Audition and any Windows sound card puts the power of an entire digital recording studio at your fingertips. So start making some great sounds, and thanks for using Adobe Audition.

Getting Help

Adobe provides a variety of options you can use to learn Adobe Audition, including printed guides, online Help, and tool tips. You can also access a host of continually updated Web resources for learning Adobe Audition, from tips and tutorials to tech support information.

If you ...	Try this ...
Want information about installing Adobe Audition	Follow the on-screen installation instructions. For more detailed information, see the <i>HowToInstall</i> file on the CD.
Are new to digital audio editors	<ul style="list-style-type: none"> • See the "Looking at the Work Area" chapter to get familiar with the work area and tools. • Use the Adobe Audition tool tips feature to help identify tools, buttons, and palette controls as you work in Adobe Audition.
Are looking for a detailed information about a feature	Search for the feature in Help or look it up in the index.
Want a complete list of keyboard shortcuts	See the Keyboard Shortcuts chapter.
Want answers to common troubleshooting questions	Search the Adobe Support Knowledgebase and Adobe Audition Top Issues, which you can access from Adobe's Web site at www.adobe.com/support/products/audition.html .

Registration

In order for Adobe to provide you with the highest quality software, offer technical support, and inform you about new Adobe Audition software developments, please register your application. When you first start the Adobe Audition application, you're prompted to register online. You can choose to submit the form directly or fax a printed copy. You can also register by filling out and returning the registration card included with your software package.

Customer support

When you register your product, you may be entitled to technical support. Terms may vary depending on your country of residence. For more information, refer to the technical support card provided with the Adobe Audition documentation.

Adobe also provides several forms of automated technical support:

- See the ReadMe file installed with the program for information that became available after this guide went to press.
- See the Adobe Audition support page for information on top support issues and troubleshooting information for common problems.

Adobe Audition at a Glance

Adobe Audition software is a professional audio editing and mixing environment. Designed for audio and video professionals in studios, broadcast facilities, and post-production facilities, Adobe Audition delivers advanced audio mixing, editing, mastering, and effects-processing capabilities. Mix up to 128 tracks, edit individual audio files, create loops, and use more than 45 DSP (digital signal processing) effects. Adobe Audition is a complete multitrack recording studio that offers a flexible workflow coupled with exceptional ease of use. Whether you're producing music, radio broadcasts, or audio for video, the precise tools in Adobe Audition give you the power to create rich, nuanced audio of the highest possible quality. A complete solution for digital audio, Adobe Audition lets you:

Sound your best Adobe Audition supports files with bit depths of up to 32-bit and sample rates in excess of 192 kHz, so whether your destination is tape, CD, DVD, or DVD-Audio, Adobe Audition is up to the task. All processing is done at 32-bit resolution for the highest quality sound.

Work efficiently Adobe Audition puts all the tools you need at your fingertips so that you can get your work done quickly and efficiently. An accessible interface helps you to get up and running in no time, and dockable palettes enable you to arrange your workspace to match a project's needs.

Use integrated tools Adobe Audition provides an all-in-one audio editing and mixing solution. Integrated multitrack and edit views, real-time effects, looping support, analysis tools, restoration features, and video support provide the power you need in a fully integrated audio toolset.

Sound Your Best

Adobe Audition enhances your sound with the following advanced audio features:

Highest-quality audio Record, edit, and mix high-resolution 32-bit files using any sample rate up to 10 MHz, including standard rates such as 44.1kHz, 88.2kHz, 96kHz, and 192kHz. Support for 24-bit/96kHz recording ensures DVD-ready sound.

Sample-accurate editing Be as precise in your cuts as you like. Edits can be accurate down to the sample level and can be automatically snapped to zero crossings; short crossfades can be added for smooth, pop-free cuts every time and quick fades can be added using keyboard shortcuts. Check for in-phase material and mono compatibility in the Phase Analysis window, and use the Frequency Analysis window to take snapshots of frequency (FFT) data and export that information to the clipboard.

Powerful DSP tools and effects Work with more than 45 DSP tools and effects, mastering and analysis tools, and audio restoration features. Adobe Audition includes standard effects such as Reverb, Delay, Dynamics Processing, and Flange, as well as advanced processing such as Parametric Equalizer, Scientific Filters, Doppler Shifter, Hard Limiting, Noise Reduction (including click and pop elimination), and Time/Pitch stretch. Adobe Audition also supports third-party DirectX plug-ins.

Precise sample rate conversion Convert audio sample rates with confidence: transparent sample-rate conversion guarantees that if you need to change the sample rate of audio, it will sound its best. Ideal for taking CD material at 44.1kHz and upsampling it to 48kHz for video or 96kHz for DVD.

Automation envelopes Create smooth volume, pan, effects parameter, and wet/dry transitions for clips in a multitrack mix using sample-accurate automation envelopes or “rubber bands.” Envelope curves can be adjusted using splines to produce gradual, non-linear transitions.

Customizable dithering Convert high-bit-rate material to lower resolution formats without introducing audible artifacts. User-definable dither depth and a wide array of Noise Shaping curves assure that your audio will retain its character and definition.

High-quality stretching Change tempo without shifting pitch or shift pitch without changing tempo thanks to high-quality stretch support.

Work Efficiently

Adobe Audition increases your productivity with extensive customization and performance options:

Customizable workspace Get up and running quickly using an interface that is dockable, resizable, and can be configured to meet your production needs. For example, create default sessions that can be called up as templates for projects that share similar configurations.

Real-time effects Process effects in real-time instead of waiting for each effect to render before moving on. To reduce CPU usage, lock a track to cache the rendered effect; unlock it instantly to make adjustments.

Integration with Adobe Premiere® Pro and Adobe After Effects® Send your audio from Adobe Premiere Pro or After Effects into Audition with the Edit Original command. WAV files are stamped with session data, so Audition can automatically open the session that created the WAV file you are using in the other software.

Organizer window Keep your project resources at your fingertips using the Organizer window, where you can quickly access currently open audio, MIDI, and video files; available effects; and favorites.

Batch file processing Use the dedicated batch-processing wizard to easily convert large numbers of files among different sample rates, bit depths, and file formats. Also, automate repetitive tasks with scripts.

Surround sound support Use the multichannel encoder to create a surround mix from any session in the multitrack view, then export the mix as a 6-channel wave, six mono waves, or a multichannel WMA 9 file.

Favorites Use Favorites to build a customized collection of commands, scripts, effects, and external software tools. Favorites appear in the Organizer window and the Favorites menu, and can be assigned keyboard shortcuts for quick access.

Precise analysis tools Work with built-in analysis tools (frequency analysis, phase analysis, statistics, and spectral view) in real-time while editing to save time and see your audio characteristics from edit to edit.

Use Integrated Tools

Adobe Audition integrates many powerful digital audio tools:

Complete toolkit for digital audio Handle all of your digital audio editing tasks within a single application: Audition includes an integrated multitrack mixing view, a mono and stereo waveform editing view, effects support, and looping capabilities.

Powerful multitrack editing Use the multitrack view for recording, mixing, and editing sessions with up to 128 stereo tracks. Apply volume, pan, and effects automation envelopes, use real-time effects and EQ on every track, play back AVI and MIDI files, and more.

Audio for video Open AVI files in the multitrack view and use all of the audio tools available in Adobe Audition on the soundtrack. This means you can do anything from creating a totally new soundtrack, to sweetening the recording you already have, to reducing noise on the current soundtrack.

Loop-based soundtrack creation Use flexible looping tools to quickly construct high-quality music for songs or movie soundtracks. Loops automatically match global session tempo and key to produce dynamic musical performances.

Thousands of royalty-free production music loops Adobe Audition includes thousands of original, performance-based loops in a wide range of musical genres that you can combine to create your own music beds, songs, soundtracks, and more. Styles include Classical and Orchestral, Blues, Cocktail Jazz, Bossa Nova, Rockabilly, 70s Funk, Lounge, Dub Reggae, Ska, Alternative Rock, Techno, Ambient, and World Music.

Broad file-format support Work with files that arrive from a wide array of sources and deliver files in the audio format you need: Audition supports more than twenty file formats and variations, including Windows PCM (wav), AIFF, MP3, mp3PRO, and WMA 9.

Chapter 1: Key Concepts of Adobe Audition

To get the most out of Adobe Audition, you should be familiar with some of its general operating concepts. The following information should help you to understand Adobe Audition’s working model in a few key areas.

Destructive and Non-Destructive Editing

Being a software-based audio system, Adobe Audition deals with audio in a digital form, meaning that an analog waveform (such as your voice saying “hello”) is converted, or sampled, into a binary representation. This sampled waveform then exists as a waveform on your computer’s hard drive. When you put together a song or audio presentation in Adobe Audition, waveforms are your building blocks, much as text files are your building blocks when you use a word processor to compose a book.

One key point is the way in which an audio program operates on a waveform. There are two kinds of computer-based audio editing: destructive and non-destructive.

- Destructive editing simply means that edits (cutting, pasting, adding reverb, etc.) are applied directly to the waveform, so that the original audio data changes.
- Non-destructive editing implies that the file on disk is not actually altered; instead, each “edit” is stored as an instruction to be applied to the file. For example, in destructive editing, a change in audio volume would actually alter the amplitude of a waveform, whereas in non-destructive editing, the volume change would consist of commands that essentially instruct the program to “get louder at this point”.

Adobe Audition employs both destructive and non-destructive editing methods. However, in Adobe Audition, even “destructive” edits aren’t applied directly to the waveform in use until you save the file. In essence, Adobe Audition uses “delayed destructive editing”.

When you open a waveform in Adobe Audition, it places a copy of the file in a temporary folder, and it uses that copy for editing. The original file remains unaltered until you choose to save any changes you’ve made (by choosing Save from the File menu, for example). When you save your changes, Adobe Audition overwrites the original file with what you’ve done to the copy. Therefore, destructive edits in Adobe Audition don’t directly affect your original waveform when you perform them.

- Edits performed in Adobe Audition’s Edit View like cutting, pasting, and applying effects (such as Reverb or Limiting) are destructive in nature, so when you save your file again, those edits are applied to the waveform itself.
- Edits made in the Multitrack View, such as a cut, split, or volume change, are non-destructive. The use of real-time effects in Multitrack View is non-destructive as well.

This same model of delayed destructive editing also enables Adobe Audition to offer multiple levels of Undo. When you apply destructive edits to a waveform, Adobe Audition retains a copy of the file as it exists before the edit. It does this for each edit you perform, enabling you to travel back through your edits to previous states of the waveform.

Multiple Undo gives you tremendous freedom in working on waveforms without having to worry about destructiveness in the least. However, it also requires additional hard drive space to store these copies. (Copies are automatically deleted from your hard drive when you close a file or exit the program.) You can disable the Undo feature if your disk space runs too low.

Real-Time Preview

In the Edit View, Adobe Audition offers real-time preview for many of its effects. This means that you can monitor the processed signal before applying the effect to the waveform. (Remember: the effect is actually applied to a copy of the waveform, and can be freely undone.) The preview feature updates in real time, meaning that changes you make to effect parameters while in the dialog for that effect become audible immediately, while the audio is playing. Keep in mind, however, that your system's performance affects the preview feature. On slower systems, some effects may tend to break up or skip during preview. In multitrack the preview is not necessary, as effects are used non-destructively. Basically, every effect in the Multitrack View is in preview all the time.

Multitrack

Adobe Audition's multitrack environment enables you to place multiple waveforms, MIDI files, and video soundtracks into different tracks for simultaneous playback and mixdown. No matter which type of audio you use, each sound instance in Adobe Audition's Multitrack View is referred to as a "block."

The mixing process involves combining the audio of all of the placed blocks into two or more channels for output. Because you can edit, add, and subtract blocks in the multitrack environment, Adobe Audition must constantly watch for changes to the multitrack session, such as a moved or deleted sound file, a volume change, or new material recorded into a track. When something is altered, Adobe Audition must immediately work that change into the mixed output, and it does so through background mixing. Background mixing occurs behind the scenes and is generally very fast. The faster your system – especially your CPU and hard drive(s) – the faster Adobe Audition can mix in the background.

The Mix Gauge at the bottom of the Track Controls in the Multitrack View shows how far along Adobe Audition is in the process of mixing your session. The meter goes from "empty" to "full" as it mixes the session, and it turns a bright color when the entire session is fully mixed. You can think of this meter as a "ready" indicator. However, you need not wait for the meter to reach completion entirely before beginning playback. As stated above, Adobe Audition continuously mixes, and will continue to do so while playing, so you can safely begin playback when the Mix Gauge meter is about halfway up.

In general, if the background mix is not sufficiently complete, you'll hear a break-up or skipping when playing back the mix. If that happens, just wait a few seconds – Adobe Audition will usually catch up very quickly, and you can begin playing again. Your audio might also start to break up if you are using real-time effects in multitrack view and your system cannot keep up with the processing. If this happens, try using the Lock feature on some of your effected tracks to mix the effects into the background mix and take some load off of your CPU.

Playback can be directed to a pair of outputs (as in a single stereo sound card) or to multiple outputs (as in multiple stereo sound cards, or a single card with multiple outputs). Adobe Audition produces a mix for each set of outputs used. If you're using one stereo sound card, Adobe Audition generates just one mix, but if you have multiple outputs, it must create a separate mix for each output device (typically a stereo pair). The additional mixing required for multiple outputs demands more processing power and therefore tends to slow down the mixing process.

Session Files

In addition to waveforms, MIDI files, and video soundtracks, Adobe Audition also makes use of a type of file called a session file. Session files (.ses) are very small in size and simply contain the details of your Adobe Audition multitrack session, as in what audio and video files are used and where, the volume and pan levels, the names of the tracks, what real-time effects are used, etc. With this in mind, you cannot simply copy a session file to a floppy disk and have a friend open it up and play it.

For example, if you create a session with a file called C:\Music\hihat.wav, and later use another program (like Explorer) to move hihat.wav to a new folder called C:\Music\Drums, Adobe Audition won't know that you moved the file the next time you load the session. Try to keep this in mind as you manage your audio elements and session files so that you don't end up losing tracks or objects inside your sessions.

If you want to move an entire session file and all of its embedded audio elements to a new directory, choose File > Save Session As, and select Save Copies of All Associated Files.

Working with Premiere Pro and After Effects

You can easily move projects between Adobe Audition and Adobe's video applications by using the Edit Original command in Premiere Pro and After Effects. To enhance the functionality of this command, you can configure Adobe Audition to link mixdown files with related session files so you can quickly edit and remix soundtracks. To link mixdown and session files, choose Options > Settings, click the Data tab, and select Embed Project Link Data for Edit Original Functionality.

Chapter 2: Looking at the Work Area

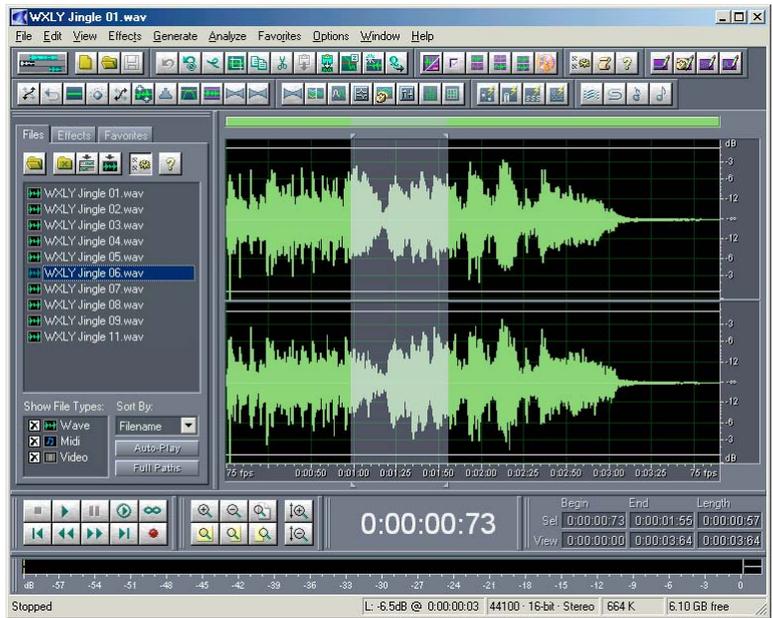
You might think of Adobe Audition as two major audio programs in one. It's a powerful single-waveform editor that can be used to play, record, modify, and process mono and stereo waveforms. And Adobe Audition can also digitally record and mix numerous audio files (using up to 128 tracks) to either a single sound card or multiple sound cards, while providing for real-time effects and level/pan mix adjustments in a non-destructive editing environment.

This chapter is meant to be a brief introduction to the various navigation and functional controls of this deceptively simple – yet powerful – program. Here we'll concentrate on the major interface elements that are common between Adobe Audition's two major work areas: Edit View and Multitrack View. Parts of the program that are more specific to either area will be covered in subsequent chapters.

 *Adobe Audition makes liberal use of your mouse's right button. Whenever you see a simple function button, control, window, or waveform action, try right-clicking it. Chances are you'll be surprised by a useful shortcut menu or a set of handy options that can make Adobe Audition's operation even easier.*

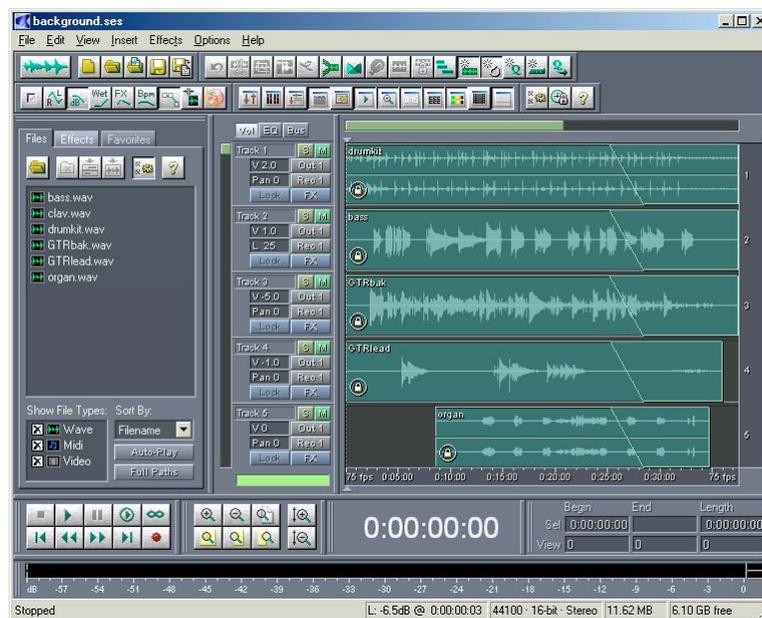
Edit View and Multitrack View

Adobe Audition offers two major workspaces: Edit View and Multitrack View. While in Edit View, Adobe Audition can be thought of as being a single-waveform editor that can be used to record, edit, and process mono and stereo waveforms. Audio files can be saved to disk (including CDs), or played back through any sound card that has been installed within your computer.



Edit View with Organizer Window Active

On the other hand, Multitrack View is where Adobe Audition allows you to record, play, and mix multiple tracks of audio in a virtual 128-track recording studio. Real-time effects can be applied, relative track volume and stereo placement may be adjusted, and audio can be played back through single or multiple sound cards.



Multitrack View with Organizer Window Active

Switching from Edit View to Multitrack View

There are several ways to switch from Edit View to Multitrack View. Here are the most popular:

- Select Multitrack View from Adobe Audition's View menu.
- Click on the left-most icon on the Adobe Audition File toolbar.
- Press F12.

Switching from Multitrack View to Edit View

Several methods are provided to switch from Multitrack View to Edit View. These include:

- Double-click on a waveform's entry in the File tab of the Organizer Window.
- Double-click on a wave block.
- Select Edit Waveform... from a wave block's right-click menu.
- Select Edit Waveform View from Adobe Audition's View menu.
- Click on the left-most icon on the Adobe Audition File toolbar.
- Press F12.

Dockable Windows

If Adobe Audition's default interface layout doesn't quite meet the current needs of your project or screen resolution, you'll be happy to know that many individual parts of the interface are dockable. This means that interface components like the Transport Buttons, the Time Display window, the Level Meters, and Rulers may be detached from their current location to float above Adobe Audition's main window, or they may be repositioned and resized within the main window so they better suit your requirements.

How can you tell if a part of the Adobe Audition interface is indeed a dockable window? Look for two thin vertical or horizontal lines – these lines are the “handle” (or grab bar) of a dockable window. Move your mouse over a handle, and your mouse cursor looks like a plus sign with arrows at each end.

For quick reference, here's a list of all Adobe Audition dockable windows:

- Organizer Window
- Cue List
- Play List *
- Transport Buttons
- Zoom Buttons
- Time Window
- Sel/View Controls
- Level Meters
- Placekeeper
- Session Properties **
- Mixers Window **
- Track EQ Window **
- Track Properties **
- Load Meter **
- Video Window **

* (Available in Edit View only)

** (Available in Multitrack View only)

Some docked windows may be resized. If resizing is possible, the docked window will have a single, thicker horizontal or vertical bar, known as a resize control. If your mouse moves over a resize control, your mouse cursor takes on the appearance of two lines with two arrows.

Note: Even if the resize control is visible, resizing might not be possible due to the other windows that are in the row with the window you're trying to resize.

To select a docked window, left-click when you're over a handle. (Notice how the window becomes outlined when selected.)

To move a docked window, select it and drag its hollow shape while holding down the left mouse button. As you move it around the Adobe Audition interface, two things will happen. If the outline of a dockable window retains its original dimension, releasing the left-mouse button will cause the window to “materialize” as its own standard floating window. However, if you notice the resize bar of another docked window “light up”, this designates a docking location. If this is where you'd like the dragged window to appear, release the left-mouse button and the window will snap into its new location.

To close a docked window, right-click on its handle to see a pop-up menu with a Close option. Check this option and the window will disappear. Don't worry if you change your mind, since you can bring back any closed dockable window by checking its name on Adobe Audition's View menu.

A docked window's right-click menu also contains an option called Force New Row. If this item is checked, a new, empty row to dock windows is created.

To move a floating window, left-click on its title bar and, while holding down the left mouse button, drag the window to the desired location. Close a floating window by left-clicking on the “X” control in its title bar.

To dock a floating window, left-click and drag it while holding down the left mouse button. As you move it around the Adobe Audition interface, potential docking locations will appear; you'll see the resize bar of dockable windows “light up” wherever docking is possible. If this is where you'd like the floating window to dock, release the left-mouse button and the window will snap into its new home.



Press the Ctrl key while moving a floating window around to force it to not dock. That way you can float it over an area that it would normally try to dock to.

Placekeepers

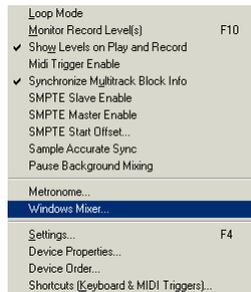
Select Show a Placekeeper from Adobe Audition's View menu to create a blank, dockable window called a “placekeeper”. Use it to more accurately define where you want a dockable window to be, especially for windows that are only useful when viewed in a certain aspect ratio.

For instance, if you try docking the Track EQ controls above the Transport Buttons, they end up going underneath the whole session display, which creates a view that isn't very useful (or aesthetically pleasing). You can use a placekeeper, though, on either side of the Track EQ window in order to force the EQ into a certain aspect ratio. Placekeepers can also be used just for appearance's sake, just because you like the way they let you customize Adobe Audition's look.

Placekeepers may be inserted at any point in the Adobe Audition interface where docking is possible, and they automatically resize to fit their docked area. You can right-click within a placeholder window to select Squares to fill the placekeeper's appearance. Check the Make Default item to make future placekeepers automatically adopt the current appearance.

You can have up to four placekeeper windows, each created with a click of the Show a Placekeeper command. To delete a docked placekeeper, right-click on its grab bar and select Close from the pop-up menu.

Drop-down Menus



The Multitrack View's Options drop-down Menu

Adobe Audition's drop-down menus offer quick and easy access to all session and audio file handling, editing, and signal processing functions, as well as options, viewing properties, and help. Complete details about the commands found in both the Edit View and Multitrack View's drop-down menus can be found later in this manual.

Toolbars

Many of Adobe Audition's most commonly used functions are represented as icons within toolbars, which appear near the top of the main interface. These icons give you instant access to effects, file handling functions, viewing options, and more, at the press of a button.



The Edit View's Edit Toolbar

You can decide which sets of toolbar icons to display by going to View > Toolbars (or right-click on the toolbar area) and checking or unchecking the desired toolbar group(s). Another option is to limit how many rows of toolbar icons Adobe Audition displays at any one time: one row, two rows, or three rows.

Note: To see what a toolbar button does, hold your mouse pointer over it to display a small “tool tip” that describes the function in simple terms.

Organizer Window

Adobe Audition's Organizer Window is a handy, tabbed window that allows you to easily open and close files, see a list of all open waveforms and MIDI files, choose effects with ease, and more. The Organizer Window is hidden by default. To display it, check Show Organizer Window from Adobe Audition's View menu, or press the Alt+9 keyboard shortcut. The Organizer Window initially appears to the left of the Wave Display in Edit View or to the left of the Session Display in Multitrack View. However, it's dockable so you can reposition it at one of several places in the Adobe Audition interface. Or, you may detach it and have it float above Adobe Audition's main window.

To close the Organizer Window when it's floating, click on the "X" control in its title bar. To get rid of the Organizer Window when it's docked, right-click on its grab bar and select Close from the pop-up menu. Or, you can simply uncheck the Show Organizer Window item on the View menu to close it in either docking or floating mode, or press the Alt+9 hotkey.

Files Pane



The Files Pane of the Organizer Window with Advanced Options displayed

The Files Pane of the Organizer Window displays a list of open waveforms and MIDI files. Here are its components:

Open File Click on the Open File button to access Adobe Audition's Open a Waveform dialog. Double-clicking on a waveform in the Organizer opens it for editing in the Edit View. Dragging and dropping a Wave or MIDI file into a track in the Multitrack View drops it in that track.

Close File Pressing this button closes all of the files that are highlighted in the Files pane.

Insert Into Multitrack Push this button to insert all highlighted files, each into their own track, in Adobe Audition's Multitrack View.

Note: If you highlight a file and press the Insert Into Multitrack button multiple times, that file is inserted the same number of times, each time into a separate track. For instance, if no other waveforms are in the Multitrack View, highlighting a file and pressing the Insert Into Multitrack button five times places the same file into tracks 1, 2, 3, 4, and 5.

Display In Edit View Push this button to show a highlighted file in Adobe Audition's Edit View.

Note: Should multiple files be highlighted, only the last file you clicked on will appear in Edit View.

Advanced Options Depressing this button causes additional options to appear near the bottom of the Files pane.

Show File Types You're given the option to show or hide the display of Wave and/or MIDI files in the Files pane. Click on the appropriate box to toggle its view status.

Sort By This control lets you decide how you want to sort the entries in the Files pane: Recent Access, Filename, Recent Type, or Recent Filename.

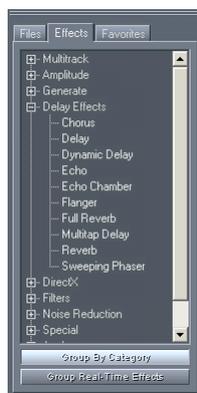
Auto-Play When this button is pushed down, clicking a file's name causes the file to begin playing.

Full Paths Push this button to display the full path [drive, folder(s), filename] of the entries in the File pane. Push it again to show just the filenames.

Note: Hold down the Shift key to select multiple contiguous files in the Files pane; holding down the Ctrl key allows you to highlight multiple non-contiguous entries.

Effects Pane

The Effects Pane lists all of the effects at your disposal. The listing includes all of Adobe Audition's effects as well as all installed DirectX audio plug-ins. Double-clicking on an Effect in the Organizer brings up the dialog to run that effect on the open waveform in the Edit View. Dragging and dropping an Effect onto a track in the Multitrack View adds that effect to the track.



The Effects Pane of the Organizer Window with Group By Category active

Group By Category With this control pushed down, the list is shown as an Explorer-like hierarchical tree-type display. Categories and their entries are shown in the same order as they appear on Adobe Audition's Effects menu. With the Group By Category button in the up position, all effects are displayed in roughly the same order they appear on the Effects and Generate menus.

Group Real-Time Effects When this control is pushed down, the list is shown as an Explorer-like hierarchical tree-type display, where all of the Real Time Effects are grouped together, the Off-Line Effects are grouped together, and the Multi-track Effects are grouped together. Another push of the Group Real-Time Effects button returns the File Pane to its previous view.

Favorites Pane

The Favorites Pane lists all of the favorites you've created. (These are the same items that appear on Adobe Audition's Favorites menu.)

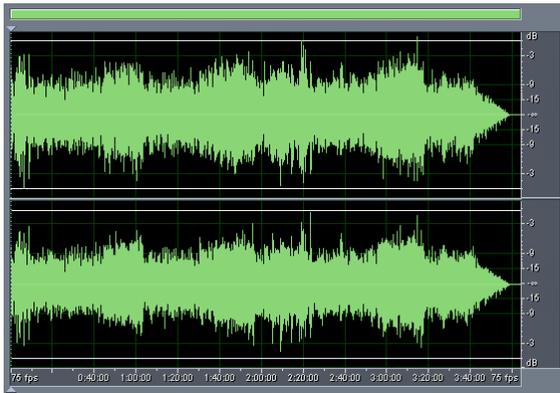


The Favorites Pane of the Organizer Window

Edit Favorites Press this button to call up the Favorites dialog, where you can create new favorites, or edit or delete existing Favorites.

Wave Display/Session Display

The Wave Display window (in Edit View) and the Session Display window (in Multitrack View) are the heart and soul of Adobe Audition. These are the areas where you'll see sound visualized in an easy-to-manipulate form, whether you're modifying one waveform in Edit View or mixing dozens of discrete audio elements in Multitrack View. More details about the Wave Display and Session Display windows can be found in the "Navigating the Edit View" and "Navigating the Multitrack View" chapters, respectively.



The Edit View's Wave Display



The Multitrack View's Session Display

Presets

Many of Adobe Audition's effects and other functions have presets that are available for easily storing and recalling your favorite settings. New presets can be added at any time.



The Presets portion of the Amplify Effect's Dialog

Double-clicking on any preset will instantly set all controls in the dialog box to that preset.

Whenever you have settings you would like to keep, you may enter a name for your settings, and press the Add button. A dialog pops up, prompting you to name your new preset. Your new preset is added to the list of other presets, which is automatically sorted alphabetically.

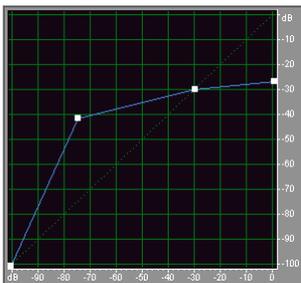
Note: Adobe Audition allows you to add two or more presets with the same name. Normally you wouldn't want to do this as this can be confusing if the presets are different.

To remove a preset from the list, choose the preset, and press Del.

To modify an existing preset, double-click on the preset name, make your modifications, then press Del immediately followed by Add. This will delete the old preset and add your current settings in under the same name.

Graph Controls

Many of Adobe Audition's effects use graph controls for adjusting parameters. By adding and moving control points on the graph, you can tailor the effect to precisely meet your needs.



The Graph Control portion of the Dynamics Processing Dialog

To work with an Adobe Audition graph control:

- To add a control point to the graph, click in the grid at the location where you want to place the point.
- To enter the values for a control point numerically, right click on the point to bring up the edit box, or double-click on the graph's curve.

- To move a point on the graph, click and hold on the point and drag to a new location.
- To remove a point from the graph, click and hold on the point and drag it off the graph.

Note: When the mouse cursor is located over a control point, you'll see it change from an arrow to a hand.

Transport Buttons

Just like many hardware-based audio recording and playback devices, Adobe Audition provides VCR-like transport buttons for playing, recording, stopping, pausing, fast forwarding, and rewinding your waveforms and sessions. The Transport Buttons are in a dockable window, allowing you to reposition them at several places in the Adobe Audition interface. Or, you can even detach the Transport Buttons window and have it float above Adobe Audition's main window.



<<CAPTION TO BE WRITTEN>>

To close the Transport Buttons window when it's floating, click on the "X" control in its title bar. To get rid of the Transport Buttons window when it's docked, right-click on its grab bar and select Close from the pop-up menu. Or, simply uncheck the Show Transport Buttons item from the Edit menu to close it in either docking or floating mode.

Here's what each transport button does:

Stop Click on this button to stop the playback of a waveform or session.

Play This button plays the portion of the wave or session that is currently being viewed (or that's highlighted) from the current cursor position. Right-click on the Play button to select from four behavior options:

- Play View: Adobe Audition will play just the portion of audio you're currently viewing from beginning to end.
- Play from Cursor to End of View: Adobe Audition plays from the current cursor position to the end of the view. (This is the default behavior for this button.)
- Play from Cursor to End of File: Audio is played from the current cursor position until the end point of the file or session.
- Play Entire File: When this option is checkmarked, Adobe Audition will play the entire file or session from beginning to end, regardless of the cursor position or view.

Pause Press this button to temporarily pause the playback or recording of audio. The control turns into a Continue button when audio is paused.

 You may also enable pause in Multitrack View by first holding down the Ctrl key and then pressing the spacebar to start playback. Hitting the spacebar again will pause the playback of the multitrack session.

Play to End Clicking on this button plays the contents of the currently-visible waveform or session window, and then stops playback. Right-click on the Play to End control to select from four behavior options:

- Play View: Adobe Audition will play just the portion of audio you're currently viewing from beginning to end.

- **Play from Cursor to End of View:** Adobe Audition plays from the current cursor position to the end of the view.
- **Play from Cursor to End of File:** Audio is played from the current cursor position until the end point of the file or session. (This is the default behavior for this button.)
- **Play Entire File:** When this option is checkmarked, Adobe Audition will play the entire file or session from beginning to end, regardless of the cursor position or view.

Play Looped This button plays the contents of the currently-visible waveform or session window in a continuous, looping fashion.

 *Right-click on this button to change its behavior. Loop View (or Selection) is the default option; it will loop just the area of the waveform or session that's currently viewed or highlighted. Checkmark Loop Entire (or Selection) to have Adobe Audition loop the entire waveform or session (or just the highlighted portion), regardless of how much is being viewed.*

Go to Beginning Press this button to place the playback cursor at the beginning of the waveform or session.

Rewind Click on this button to shuttle the playbar cursor backwards in time. This function supports scrubbing. This means that on some sound cards, the audio file will be played back at a lower volume as it shuttles over the waveform or session.

 *Right-clicking on the Rewind button allows you to set the rate at which the cursor will move.*

Fast Forward Pressing this button shuttles the playbar cursor forward in time. This function supports scrubbing. This means that on some sound cards, the audio file will be played back at a lower volume as it shuttles over the waveform or session.

 *Right-clicking on the Fast Forward button allows you to set the rate at which the cursor will move.*

Go to End Click this button to place the playback cursor at the end of the waveform or session.

Record This button starts recording from the current playbar cursor position. Any waveform data after this point will be recorded over.

 *Right-click on the Record button to select from different behavior options, which vary depending on whether you're in Edit View or Multitrack View: In Edit View, you can switch between Instant Record (the default) and Timed Record modes. (Timed Record lets you use Adobe Audition as an automated, time-shifting recording device.) In Multitrack View, you have three options: Continuous Linear Record (the default), Loop While Recording (View), and Loop While Recording (Entire). Since the Play Looped button itself only applies to playback, the last two options allow for looping while recording. If areas are marked for Punch In (i.e., the Allow Multiple Takes option is checked on a Wave Block's right-click menu), a new take is created on each looping.*

Level Meters

The Level Meters (which by default are near the bottom of both Adobe Audition's Edit View and Multitrack View windows) display both record and playback levels. The Level Meters are in a dockable window, allowing you to reposition it at one of several places in the Adobe Audition interface. Or, you can even detach the Level Meters window and have it float above Adobe Audition's main window.



Level Meters

The incoming signal from your selected sound card(s) is represented as the peak amplitude in decibels, where a level of 0dB is the absolute maximum before clipping occurs. If clipping does occur, the clip indicator to the right of the meter will light up, and remain lit. Clicking on the clipping indicator at any time resets it. Yellow peak indicators will “stick” for 1.5 seconds before resetting to allow for reading of the peak amplitude.

When displaying stereo audio, the top meter represents the left channel, and the bottom, the right.

To close the Level Meters window when it’s floating, click on the “X” control in its title bar. To get rid of the Level Meters window when it’s docked, right-click on its grab bar and select Close from the pop-up menu. Or, you can simply uncheck the Show Level Meters item on the View menu to close it in either docking or floating mode.

Right-click on the Level Meters window to call up a menu with metering options:

Monitor Record Level This option activates the Level Meters and starts monitoring the recording source. This is useful for setting input levels before recording. To stop monitoring, press the Stop control or tap your spacebar. You can also start and stop monitoring by double-clicking on the Level Meters.

Show on Play and Record Selecting this option causes the Level Meters to operate during the playback or recording of audio.

Clear Clip Indicators If clipping occurs (i.e., incoming audio is greater than 0dB), then the Level Meter’s red lights come on and stay on until cleared. Choose this option to reset the red lights. You may also click on the Clip Indicators (at the right edge of the Level Meters) to clear them.

Note: The clip indicators will always light if clipping occurs, but if Adjust for DC is enabled, the indicators may light up when the audio has a DC offset.

Adjust for DC Many sound cards record audio with a slight DC offset, which means that the center of the waveform being recorded is not at the exact center of the waveform display, but a little above or below it. This can dramatically throw the level meters off since the amount the waveform is displaced could be interpreted as a constant sound that loud. To compensate, make sure this menu item is checked. The recording meters will dynamically adjust to the DC offset, and display the true amplitude of the signal in decibels.

Show Valleys

Just as the yellow indicators show peak levels, if Show Valleys is chosen, valley levels (minimum amplitudes) will be marked as well. This gives a good indication of the dynamic range of the audio. If the valley indicators are close to the peak indicators, the dynamic range – the difference between the quietest and loudest sounds – is low. If they’re spread far apart, the dynamic range is high.

120dB Range With this option checked, the range of the Level Meters is 120 decibels.

90dB Range Select this option to have the Level Meters display a range of 90 decibels.

75dB Range Click on this entry to display a range of 75 decibels in the Level Meters.

60dB Range This option causes the Level Meters to have a range of 60 decibels.

45dB Range When you choose this option, the Level Meters have a range of 45 decibels.

30dB Range Checking this item gives the Level Meters a range of 30 decibels.

Dynamic Peaks Choosing Dynamic Peaks causes the yellow peak level indicators to reset to a new peak level after 1.5 seconds. This lets you easily see the peak amplitude “right now”. As the audio gets quieter, the peak indicators start backing off.

Static Peaks With this option selected, the peak levels never reset. This lets you retain the maximum amplitude of the signal since monitoring, playing, or recording began. The peak can still be reset manually at any time by clearing the clip indicators (clicking on the clip indicator at the right).



Having Static Peaks selected is great for finding out how loud a song will get before recording it. Just start the meters and start playing the song. When the song is over, the peak indicators will show the volume of the loudest part of the song.

Zoom Buttons

Adobe Audition’s Zoom Buttons allow you to zoom in or out of the Wave Display or Session Display. You can zoom in down to the point where the sample of each waveform can be clearly seen, or zoom out to get a complete visual overview of the entire waveform or session.



Zoom Buttons

The Zoom Buttons window is a dockable window, allowing you to reposition it at one of several places in the Adobe Audition interface. Or, you can even detach the Zoom Buttons window and have it float above Adobe Audition’s main window.

To close the Zoom Buttons window when it’s floating, click on the “X” control in its title bar. To get rid of the Zoom Buttons window when it’s docked, right-click on its grab bar and select Hide from the pop-up menu. Or, you can simply uncheck the Show Zoom Buttons item on the Edit menu to close it in either docking or floating mode.

The following items are included within Adobe Audition’s Zoom Buttons:

Zoom in to Center Click this button to zoom in on the center of the visible waveform window or session.

Zoom Out Click this button to zoom out from the center of the visible waveform window or session.

Zoom Out Full Click this button to zoom out to display the entire waveform or blocks that are contained within a session.

Zoom to Selection Click this button to zoom in on the actively selected waveform or session range.

Zoom to Left Selection Click this button to zoom in on the left-hand boundary of the actively selected waveform range or session.

Zoom to Right Selection Click this button to zoom in on the right-hand boundary of the actively selected waveform range or session.

Vertical Zoom Out In Edit View, clicking this button decreases the vertical scale resolution of a waveform's amplitude display; in Multitrack View, clicking this button shows more tracks.

Vertical Zoom In In Edit View, clicking this button increases the vertical scale resolution of a waveform's amplitude display; in Multitrack View, clicking this button decreases the number of viewed tracks in the Session Display.

Time Window

Adobe Audition's Time Window displays the current position of the playback cursor. The default display value is mm:ss:ddd (minutes:seconds:thousandths of a second), but this can easily be changed by right-clicking on the Time Window and choosing another option, such as Compact Disc, Samples, Bars and Beats, or one of several SMPTE values.

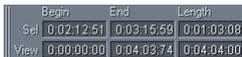


Time Window

The Time Window is dockable, so you can reposition it at one of several places in the Adobe Audition interface. You may also detach the Time Window and have it float above Adobe Audition's main window. To close the Time Window when it's floating, click on the "X" control in its title bar. To get rid of the Time Window when it's docked, right-click on its grab bar and select Close from the pop-up menu. Or, just uncheck the Show Time Window item on the View menu to close it in either docking or floating mode.

Sel/View Controls

The Sel/View Controls window (Sel is an abbreviation for Selection) shows the beginning and ending points, as well as the total length of both the selection and the section of the waveform or session that's currently viewed. Both the selection and display range is shown using the current time display format (which is selected in View > Display Time Format). Left-click on any one of the individual value windows to directly enter numeric time display data. A subsequent right-click calls up a copy/paste and options selection box.



Sel/View Controls

Since Sel/View Controls is a dockable window, you can reposition it at one of several places in the Adobe Audition interface. You may also detach the Show Sel/View Controls window and have it float above Adobe Audition's main window. To close the Sel/View Controls window when it's floating, click on the "X" control in its title bar. To get rid of the Sel/View Controls window when it's docked, right-click on its grab bar and select Close from the pop-up menu. Or, you can simply uncheck the Show Sel/View Controls item on the View menu to close it in either docking or floating mode.

Status Bar

The Status Bar window runs along the very bottom of Adobe Audition's main window. It can display information such as sample format, file size, and free disk space.



Status Bar

Right-click on the Status Bar to show a menu of display options (which are described below). Check the items you wish to display; uncheck the items you don't want to see. You may also access Status Bar options from Adobe Audition's View menu.

Data Under Cursor This shows useful information such as the channel (if a current waveform is stereo), the amplitude (measured in dB), and the time (hours:minutes:seconds:hundredths of seconds) from the beginning of the audio file. This data is computed at the precise point where your mouse cursor is placed within the wave display, and changes dynamically when the mouse cursor is moved. Example: If you see "R: -15.2dB @ 0:00:242" in the status bar when in Edit View, this means that your cursor is over the right channel at 0.242 seconds into the waveform, and the amplitude at that precise point is -15.2 decibels.

In the Multitrack View, you'll see even more beneficial data such as Pan and Volume envelope positions, envelope positions for effects envelopes, dynamic effect settings, and the current position of the wave block as you drag it around.

Sample Format When this item is checked, sample information about the currently loaded waveform is displayed. For instance, a 44,100 kHz 16-bit stereo file shows up as "44100 - 16-bit - stereo".

File Size (K) This value represents how large the active audio file is, measured in kilobytes. If you see "308 K" in the status bar, then the current waveform is 308 kilobytes (KB) in size.

File size (time) This figure tells you the length (measured in time) of the current waveform. For instance, "0:01:247" means the sound file is 1.247 seconds long.

Free Space (K) This option lets you see how much space is available on your hard drive. A typical display might be "7028.32 MB free".

Free Space (time) Check this item to have Adobe Audition display the amount of available time left for recording, based upon the currently selected sample rate. This value is shown as minutes, seconds, and thousandths of seconds. For instance, if Adobe Audition is set to record an 8-bit mono waveform at 11,025 kHz, the time left might read something like “4399:15.527 free”. Change the recording options to 16-bit stereo at 44,100 kHz, and the remaining time value becomes “680:44.736 free”.

Keyboard Modifiers Select this option if you wish to have the status of your keyboard’s Ctrl, Shift, and Alt keys shown in the status bar.

Window Menu – Edit View

When you’re working in Edit View mode, the Window menu gives you a simple way to switch between open files.

Currently Open Waveform List Entries for up to 10 open waveforms are listed.

Note: If you have more than 10 waveforms active, only the last 10 opened will appear on the Windows menu. Think of this section of the Windows menu as a MRU (“Most Recently Used”) list of open waveforms. If you need to access another open waveform whose name isn’t listed, use the Switch To... command or Adobe Audition’s Organizer window.

Switch To Access the Switch To... command to show a list of all open files. Click on a file’s name to make it the active file in the Wave Display.



Switch To Window

Note: If you have a waveform open that has a cue range in it, there will be a + sign next to its name. Click on the + sign to expand the entry so you can jump directly to any of the cue ranges it includes.



Instead of using the Window menu, try using the Files tab of Adobe Audition’s Organizer window. To activate the Organizer window, go to the View Menu and click on Show Organizer Window, or press the Alt+9 keyboard shortcut.

Chapter 3: Wave File Formats Supported by Adobe Audition

The following is a list of the various wave file formats that Adobe Audition currently supports, arranged in the order they appear in Adobe Audition's Save Waveform As dialog. Some of these formats will have options, so look for the Options... button to be enabled in Adobe Audition's Save Waveform As dialog.

Note: If you want to load from or save to a format that's not listed here, you may be able to use an ACM Waveform codec to do so. To do this, use File > Open As or File > Save As, click on Options..., and try to find the format you want to use. When exporting to an ACM format, you may first need to use Edit > Convert Sample Type to convert the file to a format supported by the ACM codec.

64-bit doubles (RAW) (.dbl)

This format is 8-byte doubles in binary form – 8 bytes per sample mono, or 16 bytes per sample stereo interleaved. Just as raw PCM has no header and is just audio data, so does the 64-bit doubles format.

8-bit signed (.sam)

This format is popular for building MOD files, since audio in MOD files is 8-bit signed. Many MOD editors allow samples to be inserted from files, or exported to files in this format. 8-bit signed raw format data with the .sam extension is assumed to be 8-bit signed raw data with no header. The sample rate is assumed to be 22050Hz, but the actual sample rate can be changed once the file is loaded using Edit > Adjust Sample Rate.

A/mu-Law Wave (.wav)

The A-Law and mu-Law formats (CCITT standard G.711) are common in telephony applications. These encoding formats compress the original 16-bit audio down to 8 bits (for a 2:1 compression ratio) with a dynamic range of about 13-bits (78 dB). While A-law and mu-Law encoded waveforms have a higher signal-to-noise ratio than 8-bit PCM, it's at the price of a bit more distortion than the original 16-bit audio. Still, the quality is higher than you would get with some 4-bit ADPCM formats.

Note: Files saved in this format will automatically be expanded to 16-bits when loaded, so you should not save to this format from 8-bit.

Options Choose from the following:

- A-Law 8-bit is a slight variation of the standard mu-Law format, and is found in European systems.
- mu-Law 8-bit is the international standard telecommunications encoding format, and is the default option.

ACM Waveform (.wav):

Nearly any file format supported by the Microsoft Audio Compression Manager (ACM) can be loaded or saved in Adobe Audition. The available ACM formats are displayed in the Choose ACM Format dialog that appears when you press the Options... button on the Save Waveform As dialog.

Note: The ACM driver you want to use may require that the file be in a specific format before saving. For example, if you want to save a file in the DSP Group TrueSpeech format, you should first use Edit > Convert Sample Type to convert the file to 8KHz/mono/16-bit, because that's the only format supported by the TrueSpeech ACM driver. For more information on any particular ACM driver, contact the creator of the format (such as DSP Group for TrueSpeech, or CCITT for the various CCITT formats) or the manufacturer of the hardware that uses the format in question.

Options Set the following:

- Name displays the name of any saved format presets. Three are provided initially: CD, Radio, and Telephone Quality.
- Save As: Click this button to save additional preset format/attribute combinations.
- Filter is the ACM format you wish to save in. Note that this list will display only those formats that are compatible with the current waveform's properties (stereo/mono, resolution, etc.).
- Attributes: The available attributes will depend on the original attributes of the file (sample rate, etc.) and the Filter chosen.
- Write as RAW Data: Check this box to save the waveform as RAW data.

Amiga IFF-8SVX (.iff, .svx)

The Amiga 8SVX format is an 8-bit mono format from the Commodore Amiga computer. A variety of sample rates are supported.

Options Choose from the following:

- Data Formatted As 8-bit Signed: Check this option to save the audio file in uncompressed 8-bit Signed format, which is the default.
- Data Formatted As 4-bit Fibonacci Delta Encoded: With this selection made, the file will be saved using the compressed 4-bit Fibonacci Delta encoded format.
- Dithering from 16-bit: If you're saving from a 16-bit file, choose the type of dithering you wish: Triangular Dither, Shaped Gaussian Dither, Noise Shaping A, or Noise Shaping B. No Dithering is the default.

Apple AIFF (.aif, .snd)

This is Apple's standard wave file format. Like Windows WAV, AIFF files support mono or stereo, 16-bit or 8-bit, and a wide range of sample rates. Adobe Audition only supports the PCM encoded portion of the data, even though this format (like Windows WAV) can contain any one of a number of data formats.

The AIFF format is a good choice for PC/Macintosh cross-platform compatibility. Before you open AIFF files in Adobe Audition, add the .aif or .snd extension to the file and load it using the Apple AIFF file filter. When transferred to a Mac, you can add the four character code “AIFF” in the file’s resource fork to have it recognized (The Macintosh identifies a file through its “resource”, which is stripped away when a file is opened on a PC. However, many Mac applications that support AIFF can recognize the PCM data without this identifier.)

ASCII Text Data (.txt)

Audio data can be read to or written from files in a standard text format, with each sample separated by a carriage return, and channels separated by a tab character. An optional header can be placed before the data. If there is no header text, then the data is assumed to be 16-bit signed decimal integers. The header is formatted as KEYWORD: value with the keywords being: SAMPLES, BITSPERSAMPLE, CHANNELS, SAMPLERATE, and NORMALIZED. The values for NORMALIZED are either TRUE or FALSE. For example,

SAMPLES: 1582

BITSPERSAMPLE: 16

CHANNELS: 2

SAMPLERATE: 22050

NORMALIZED: FALSE

164 <tab> -1372

492 <tab> -876

etc...

Options Choose any of the following:

- Include Format Header places a header before the data.
- Normalized Data normalizes the data between -1.0 and 1.0.

Audition Loop (.cel)

Audition Loop files are essentially MP3 files with a .cel extension, but with a header that contains two important items. The first is the loop info, just as it’s saved with a .wav file and is set under View > Wave Properties > Loop Info. This includes number of beats, tempo, key, and stretch method.

The .cel file also works to avoid a problem with MP3 files. It’s in the nature of MP3 files that a very small amount of silence is added to the beginning and/or end of an MP3 file when it’s encoded. This is very short, and often only a few samples long. When working with a loop, though, it’s enough to throw the entire loop off.

As it saves a .cel file, Adobe Audition calculates how much silence is going to be added to the MP3 file and writes this information into the .cel header. Then, when Adobe Audition loads a .cel file, it reads this silence information and automatically removes the silence from the file so that it'll loop smoothly.

Options The file saving options are identical to the ones for mp3PRO® (.mp3) files, which are documented later in this chapter.

Creative Sound Blaster (.voc)

This is the Sound Blaster and Sound Blaster Pro voice file format. Adobe Audition supports both old and new style .voc files. The older .voc format only supports 8-bit audio; mono to 44.1 kHz, and stereo to 22 kHz. The newer type of .voc file supports both 8-bit and 16-bit audio.

VOC files can contain information for looping and silence. If the file you're loading contains loops and silence blocks, they'll be expanded while loading.

Options Choose one of the following:

- Old Style saves your audio as an 8-bit .voc file that can be played on any Sound Blaster card.
- New Style allows you to save to the newer .voc format that supports both 8-bit and 16-bit audio.

Dialogic ADPCM (.vox)

The Dialogic ADPCM format is commonly found in telephony applications, and has been optimized for low sample rate voice. It will only save mono 16-bit audio, and like other ADPCM formats, it compresses to 4-bits/sample (for a 4:1 ratio). This format has no header, so any file format with the extension .vox will be assumed to be in this format. When opening VOX files, you'll be prompted for a sample rate unless Don't ask for further details is checked in the Open a Waveform dialog.

Note: Take note of the sample rate of your audio before saving as Dialogic VOX, as you'll need to enter it upon reopening the file.

DiamondWare Digitized (.dwd)

This is the audio format used by DiamondWare's Sound Toolkit, a programmer's library that lets you quickly and easily add high quality interactive audio to games and multimedia applications. It supports both mono and stereo files at a variety of resolutions and sample rates. See <http://www.dw.com> for more information.

DVI/IMA ADPCM (.wav)

The International Multimedia Association (IMA) flavor of ADPCM compresses 16-bit data to 4 bits/sample (4:1) using a different (faster) method than Microsoft ADPCM, and has different distortion characteristics, which can produce either better or worse results depending on the sample being compressed. As with Microsoft ADPCM, it's best to save to this format from 16-bit rather than from 8-bit. This compression scheme can be a good alternative to MPEG; it provides reasonably fast decoding of 4:1 compression, and it degrades sample quality only slightly.

Options Choose from the following:

- 2 bits/sample, 8:1
- This option creates ADPCM .wav files with the highest compression ratio (8:1), but with the lowest number of bits. Select this option when smaller file size is more important than audio quality. Keep in mind that this compression rate is less compatible than the standard 4-bit, and may not be able to be played back on all systems.
- 3 bits/sample, 5.3:1 -- The 3-bit ADPCM files produced with this option use a compression ratio of 5.3:1. Quality is better than with the 2 bits option, but not quite as good as the 4 and 5 bits choices. Some systems might have problems playing back files compressed with this option, especially stereo files.
- 4 bits/sample, 4:1 -- This is the default option, which produces 4-bit files at a compression ratio of 4:1.
- 5 bits/sample, 3.2:1 -- Use this option for the best quality, since more bits and a lower compression ratio are used. However, this compression rate is less compatible than the standard 4-bit, and may not work on all systems.

Microsoft ADPCM (.wav)

The Microsoft ADPCM format consists of 4-bit per channel compressed data, which provides 4:1 compression. Files saved in this format will automatically be expanded to 16-bits when loaded, regardless of their original resolution. For this reason, it's best to save to this format from a 16-bit waveform rather than 8-bit, as the quality will be much greater.

Options Choose from the following:

- Single Pass compresses files in a single pass. Use this option if you're in a hurry. However, the quality will be lower than if you use the Multiple Pass option. The time taken to read an ADPCM compressed file is the same no matter which option you choose.
- Multiple Pass compresses files using multiple passes, which provides better quality. This is the default setting.
- Block Size offers three size options, each with a different compression ratio and quality level.
 - Large (Default Quality), with a compression ratio of 3.98:1.
 - Medium (Good Quality), with an increased quality level, but because the compression ratio is a bit smaller (3.81:1), your files will be slightly larger than with the Large option.
 - Small (High Quality) offers the best quality ADPCM sound at a trade-off of larger files due to the smaller compression ratio (3.25:1).

mp3Pro® (.mp3)

The mp3Pro filter enables Adobe Audition to directly encode (save) and decode (open) .mp3 files. When a file is saved as MP3, it's encoded and compressed as specified by the MP3 settings. As an .mp3 file is opened, it's decompressed into Adobe Audition's uncompressed internal format. This allows you to save the .mp3 file back to disk as any file format (including as a standard .wav file).

Check your online documentation and help for details on the options for saving mp3PRO.

Note: The mp3PRO filter encapsulates standard mp3 file support, which means that you can save standard, non-pro mp3s by unselecting the “pro” setting in the mp3 options.

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Next/Sun (.au, .snd)

This format is the standard found on NeXT and Sun computers, and has many data types. Adobe Audition supports the CCITT mu-Law, A-Law, G.721 ADPCM, and linear PCM data variants. Like Windows WAV and AIFF files, this format can support mono or stereo, 16-bit or 8-bit, and a wide range of sample rates when saved as linear PCM.

The most common use for the AU file format is for compressing 16-bit data to 8-bit mu-law data. AU is used quite extensively for distribution on the Internet, and for inclusion in Java applications and applets.

Options Choose from the following:

- mu-Law 8-bit compresses the file using the mu-Law 8-bit format. (This is the default.)
- A-Law 8-bit uses the A-Law 8-bit format to compress the file.
- G.721 ADPCM 4-bit uses the CCITT G.721 (ADPCM at 32 kbits/second) to compress the file.
- Linear PCM saves the file as uncompressed linear PCM.

SampleVision (.smp)

The SampleVision format is used by Turtle Beach's SampleVision program. It supports only mono 16-bit audio. If your data is in a different format, you'll be asked to convert it before saving. This format also supports loop points, which can be edited using Adobe Audition's Cue List.

The Label of the cue must be in the format Loop n, m where n is the loop number from 1 to 8, and m is the mode: 0 = no looping, 1 = forward loop, 2 = forward/back loop. Adobe Audition's Play List is used to enter the number of times to loop the cue range. Add the cue range to the Play List, and then enter the number of times to loop.

Windows PCM (.wav)

The Microsoft Windows WAV format supports both mono and stereo files at a variety of resolutions and sample rates. It follows the RIFF (Resource Information File Format) specification, and allows for extra user information to be embedded and saved with the wave file. WAV files contain PCM coded audio – which is pure, uncompressed Pulse Code Modulation formatted data.

Options These formatting options are available only to 32-bit files; they’ll be disabled (“grayed-out”) at other times:

- 32-bit 24.0 float (type 1 – 24-bit) saves full 32-bit floats (in the range of +/-8million), but the .wav BitsPerSample field is set to 24 while BlockAlign is still set to 4 bytes per channel.
- 32-bit 16.8 float (type 1 – 32-bit) is Adobe Audition’s internal format. Floating point values are in the range of +/- 32768.0, but larger and smaller values are valid and not clipped since the floating point exponent is saved as well. The .wav BitsPerSample field is set to 32 and BlockAlign is set to 4 bytes per channel.
- 32-bit 0.24 normalized float (type 3 – 32-bit) is the standard floating point format for type 3 .wav files. Values are normalized to the range +/-1.0 and although values above and below this range are saved, some programs may clip when reading them back in. (Adobe Audition will not clip, but read the same value back if it’s beyond this range.)
- 24-bit pack int (type 1 – 24-bit) saves straight 24-bit integers so any data beyond the bounds are clipped. The .wav BitsPerSample field is set to 24 and BlockAlign is set to 3 bytes per channel.
- 24-bit packed int (type 1 – 20-bit) saves straight 24-bit integers (so any data beyond the bounds is clipped). The .wav BitsPerSample field is set to 20 and BlockAlign is set to 3 bytes per channel. The extra 4 bits that are saved are actually the remaining valid bits when saving, and are used when reading (thus still giving 24-bit accuracy if those bits were actually present when writing). Applications should either fill those last 4 bits with zeros, or with actual data, but generally analog/digital converters that generate 20 bits of valid data automatically set the remaining 4 bits to zero. Any type 1 format with BlockAlign set to 3 bytes per channel is assumed to be packed integers, and a BitsPerSample field between 17 and 24 inclusively will read in all 24 bits and assume the remaining bits are either accurate or set to zero.
- 32-bit int (type 1 – 32-bit) saves 32-bit audio as 32-bit integers.
- Enable Dithering will dither 32-bit files when saving to a PCM format (20-bit, 24-bit, or 32-bit). This option is only available if you have a 32-bit file and you select to save it as a nonfloating-point type format. The dither used is a Triangular dither, depth 1.0, with no noise shaping. If you wish to apply a noise shaped dither, use Edit > Convert Sample Type... to dither the audio first, then save without dither enabled in the file format options.

Windows Media Audio(.wma)

The Windows Media Audio filter enables Adobe Audition to directly encode (save) and decode (open) .wma files. When a file is saved as a .wma file, it’s encoded and compressed as specified by the .wma settings. As a .wma file is opened, it’s decompressed into Adobe Audition’s uncompressed internal format. This allows you to save the .wma file back to disk as any file format (including as a standard .wav file).

Check your online documentation and help for details on the options for saving Windows Media Audio files.

PCM Raw Data (.pcm) (.raw)

This format is simply the PCM dump of all data for the wave. No header information is contained in the file. For this reason, you're asked to select the sample rate, resolution, and number of channels upon opening.

By opening audio data as PCM, you can interpret almost any audio file format – but make sure you have some idea about the sample rate, number of channels, etc. You can also interpret the data as A-Law or mu-Law compressed. When guessing at these parameters upon opening, the waveform may sound incorrect (in different ways, depending on which parameters you have mixed up). Once the waveform is loaded and sounds fine, you may hear clicks at the start, end, or sometimes throughout. These clicks are various header information being interpreted as waveform material. Just cut these out, and Voilà! – you've read in a wave in an unknown format.

Options Choose from the following:

- Data Formatted As 16-bit PCM (LSB, MSB) saves data formatted as an uncompressed 16-bit PCM file with the Least Significant Bit first and the Most Significant Bit last. With this option you'll be able to offset the input data by +1 byte if desired. (The input data of 32-bit files may be offset by +2 or +3 bytes as well.)
- Data Formatted As 16-bit PCM (MSB, LSB) saves as an uncompressed 16-bit PCM file with the Most Significant Bit first and the Least Significant Bit last. With this option you'll be able to offset the input data by +1 byte if desired. (32-bit files can have input data offset by +2 or +3 bytes as well.)
- Data Formatted As 8-bit mu-Law Compressed saves PCM data in 8-bit mu-Law compressed format.
- Data Formatted As 8-bit A-Law Compressed saves data in 8-bit A-Law compressed format.
- Create .DAT header on save writes a header to a separate .dat file to make reloading the file easier.

Chapter 4: Navigating the Edit View

Edit View is one of Adobe Audition's two major workspaces (Multitrack View is the other). Use Edit View to record, play, convert, and edit single waveforms...and lots more. A wealth of effects are at your disposal, from delay effects like echoes and reverbs, to advanced functions such as noise reduction and dynamics processing. Most of this chapter will concentrate on the largest portion of the Edit View: the Wave Display. The Wave Display is the area where waveforms are visualized and modified.

Note: Interface components that are common to both Edit View and Multitrack View (such as the Level Meters and the Transport Buttons) are documented in the "Looking at the Work Area" chapter, while all of the items referenced on the Edit View's drop-down menus are explained in the Edit View menu chapters.

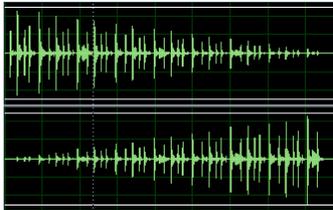
About Edit View

The following is a typical layout for Adobe Audition's Edit View Screen:



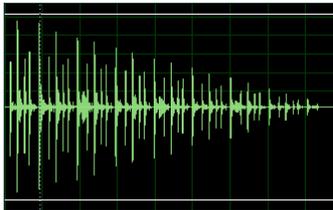
A typical Edit View screen with the Organizer Window open

Stereo vs. Mono If a stereo waveform is loaded, the Wave Display is split horizontally. The waveform’s left channel appears at the top of the window while the right channel is at the bottom.



A stereo waveform

The total height of the Wave Display window is utilized for mono waveforms.



A mono waveform

Navigating the Wave Display

The following movements are crucial to navigating the Wave Display window:

Defining a Range Left-click on one side of the range portion you want to select and drag left or right.

Changing Range Boundaries Hold down the Shift key and left-click on the range boundary you want to move, and drag the boundary to its new location.

Select All Double-click in the Wave Display to select all of the waveform you currently see. Triple-click in the Wave Display to select the entire waveform, whether you can see it all or not.

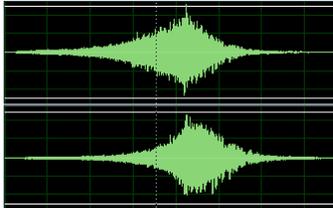
Selecting Only the Left Channel Click near the top of the left (upper) channel to select the left channel only. The mouse cursor will acquire a boxed “L” when you do this.

Selecting Only the Right Channel Click near the bottom of the right (lower) channel to select the right channel only. The mouse cursor will acquire a boxed “R” when you do this.

Waveform View/Spectral View

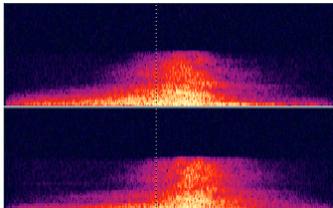
The Wave Display offers two ways to represent audio data: Waveform View and Spectral View. Switch between the two methods using Adobe Audition’s View menu.

- Waveform View – which is Adobe Audition’s default – displays audio data drawn as a series of positive and negative peaks. Spikes in the x-axis (vertical) indicate increased amplitude, and the y-axis (horizontal) represents time.



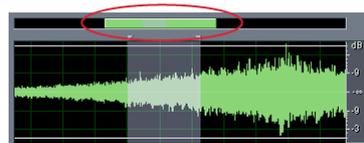
A 15-second stereo .wav file displayed using Waveform View

- Spectral View displays a waveform by its frequency components, where the x-axis is frequency, and the y-axis is time. This allows you to analyze your audio data to see which frequencies are most prevalent. The greater a signal’s amplitude component within a specific frequency range, the brighter the displayed color will be. Colors range from dark blue (which means that next to no amplitude components exist in this frequency range) to bright yellow (meaning that frequencies in this range are high in amplitude).



The same 15-second stereo .wav file displayed using Spectral View

Horizontal Portion Bar Use the Horizontal Portion Bar – which, by default, is at the top of the Wave Display – to scroll forwards and backwards in time throughout a waveform. It can also be used to zoom in and out horizontally in the Wave Display.



Horizontal Portion Bar

- Left-click and drag the little hand over the Horizontal Portion Bar’s “handle” to scroll time-wise through the waveform.

- Left-click and drag the magnifying glass (which appears when your mouse passes over the edge of the Horizontal Portion Bar’s “handle”) to change the viewable range of the Wave Display. Moving the magnifying glass to the left zooms in, while moving it to the right zooms out.
- Right-click anywhere on the Horizontal Portion Bar to call up a menu with three zooming options: Zoom In, Zoom Out, and Zoom Full. The menu also has options for moving the Horizontal Portion Bar to the bottom of the Wave Display, as well as returning it to its default position at the top of the Wave Display.

Horizontal Ruler

The Horizontal Ruler runs along the bottom of the Wave Display and acts as a timeline. It measures time in the same format specified in the View > Display Time Format menu (which is hours, minutes, and seconds by default).



Horizontal Ruler

- Left-click on the Horizontal Ruler and drag the little hand over the ruler to scroll time-wise through the session.
- Right-click on the Horizontal Ruler and drag to select an area to zoom in on. The zoom in occurs when the right mouse button is released.
- Right-click anywhere on the Horizontal Ruler to call up a menu with several options.

Display Time Format This submenu allows you to choose the time format the Horizontal Ruler uses.

Snapping Snapping allows the edges of any block to “snap” to certain time locations when being dragged. Check the snapping behavior(s) desired.

Zooming Six zoom commands are provided here: Zoom In, Zoom Out, Zoom Full, Zoom to Selection, Zoom In to Left Edge of Selection, and Zoom In to Right Edge of Selection.

Vertical Ruler

The Vertical Ruler – at the right edge of the Wave Display – shows the relative amplitude of a waveform over time.



Vertical Ruler

By default its display format is set to Sample Values (the exact sample value of the audio data), but it can easily be switched to three other types – Normalized Values, Percentage, and Decibels – by doing one of the following:

- Right-click on the Vertical Ruler to switch display formats from a pop-up menu.
- Go to Adobe Audition's View > Vertical Scale Format menu and make a selection.
- Double-click on the Vertical Ruler to cycle through all of the display formats.

Note: When the wave display is set to Spectral View, the vertical scale is always in Hertz (Hz).

The Vertical Ruler also provides access to the following display options:

- Left-clicking and dragging the little hand vertically over the Vertical Ruler allows you to scroll through Adobe Audition's visible amplitude ranges that might extend beyond the traditional amplitude scale values.
- Right-clicking and dragging over the Vertical Ruler automatically zooms the waveform window to show only the specified amplitude scale.
- Right-clicking on the Vertical Ruler calls up a menu that not only lets you choose one of the aforementioned display formats, but provides three Zoom options:
 - Zoom In: Decreases the vertical scale resolution of a waveform's amplitude display incrementally.
 - Zoom Out: Increases the vertical scale resolution of a waveform's amplitude display incrementally.
 - Zoom Full: Zooms out the vertical scale resolution of a waveform's amplitude display completely.

Recording in Edit View

Recording with Adobe Audition in Edit View is easy. You can record from a microphone, your computer's CD player, a MIDI source, or any signal you can plug into the "Line in" port of your sound card.

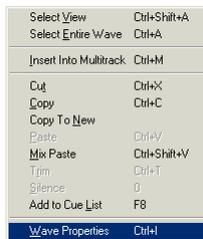
Note: You may need to adjust your input signal to obtain the optimum recording and signal-to-noise levels. (For most folks this is done using the Windows Mixer.)

To record in Edit View.

- 1 Use File > New to create a new file.
- 2 Select the sample rate, bit resolution, and number of channels (stereo or mono) that you want to use and press OK.
- 3 Click on the Record button in the Track Controls window to begin recording.
- 4 When you're done recording, click on the Stop button in the Track Controls window.
- 5 Edit as needed, and then save your recording to disk using File > Save As.

Wave Display Right-Click Menu

Right-click in the Wave Display window to bring up a menu with several useful commands. Notice that most have keyboard shortcuts listed next to their name. You'll find that using these shortcuts for frequently used commands will speed up the editing process considerably.



Wave Display Right-Click Menu

Note: By going to *Options > Settings*, you can change the behavior of your mouse's right-click button in the Wave Display. By default, a Wave Display right-click brings up a menu. However, you can push the *Extend Selection* radio button in the Settings window's *General* tab to have a right-click extend a selection in the Wave Display; this means you'll then need to hold down the *Ctrl* key as you right-click to display the menu.

Select View This command selects the contents of just the Wave Display area that you're viewing.

Select Entire Wave The entire waveform is selected (highlighted) with this command.

Insert Into Multitrack This command takes the currently highlighted portion of a waveform and inserts it as a waveform block in the Multitrack View window. If there isn't a highlighted segment, the entire current waveform is inserted. Either way, the inserted audio is placed in the next available track at the beginning of the window's timeline.

Cut Use Cut to remove the current waveform, or highlighted part of the waveform, to the active clipboard.

Note: While Adobe Audition can use the Windows clipboard for cut and paste operations, by default it uses its own internal clipboard which is faster and more suitable for working with large waveforms. (As a matter of fact, it puts five such "super clipboards" at your disposal.)

Copy This command copies the current waveform, or highlighted part of the waveform, to the active clipboard. Keep in mind that if you want to copy audio to another Windows application, the current clipboard must be set to Windows. Otherwise, use one of Adobe Audition's internal clipboards, which are faster and can handle more data.

Copy to New When this command is run, the current waveform, or highlighted part of the waveform, is copied to the active clipboard and then automatically pasted into a new Wave Display window.

Paste Paste inserts data from the active clipboard at the insertion point, or it replaces the currently highlighted data. If the format of the waveform data on the clipboard is different from the format of the file it's being pasted into, Adobe Audition automatically converts it before pasting.

Mix Paste Use Mix Paste to mix any audio data from the active clipboard with the currently shown waveform. (You might think of it as a “quick and dirty” alternative to Adobe Audition’s more powerful and flexible multitrack functions.) Clipboard data is inserted or overlapped beginning at the current insertion point or selection. Should the format of the waveform data on the clipboard differ from the format of the file it’s being pasted into, Adobe Audition automatically converts it before pasting.

Trim Trim is used to get rid of unwanted audio (like silence) on one or both sides of the selected audio. Everything in the waveform is removed except for the highlighted selection.

Note: *The deleted portion(s) aren’t copied to the clipboard and can only be retrieved using the Edit > Undo command, or by File > Revert to Saved if you haven’t saved the file since trimming.*

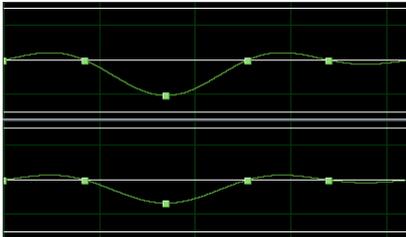
Silence This function silences out the selected range of the waveform. Unlike deleting or cutting a selection – which splices the surrounding material together – Silence leaves the duration of the selection intact, and simply zeroes the amplitude within it.

Note: *You must select part or all of your waveform first in order for Silence to work.*

Add to Cue List This command adds the current cursor position to Adobe Audition’s Cue List.

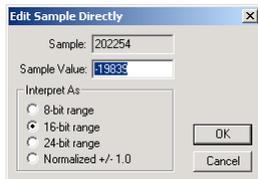
Wave Properties Access this command to bring up a tabbed window that lets you embed text and graphics, specify certain parameters, and get information about the active waveform.

Edit Sample Directly If you Zoom In far enough in Edit View, you can modify each sample of a waveform directly. To do this, select a small portion of the waveform and use the Zoom to Selection button (one of Adobe Audition’s Zoom Buttons) to zoom in as far as you can go.



The Wave Display zoomed in to the sample level

Instead of looking “wavy,” the waveform is displayed as a sequence of squares strung together by a line. Those squares are samples. Move your mouse cursor over one. You’ll notice the cursor turns into a hand pointer. Left-click on a sample and drag it up or down as desired. Right-click on a sample to display the Edit Sample Directly dialog.



The Edit Sample Directly dialog.

Here are the major components of the Edit Sample Directly dialog:

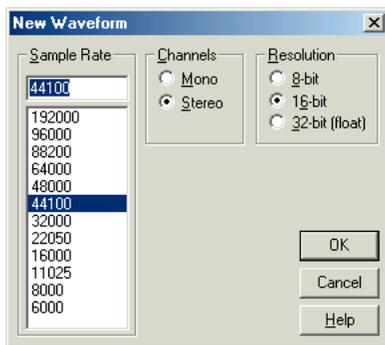
- Sample is the number of the sample as measured from the beginning of the waveform.
- Sample Value is the value of the sample as measured using one of the four Interpret As options below. Negative numbers mean that the sample is below the horizontal center of the waveform, while positive numbers mean that the sample is above the waveform's center.
- Interpret As is four options that determine how the Sample Value is measured. Choose from 8-bit range, 16-bit range, 24-bit range, or Normalize +/- 1.0.

Chapter 5: File Menu – Edit View

Use the commands on the File Menu to open, save, and close wave files; import audio from video files; perform batch conversions; access recently used waveforms; and more.

New

Select File > New to create a new, empty waveform. The New Waveform window appears so that you can choose the new file's properties. Options are provided for Sample Rate, Channels, and Resolution.



New Waveform Dialog

 *You'll get better-quality sound when you use a higher sample rate and resolution, but you'll use more hard disk space. Audio CDs use a sample rate of 41000 and a resolution of 16 bits per sample.*

Sample Rate This value determines how many frequencies can be encoded in the audio signal. Higher sampling rates mean a wider bandwidth. Popular sample rates are already listed; just click on one to select it. Or, enter a custom number in the Sample Rate box to specify a different rate from those provided. However, keep in mind that most sound cards only support certain sample rates.

Channels Select Mono if you want to record a waveform with just one channel of audio information (such as a voice-only recording).

The Stereo option allows you to create a two-channel waveform with separate right and left channels. This is usually the best option for a music recording. Because they contain twice as much data, stereo waveforms consume double the storage space of mono waveforms.

Resolution Three options are provided:

- 8-bit creates waveforms where quality is not much of a concern, but small file size is. 8-bit waveforms are usually fine for telephony applications, or for use as embedded sounds in Web pages. While they tend to be noisier than their 16-bit counterparts, they're half the size.

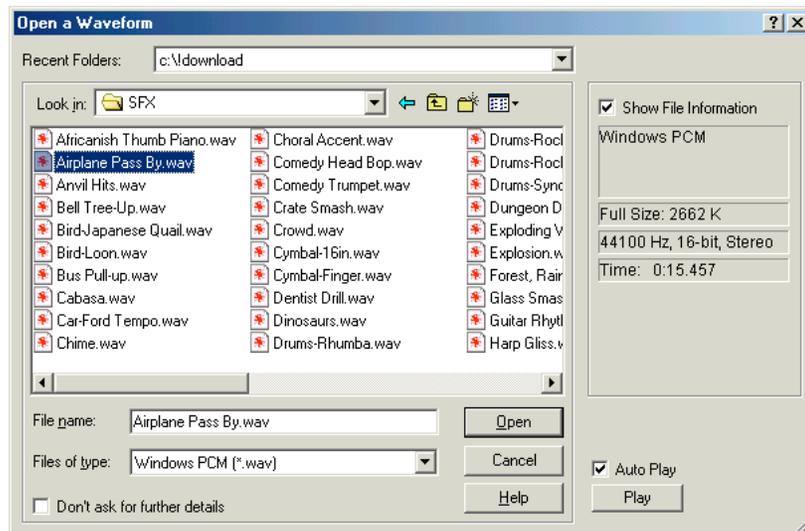
- 16-bit produces CD-quality waveforms. This setting is suitable for most broadcast and music recording.
- 32-bit creates a waveform that lets you work at the 32-bit level. One benefit of 32-bit audio is more precise audio processing, plus the ability to take advantage of the power in sound cards that go beyond the 16-bit standard (such as a 24-bit card). Press the Options button in the Edit View's File > Save As dialog to access 32-bit options.

Note: Although Adobe Audition can create and edit 32-bit files, your sound card may not be able to play them properly. If that's the case, you'll need to convert waveforms stored and processed at this rate to a lower bit rate (such as 16-bit) for playback. To see the capabilities of your system, go to Options > Device Properties.

Open

Use this command to load an existing waveform or CD track into an Adobe Audition editing window. The Open a Waveform dialog appears, allowing you to select one or more audio files for listening or modification. This dialog looks and works similarly to a standard Windows Explorer Open window (including allowing you to use the Ctrl and Shift keys to select multiple non-contiguous and contiguous files respectively), but adds useful Adobe Audition-specific features.

No matter which of the wide variety of supported waveform types you load, Adobe Audition automatically converts the audio file into an internal temporary waveform for quicker editing and better handling of large files.



Open a Waveform Dialog

Here are the components of the Open a Waveform dialog:

Recent Folders This shows the folders you've most recently opened with Adobe Audition. Select a folder name from the drop-down box to quickly navigate to it.

Look in Use this control to navigate through all available drives and folders.

File Name Here's where you may specify the name of the file you desire to open, if it resides in the current folder. Entering *.extension (such as *.wav or *.mp3) displays all files with that extension in the open folder.

Files of type Use this control to filter the display of files in the current folder to a specific type of supported waveform, such as Windows PCM (*.wav) or Next/Sun (*.au;*.snd). As you'd expect, selecting the All files (*.*) listing displays all files in the current folder.

Don't ask for further details If you check this box, you won't be prompted for more information about the waveform format after you choose a file to open. For instance, if you double-click on a raw, headerless PCM file, Adobe Audition won't ask you for the sample rate, bit resolution, A-law/mu-law compression, or any other details. Instead, the program uses the last settings specified when you opened a headerless file.

Show File Information With this box checked, Adobe Audition displays useful details about the selected waveform such as format type, full size, sample rate, and running time.

Auto Play Check this box to have Adobe Audition automatically play a sound file when you highlight it in the Open a Waveform window. This handy feature lets you quickly audition files before you load them into an editing window. (Keep in mind that you can only play file types that are supported by your system.)

You can also play individual files when Auto Play is unchecked. Just select a waveform and click on the Play button in the Open a Waveform window. Terminate playback by clicking on the Stop button, which replaces the Play button when a sound is playing.

***Note:** When multiple files are open in the Edit View, you can easily select between them holding down the Ctrl+Tab keys simultaneously, which shifts between open files in the forward direction. Use the Ctrl+Shift+Tab key combination to move between open waveforms in the reverse direction.*

Open As

The File > Open As command is similar to File > Open, except that it lets you specify the sample format before opening a file. Think of it as a quick way to make changes to one or more existing waveforms' sample rate, bit rate, and/or channel type.



Since this function doesn't pre-filter or post-filter the samples, you'll get better-sounding results when changing sample rates by opening the file(s) normally and using the Edit > Convert Sample Type function.

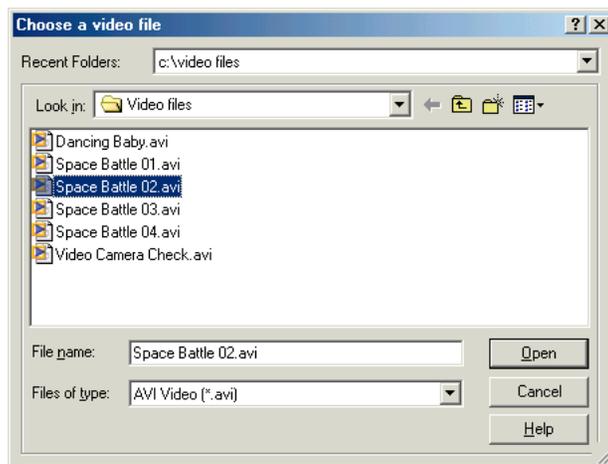
Open Append

Use this command to add one or more waveforms to the end of the currently shown waveform in the editing window. If the waveform being added is of a different type, Adobe Audition automatically converts it as it's opened. However, better results are achieved if you only append files of the same sample rate, since no number-crunching is required to change between rates.

Note: Appended cue points are automatically placed within Adobe Audition’s Cue List, making it simple to locate the edit boundaries later.

Extract Audio from Video

The command lets you extract the soundtrack from any type of video file that Adobe Audition supports. The Choose a Video File window (similar to a standard Windows File > Open window) appears, allowing you to select one or more video files. The video’s audio is placed in an editing window, where it can be modified just like any supported sound file.

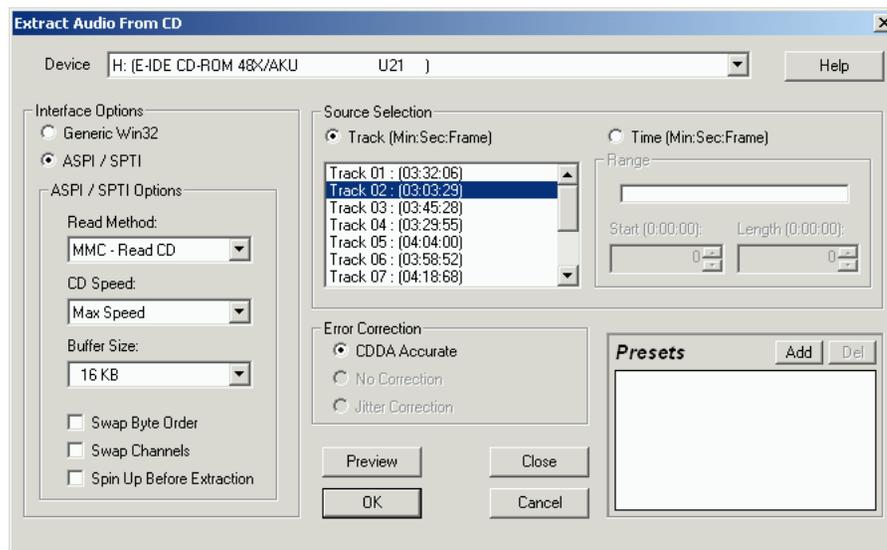


Choose a Video File Dialog

Extract Audio From CD

Use Adobe Audition’s Extract Audio From CD feature to digitally pull out sound from audio compact discs and place it into the Wave Display for editing. This extraction process is commonly known as “ripping”. (Keep in mind that you must have a CD-ROM drive that supports audio extraction in order to use this feature. Luckily, most newer CD-ROM drives do.)

While CD tracks may be ripped using Adobe Audition’s File > Open dialog, The Extract Audio From CD function gives you more control over the extraction procedure, such as the abilities to rip partial tracks and specify the ripping process used.



The Extract Audio From CD dialog

Here are the major components of the Extract Audio From CD dialog:

Device If your system has more than one CD-ROM drive, use this drop-down list to select the drive you want to use for extraction.

Source Selection Two methods of choosing the exact audio you wish to extract are provided: Track and Time.

- **Track:** If you want to extract one or more complete CD tracks, push the Track button. A list of all tracks on the compact disc is shown, along with their lengths stated in Min:Sec:Frame format. (There are 75 frames in each second of compact disc audio.) Click on the desired track names to select them for extraction, simultaneously holding down the Ctrl key to choose non-contiguous tracks or the Shift key for contiguous selection.
- **Time:** To extract part of a track or a segment of audio that spans multiple tracks, push the Time button. Enter the beginning frame in the Start box, and the total number of frames you wish to extract in the Length box. (There are 75 frames in one second of CD audio.) The actual start and length times are displayed in Min:Sec:Frame format above their respective boxes. The Range bar provides a graphical representation of not only the amount of audio that will be extracted, but where the audio appears within the CD. However, if only a short bit of audio is selected for extraction, you might not see any change in the appearance of the Range bar.



The Time option is great for pulling “hidden” tracks from CDs, as well as to join tracks that have been broken up by track indices (such as performance track CDs and live albums).

Interface Options Two main options are provided: Generic Win32 and ASPI/SPTI. Under normal circumstances, ASPI/SPTI should be selected. (ASPI, developed by Adaptec, stands for Advance SCSI Programming Interface. SPTI is an acronym for SCSI Pass Through Interface, and was implemented by Microsoft as an Operating System-supported way of sending SCSI commands; at this writing it's only available on Windows NT, Windows 2000, and Windows XP.)

Generic Win32 should only be used if the ASPI/SPTI setting doesn't produce satisfactory results. (The Generic Win32 option causes the Extract Audio From CD feature to use Input/Output control codes instead of SCSI commands.)

When the ASPI/SPTI button is pushed, several additional options are available:

- Read Method is a drop-down menu that lets you choose the way Adobe Audition reads CD audio. Several methods are provided, many of them developed before the SCSI 3 specifications were published. (The SCSI 2 specs made no accommodations for CD ripping.)
 - MMC – Read CD: This is the SCSI 3 specific command, and it's the command that most all recent drives will work with. If you have a newer CD-ROM drive, try this setting first.
 - SBC – Read10: This is a standard SCSI read command that uses a 10-byte SRB (SCSI Request Block). All SCSI devices are required to support this command.
 - SBC – Read6: This is a standard SCSI read command that uses a 6-byte SRB (SCSI Request Block). Many SCSI devices support this command, but since it's optional, not all do.
 - Plextor (D8): Use this setting with older Plextor CD-ROM drives. It sends the D8 SCSI Op Code to the CD-ROM drive.
 - D5: This command sends the D5 SCSI Op Code to the CD-ROM drive.
 - NEC (D4): Use this setting with older NEC CD-ROM drives.
- CD Speed: This drop-down control lists all extraction speeds that your selected CD-ROM drive supports, and lets you choose the speed you want to use. Many people will find satisfactory results with the Max (Maximum) Speed option. However, if extraction at maximum speed produces errors, try choosing a slower speed.
- Buffer Size: The figure in the Buffer Size box specifies the amount of data Adobe Audition calls into the CD Extraction module to fetch, which therefore determines how much data is pulled from the CD in each call to the read command. The default is 16 KB, but you're welcome to experiment with the other sizes (which range all the way to the highest buffer size your CD-ROM drive supports). While higher sizes mean faster ripping, they could introduce errors into the ripped file.
- Swap Byte Order: Some CD-ROM drives designed to work only with other types of computers (like DEC and Macintosh systems) report data using the Little Endian byte order, while PCs need data reported using the Big Endian method. Checking this box changes the byte order from Little Endian to Big Endian or vice-versa. You would normally leave this box unchecked; check it only if the extraction process seems to work fine but the audio results are "garbage".
- Swap Channels: When this box is checked, the left channel of the CD's audio is placed in the right channel of Adobe Audition's Wave Display, while the CD's right channel audio is placed in the Wave Display's left channel.

- **Spin Up Before Extraction:** Some CD-ROM drives have better accuracy when they first read the CD if the drive is already spinning. If your drive is like this, check the Spin Up Before Extraction box. Otherwise, having this box checked doesn't really provide any advantages.

Error Correction If a CD-ROM drive supports SCSI3 and ReadCD, it's automatically "CDDA Accurate". This means that the drive has ripping error correction built in. For those types of drives, no error correction is needed, so you won't be able to select any options from this part of the Extract Audio From CD dialog. However, if your drive isn't CDDA Accurate, you have access to No Correction and Jitter Correction options.

- No Correction, as you'd expect, means that no error correction will be performed.
- The Jitter Correction option compensates for data reading problems that older drives might have.

Presets The Extract Audio From CD dialog supports Presets, which allow you to save and recall favorite settings. To add the current settings as a preset, press the Add button. A box will pop up to allow you to name the preset. To remove a stored preset, select it and press the Del button.

Preview Press the Preview button to audition the CD audio that will be extracted. (Your CD-ROM drive doesn't need to be physically hooked up to your sound card in order to hear the preview, since Adobe Audition actually rips the preview audio.) When the audio begins to play, the Preview button turns into a Stop button. Click it to discontinue the preview.

Revert to Saved

Use Revert to Saved to reload the currently visible waveform from disk. This discards any edits that were made since the last time the file was saved.

Close

This closes the currently displayed editing window. If any changes have been made to the waveform since the last time the file was saved to disk, Adobe Audition asks you if you wish to save them. When a waveform is closed, its associated temporary file is automatically deleted as well.

Close All Waves and Session

Use this command to close all editing windows and the current multitrack session (if any) simultaneously. You're prompted to save, if necessary.

Close Only Non-Session Waveforms

All waveforms not currently inserted in a multitrack session are closed when this command is used. You're given the option to save any modified, unsaved files.

Save

This saves a modified waveform to disk using the waveform’s existing name and format. No options are provided.

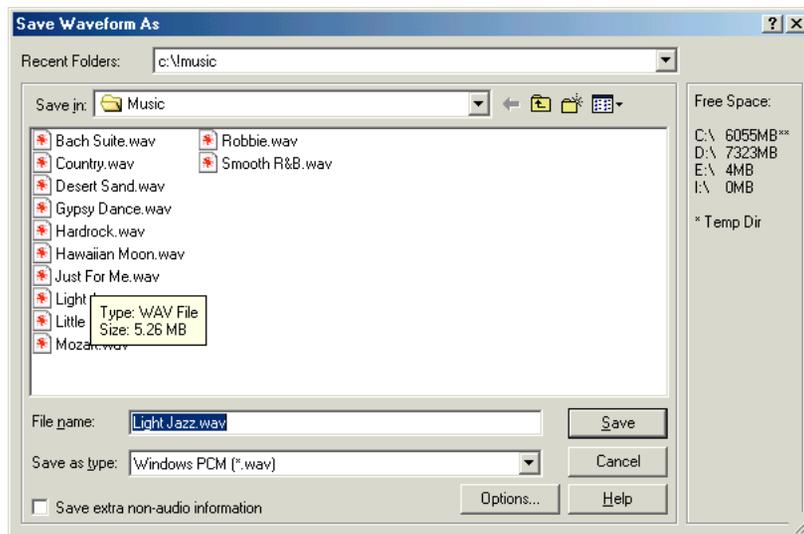
Note: To save newly recorded waveforms, use the File > Save As command.

Save As

This command brings up the Save Waveform As window, which allows you to save the current waveform to a new name, location, or format.

The File > Save As function is also used to save newly recorded waveforms to disk.

Free space on all available drives is displayed in the right section of the window.



Save Waveform As Dialog

Recent Folders This shows the most recent folders that Adobe Audition has saved files in. Select a folder name from the drop-down box to quickly navigate to it.

Save in Use this control to navigate through all available drives and folders.

File Name Here’s where you specify the name of the file.

Save as type Use this control to choose the format you wish to save from the drop-down list of supported file types, such as Windows PCM (*.wav) or MPEG 3 (*.mp3).

Note: Some formats can't be written to using waveforms of certain sample types. In this case, Adobe Audition asks you if you want to convert before saving.

Adobe Audition also warns you if you're trying to save a waveform to a compressed (lossy) format, and suggests that you keep a backup of your original file if you wish to preserve full audio fidelity. If you don't want to see this dialog box again, check the Don't display this message in the future box.

Save extra non-audio information Uncheck this box if you don't want to save header fields such as copyright/author information and cue marks. If you plan to burn the waveform to compact disc using another program, unchecking the box is highly recommended. Some CD recording applications interpret non-audio information incorrectly and will place an unpleasant burst of noise as the beginning of each CD track.

Options Most audio file formats support various options. (See the Waveform File Formats chapter for more information.) Press the Options button to bring up a window to select the desired settings.

If the format doesn't have any options, the Options button is disabled ("grayed out").

Save Selection

File > Save Selection works similarly to the File > Save As command, except it allows you to save just the highlighted selection of a waveform to its own file.

 *This feature is very useful for saving small segments of a larger file. For instance, should you wish to record a number of samples from a friend's sampler, you could record them as a single, continuous file and then split them into their own waveforms. Or, you can break up a long music recording into smaller, more manageable tracks.*

Save All

Run the File > Save All command to automatically save all open, modified waveforms to disk.

Batch File Convert

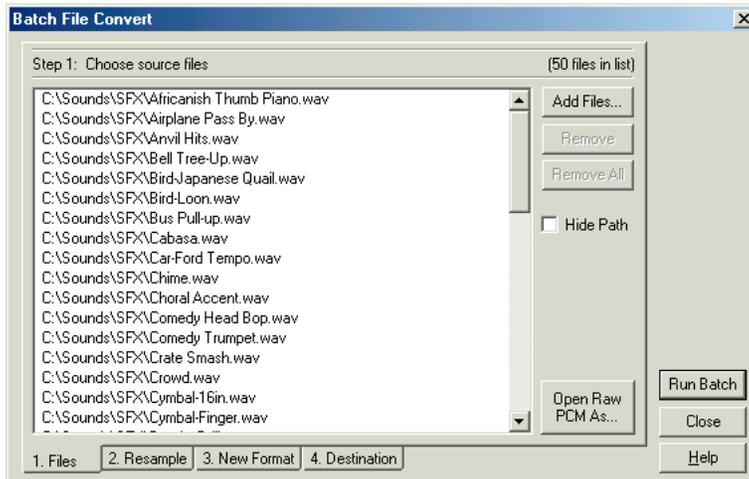
Adobe Audition's handy Batch File Convert tool allows you to easily change multiple waveforms from one audio format to another (such as from .wav to .mp3) using a four-window process.

1 Files

This window is where you add files to the list of waveforms to be converted.

Press the Add Files button to bring up the Please Choose the Source Files dialog (similar to Adobe Audition's File > Open dialog) to select one or more waveforms.

Hold down your Ctrl or Shift keys to select non-contiguous or contiguous files respectively. You can even select an entire folder with the Add Folder button.



The Files Screen of the Batch File Convert Dialog

Press the Remove button to delete highlighted files from the list, or click on the Remove All button to delete all files from the list.

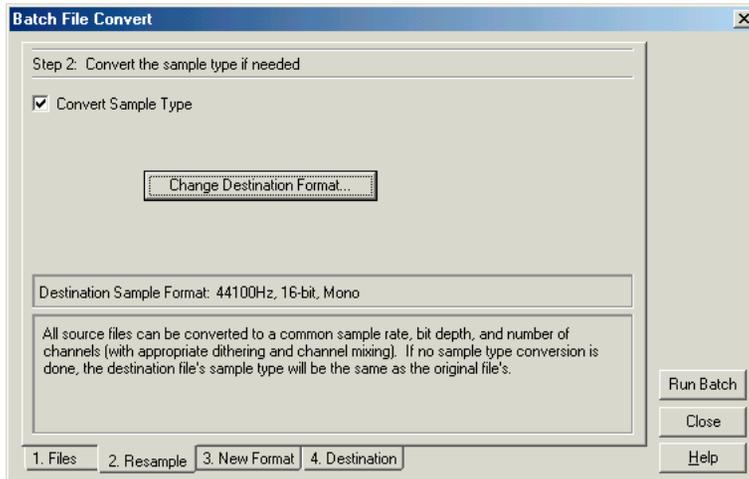
Check the Hide Path box if you don't want to see the full path of each file.

If you're converting Raw PCM files, press the Open Raw PCM As button to pick the desired Sample Rate, Channels, Resolution, and other properties.

2 Resample

Check the Convert Sample Type box if you wish to change each waveform's sample properties to a common set of values. This enables the Change Destination Format button.

If the box is unchecked, the destination sample format is the same as the source and the Change Destination Format button is disabled.

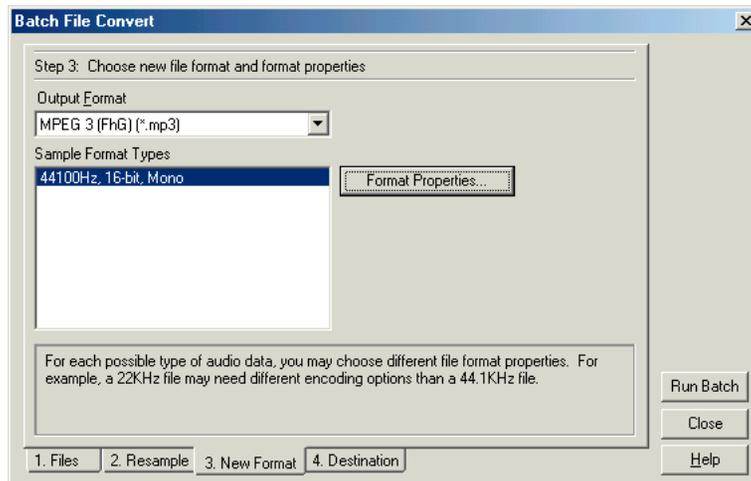


The Resample Screen of the Batch File Convert Dialog

Pressing the Change Destination Format button brings up Adobe Audition's Convert Sample Type window, where you can select the destination files' sample properties.

3 New Format

Here's where you choose the format of the destination files. Use the Output Format drop-down control to pick the desired format from all of Adobe Audition's supported file types.



The New Format Screen of the Batch File Convert Dialog

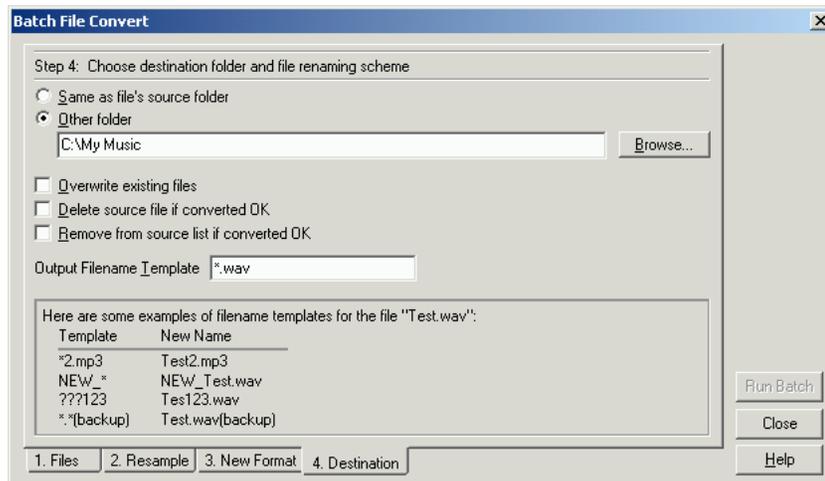
Click on the Format Properties button to bring up the Options screen for the destination format. For example, if you're converting waveforms to .mp3 format, you'll see the MPEG-1 Layer 3 Encoder Options screen, where you can adjust the properties for the .mp3 files you're creating.

The Sample Format Types box lists the sample properties of the waveforms that are to be converted. If you see more than one entry in this list, you may have to select different file format properties for each entry, depending on the destination format. For instance, a 22 kHz mono waveform might need different encoding options than a 44 kHz stereo file. Highlight the desired Sample Format Type entry and press the Format Properties button to make the necessary adjustments.

4 Destination

This screen allows you to select the destination folder and the way files are renamed. By selecting the desired radio buttons and check boxes, you can:

- save modified files in the same folder as the file's source file or another folder.
- save existing files with the same name overwritten.
- delete the source file(s) if the conversion process was successful.
- remove filenames from the source list once their waveforms have been successfully converted.



The Destination Screen of the Batch File Convert Dialog

Decide how you want the files renamed by modifying the text in the Output Filename Template field. In most cases, the default renaming method is fine (the first part of the filename remains the same but the extension is changed). However, several example templates are shown as guides in case you wish to be more creative in your renaming. After the desired parameters have been set in Batch File Convert's four tabbed screens, press the Run Batch button and Adobe Audition automatically processes each file.

Flush Virtual File

Adobe Audition opens waveforms and uses them directly without making a backup in the temp folder.

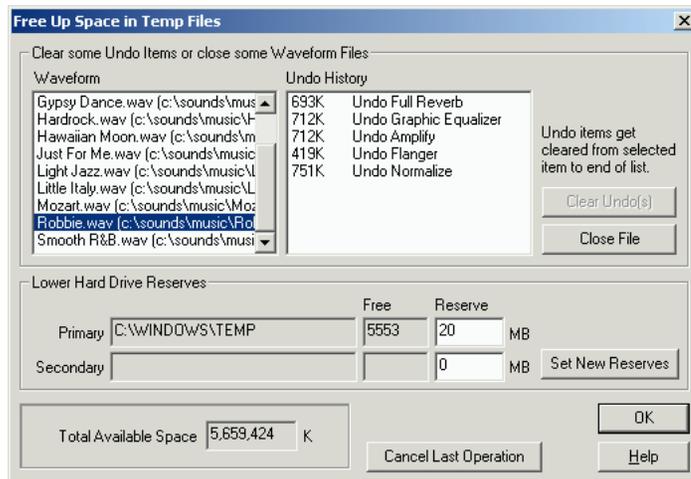
If a waveform needs to be closed for deleting or for opening in another program, running the Flush Virtual File command copies the waveform's contents to the temp folder and closes the file. Files are automatically flushed if the entire waveform is modified by Adobe Audition, since all changes are written to the temp folder.

In other words, this command lets an audio file that's currently being used by Adobe Audition to be simultaneously used by another program or resource application.

Free Up Space in Temp Files

Use this command to bring up the Free Up Space in Temp Files window for managing Adobe Audition's use of the space on your hard drive(s), including the instances of Undo for each waveform.

The window also automatically appears when available hard drive space nears zero kilobytes. If your system has lots of available free hard drive space, Adobe Audition's undo levels are cleared regularly, so you may not see this window unless you manually call it from the File menu.



The Free Up Space In Temp Files Dialog

Waveform The Waveform list displays the names of the currently open waveforms. Select the name of the file for which you'd like to remove Undo instances.

Press the Close File button to close the highlighted file if you no longer need it opened in Adobe Audition. This frees up hard drive space as well. (The waveform in the currently active editing window can't be closed this way, however.)

Undo History This part of the window lists the “undone actions” that are currently being retained on your system. The amount of hard drive space each instance consumes is shown.

To remove items from the Undo History list, highlight an item from the list and press the Clear Undo(s) button. All Undo actions at the selected level and below are removed.

Lower Hard Drive Reserves Adobe Audition creates temporary files for use when performing audio edits. The Primary and Secondary drives listed are those that have been set in the System tab of the Options > Settings window, and are used to store these temporary files.

You can temporarily change that reserved space by entering a different amount in MB that you wish to keep free on those drives. In case that free space gets low, Adobe Audition won't make use of it. Press the Set New Reserves button to save the new values you've entered as the program's defaults.

Total Available Space This field displays the total amount of hard drive space available to Adobe Audition. (It's the sum of the free space on your Primary and Secondary drives.) This value should increase as you free up space.

Cancel Last Operation Press this button to stop any action in progress, such as the application of an Effect or any other edit. The becomes useful only if the Free Some Hard Drive Space window has been automatically invoked by Adobe Audition, indicating that you have run out of storage space.

MRU List

The Most Recently Used list displays the last nine audio files opened in Adobe Audition. You can quickly open any of these files (providing they haven't been deleted or moved) by selecting it from the list with your mouse. Or, type the number that's associated with the desired file.

Exit

As you'd expect, this command closes Adobe Audition. If there are any modified waveforms that are open that haven't been saved to disk, you'll be asked if you want to save them. All temporary files that Adobe Audition has created are deleted as well.

Chapter 6: Edit Menu – Edit View

The Edit Menu displays all of the options that have to do with basic waveform editing in the Edit View mode.

T Undo

This command reverses the last action, as long as the Undo function is turned on (see Enable Undo/Redo below).

Adobe Audition conveniently tells you which action you're undoing. For instance, this menu item might say, Undo Delete or Undo Normalize. If you haven't yet edited a waveform, or Undo has been disabled, this menu item says Can't Undo or Undo Disabled instead.



If you forget which editing action you last performed on a waveform, check out the Undo item on the Edit menu to jog your memory, whether you want to undo the action or not.

About Undo

Adobe Audition's Undo information is stored as a temporary file on your hard drive. If the Undo function is enabled in the **Systems** tab of **Options > Settings**, an unlimited number of undos can be retained. (Actually, the number is only limited by the amount of free hard disk space).

All changes are automatically saved as one or more files in your temporary folder as long as the waveform is open in Adobe Audition – even after it's been saved to disk. This gives you the freedom to “experiment” or make mistakes, knowing that Adobe Audition's Undo function is there to cover your backside when you want to revert back to the last edit you performed.

When working with very large audio files, it's possible that you might not have enough free disk space to save the Undo information before continuing on with an edit. It's also possible that the time required to save the Undo information of very large audio files can slow your work down. In either case, the problem can be solved by turning off the Undo function – if you don't mind working without a “safety net”. Undo can be turned off three ways:

- 1 On the **System** tab of Adobe Audition's Settings window (**Options > Settings**)
- 2 By removing the check mark from the **Edit > Enable Undo/Redo** menu item
- 3 By pressing the *Skip* button in the Saving Undo Data dialog while Undo information is being saved.



*If you find that you don't have enough disk space to save the Undo information, you can change the Temp folder to a different drive, if available. To do this, go to **Options > Settings** and click on the **System** tab. Change the path in the Temp Folder field so that it points to another drive and folder.*

Enable Undo/Redo

Highlight and click Enable Undo/Redo to toggle its status. If a checkmark appears next to this menu item, Adobe Audition's Undo and Redo functions are turned on. If you don't see a checkmark, Undo and Redo have been disabled.

Repeat Last Command

Just as you'd expect, this item repeats the last command. Most editing functions in Adobe Audition can be repeated using this command, but there are a few exceptions (such as Delete).

Set Current Clipboard

Adobe Audition gives you access to five internal clipboards for temporary data storage. Each works similarly to the Windows clipboard, except without the memory limitations.

Use this command to decide which of Adobe Audition's five available internal clipboards is active – just check the one you wish to use. Or, choose Windows to have Adobe Audition use the Windows clipboard. (You'll want to use the latter option if you want to copy audio data to other Windows applications.)



Set Current Clipboard Menu

Copy

This command copies the current waveform, or highlighted part of the waveform, to the active clipboard (see Set Current Clipboard).

Note: Keep in mind that if you want to copy audio to another Windows application, the current clipboard must be set to Windows. Otherwise, use one of Adobe Audition's internal clipboards, which are faster and can handle more data.

Cut

Use Cut to remove the current waveform, or highlighted part of the waveform, to the active clipboard.

Paste

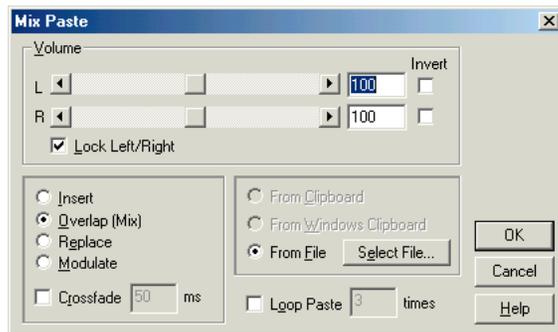
Paste inserts data from the active clipboard at the insertion point, or it replaces the currently highlighted data. If the format of the waveform data on the clipboard is different from the format of the file it's being pasted into, Adobe Audition automatically converts it before pasting.

Paste to New

This command creates a new file and inserts waveform data from the active clipboard. The new file automatically inherits the properties (sample rate, sample frequency, etc.) from the original clipboard material.

Mix Paste

Use Mix Paste to mix any audio data from the active clipboard with the currently shown waveform. (You might think of it as a “quick and dirty” alternative to Adobe Audition’s more powerful and flexible multitrack functions.)



Mix Paste Dialog

Clipboard data is inserted or overlapped beginning at the current insertion point or selection. Should the format of the waveform data on the clipboard differ from the format of the file it’s being pasted into, Adobe Audition automatically converts it before pasting. The major components of the Mix Paste window are as follows:

Volume Move the volume sliders to adjust the sound level of the left and right channels of the audio in the clipboard before pasting. Or, you can type in a value (represented in percentage) in the fields to the right of the volume sliders.



Paste in single channels (either left or right) by adjusting the level of the opposite channel to zero.

Invert Check the desired Invert box to physically turn that channel of the waveform upside-down. (Any samples above the center line will be placed below the line, and those below will come out above the line.)



This function comes in handy when you want to take the difference (or subtract one signal from another) between two samples.

Lock Left/Right When this box is checked, both volume sliders are locked so that when one is moved to a certain level, the other moves simultaneously to that same level.

Insert Choose this radio button if you wish to insert waveform data from the active clipboard at the current location or selection, replacing any selected data. If no selection has been made, Adobe Audition inserts clipboard audio at the cursor location, moving any existing data to the end of the inserted material.

Overlap When Overlap is selected, the clipboard audio is mixed at the selected volume level with the current waveform. If the clipboard audio is longer than the current waveform, the current waveform is automatically lengthened to accommodate the pasted audio.

Replace This option pastes the contents of the active clipboard beginning at the cursor location, and replaces the existing material thereafter for the duration of clipboard data. For instance, pasting 5 seconds of material replaces the first 5 seconds after the cursor with the clipboard’s contents.

Modulate Select this option to modulate the clipboard audio with the current waveform for an interesting effect. This is similar to overlapping, except that the values of the two waveforms are multiplied by each other, sample by sample, instead of added.

Crossfade Check the Crossfade box and enter a value (in milliseconds) to apply a fade to the beginning and end of the clipboard data. Adobe Audition fades in the first n milliseconds and fades out the last n milliseconds of pasted data.



Use this option for smoother transitions to and from pasted audio.

From Clipboard n Select From Clipboard n to paste audio directly from the active internal clipboard.

From Windows Clipboard Choose this option if you want to paste audio from the Windows clipboard. If there’s no audio data in the Windows clipboard, this selection is disabled.

From File Select this option and press the Select File... button to browse for an audio file to paste into the active waveform. The Please Choose a Wave File window appears, which is similar to Adobe Audition’s Open dialog.

Loop Paste When this box is checked, the audio data in the clipboard is pasted the number of times entered. Should the clipboard audio be longer than the current selection, the current selection is automatically lengthened accordingly.

Copy to New

When this command is run, the current waveform, or highlighted part of the waveform, is copied to the active clipboard and then automatically pasted into a new Wave Display window.

Insert in Multitrack

This command takes the currently highlighted portion of a waveform and inserts it as a waveform block in Adobe Audition’s Multitrack View. If there isn’t a highlighted segment, the entire current waveform is inserted. Either way, the inserted audio is placed in the next available track at the beginning of the window’s timeline.

Insert Play List in Multitrack

Use this command to take all the entries in the Adobe Audition Play List and insert them into the Multitrack View as its own item.



Use this command if you use the Play List to assemble long montages that come out of a single file. Once you’re done, you can insert the montage into the Multitrack View, make desired changes, and then render the montage to a new single file.

Select Entire Wave

Choose this command to select the current waveform. It doesn't make any difference if you can't see the entire waveform on screen – the whole thing is highlighted.



You may also select the entire wave by double left-clicking within the Wave Display window.

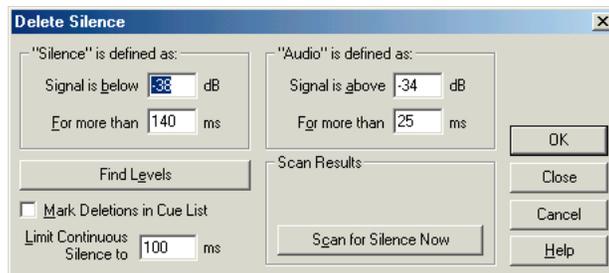
Delete Selection

This command removes the highlighted portion of the waveform.

Note: Deleted data doesn't go to the clipboard and can only be retrieved using the *Edit > Undo* command, or by *File > Revert to Saved* if you haven't saved the file since deleting.

Delete Silence

Use this feature to have Adobe Audition automatically detect and remove silence between words or other audio. It's ideal for cleaning up voice prompts and speeding up narratives without affecting the foreground audio.



Delete Silence Dialog

Here are the major components of the Delete Silence window:

“Silence” is defined as Enter the amplitude value (in decibels) you wish Adobe Audition to consider as the maximum level for Silence in the Signal is below box. The duration (in milliseconds) of this maximum amplitude value should be entered in the For more than box.

Note: For very quiet, high-quality audio, the dB value will be lower (like -60 dB). For noisier audio, the value may be much higher (such as -30 dB). Use higher values in the second box to keep groups of words together, for example.

“Audio” is defined as Enter the amplitude value (in decibels) you wish Adobe Audition to consider as the minimum level for Audio in the Signal is above box. The duration (in milliseconds) of this minimum amplitude value should be entered in the For more than box.



Use higher values for the second box to ignore short periods of undesired audio (like clicks, static, or other noise). However, if the value is too high (above 200 ms), short words may be skipped.

Find Levels Press this button to scan the waveform (or a highlighted section) to have Adobe Audition automatically determine a good starting point for signal levels. After completing the scan, suggested values are automatically placed in the appropriate boxes.

 *If these values aren't doing the job – say, words or phrases are being chopped off – try lowering the signal level values. Increase the signal level values if not enough silence is being removed.*

Mark Deletions in Cue List Check this box to have each location where silence is removed added to Adobe Audition's Cue List.

Limit Continuous Silence to Enter a value (in milliseconds) in this box to set a minimum amount of silence to keep at all times.

- If a silent region of less than this length is found, it's not removed.
- If a silent region of more than this length is detected, enough silence is removed so that there's exactly this length of silence left.

Setting this value to zero will remove as much silence as possible.

 *When shortening speech segments, a setting of 150 ms or so will leave a more realistic, natural-sounding pause where the larger region of silence was found. Removal of too much silence leads to an artificial sound.*

Scan for Silence Now Press this button to pre-perform the silence deletion. Instead of actually deleting, it reports how much silence will be removed, as well as how many sections of silence were found. This gives you an idea of what to expect using the current settings when you actually commit to the Delete Silence command.

 *If you have an audio presentation that consists of many cuts separated by silence (such as a reel of several jingles), use Edit > Delete Silence to make sure the duration of silence between each cut is the same. For instance, if the difference between cuts 1 and 2 is 3.2 seconds, the difference between cuts 2 and 3 is 4.1 seconds, and the difference between cuts 3 and 4 is 3.7 seconds, you can use Delete Silence to make the silence duration between all four cuts exactly 3 seconds.*

Trim

Use Trim to get rid of unwanted audio (like silence) on one or both sides of the selected audio.

Essentially, Trim is the exact opposite of the Delete Selection command. Everything in the waveform is removed except for the highlighted selection.

Note: Remember that the deleted portion(s) aren't copied to the clipboard and can only be retrieved using the Edit > Undo command, or by File > Revert to Saved if you haven't saved the file since trimming.

Zero Crossings

This command adjusts the starting and ending points of the current selection to the nearest place where the waveform crosses the center line (zero amplitude point).

Having a selection start and end on a Zero Crossing is useful in deleting or inserting a selection in the middle of a waveform. If you don't line up the amplitudes (at zero, in this case) on both sides of the selection, the end points drawn together when the highlighted selection is removed are at different amplitudes, which often results in an audible pop or click.



You might want to use this feature when you wish to delete a portion of a waveform, or insert a portion from the middle of the waveform.

To use Zero Crossings:

- 1 Select the portion you want to delete (or position the cursor where you want to insert).
- 2 Choose Edit > Zero Crossings and the appropriate submenu entry:
 - Adjust Selection Inward: Adjusts both region boundaries inward to the next available zero crossing
 - Adjust Selection Outward: Adjusts both region boundaries outward to the next available zero crossing
 - Adjust Left Side to Left: Adjusts the left region boundary leftward to the next available zero crossing
 - Adjust Left Side to Right: Adjusts the left region boundary rightward to the next available zero crossing
 - Adjust Right Side to Left: Adjusts the right region boundary leftward to the next available zero crossing
 - Adjust Right Side to Right: Adjusts the right region boundary rightward to the next available zero crossing

Find Beats

Use this function to have Adobe Audition locate the boundaries that make up musical beats in an existing music audio file. This lets you easily find the starting and ending loop points when you want to construct drum loops and similar musical phrases.

To use Find Beats:

- 1 Place the play cursor in the Wave Display to the left of your targeted loop starting point.
- 2 Select Edit > Find Beats > Find Next Beat (Left) to locate the beginning of the current beat.
- 3 Press Shift+[until you arrive at the desired loop starting point.
- 4 Choose Edit > Find Beats > Find Next Beat (Right) to select from the current cursor position (now the loop start) to the next beat.
- 5 Press Shift+] until you arrive at the desired ending for your loop.
- 6 Press the Play Looped button in the Transport Controls to audition your loop.
- 7 After any necessary tweaking, you can then save, paste, or add the loop to the Cue List.

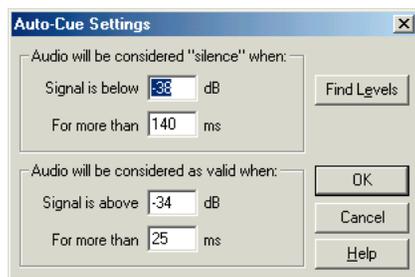
Note: *If the waveform levels don't match up at the loop point, you may hear audible clicking as the loop repeats. To adjust for this, you can revert to the original source file and use the Zero Crossings feature to match initial and ending levels.*

Auto-Cue

The Auto-Cue feature is used to define spoken/music phrases or beats, and automatically add them to the Cue List.

To use Auto-Cue, first highlight the general region in which you wish to find phrases or beats. Then, select Edit > Auto-Cue and the appropriate submenu choice from the following options:

- Adjust Selection to Phrase: Selects a phrase within the highlighted region by adjusting the highlight inward, ignoring any silence before and after the audio. Nothing is added to the Cue List.
- Find Phrases and Mark: Scans the highlighted area, marking non-silent regions as Basic cues in the Cue List.
- Find Beats and Mark: Scans the highlighted area, marking beats as Beat cues in the Cue List.



Auto-Cue Settings Dialog

- Auto-Cue Settings: Choose this option to customize the way the Auto-Cue function works. Here are the major components of the Auto-Cue Settings window:

- Audio will be considered “silence” when: Enter the amplitude value (in decibels) you wish Adobe Audition to consider as the maximum level for Silence in the Signal is below box. The duration (in milliseconds) of this maximum amplitude value should be entered in the For more than box.

Note: For very quiet, high-quality audio, the dB value will be lower (like -60 dB). For noisier audio, the value may be much higher (such as -30 dB). Use higher values in the seconds box to keep groups of words together, for example.

- Audio will be considered as valid when: Enter the amplitude value (in decibels) you wish Adobe Audition to consider as the minimum level for Audio in the Signal is above box. The duration (in milliseconds) of this minimum amplitude value should be entered in the For more than box.



Use higher values for the second box to ignore short periods of undesired audio (like clicks, static, or other noise). However, if the value is too high (above 200 ms), short words may be skipped.

- Find Levels: Press this button to scan the waveform (or a highlighted section) to have Adobe Audition automatically determine a good starting point for signal levels. After completing the scan, suggested values are automatically placed in the appropriate boxes.



If these values aren't doing the job – say, words or phrases are being chopped off – try lowering the signal level values. Increase the signal level values if not enough silence is being removed.

- **Trim Digital Silence:** This command removes pure digital silence from the beginning and ending of a file. However, if you highlight the middle of a waveform, Trim Digital Silence will act as a normal Trim command, trimming out everything else, in addition to any digital silence in the highlighted region at the endpoints.

Snapping

In this case, the term “snapping” means that whenever a waveform cursor is placed within a certain distance of a defined location point timeline or cue location point, that cursor automatically “jumps” or “snaps” to that precise boundary point. Keep in mind that snapping only occurs if you’re close to the ruler tick.

Four options are available, all accessed from the Edit > Snapping submenu. Highlight and check one to select it. (Only one of the Snap to Ruler commands can be checked at a time.)

Snap to Cues Use this option to allow a selected waveform to snap to a defined Cue point. (See the View > Cue List section of this guide for information on how to define Cue points.)

Snap to Ruler (Coarse) This option allows the selected waveform to only snap to the major numeric divisions (decimal, SMPTE, samples, etc.) in the timeline window. If you need more precise snapping, check Snap to Ruler (Fine), which is described below.

Snap to Ruler (Fine) Select this option to allow the selected waveform to snap to each of the subdivisions (decimal, SMPTE, samples, etc.) within the timeline window. Zooming in on the timeline – which is done by right-clicking and dragging within the timeline across the desired waveform area – will generally break the display down into more accurate subdivisions. This lets you place a waveform more accurately within the timeline.

Snap to Zero Crossings Choose this option to have the waveform snap to the nearest place where the waveform crosses the center line (in other words, the zero amplitude point).

Snap to Frames (Always) When Snap to Frames (Always) is chosen, the selection or your cursor is always right on a frame boundary in whatever time format you’re in – as long as the time format selected is measured in frames (such as Compact Disc and SMPTE). This option is especially handy for working on audio for CD.



Snapping settings may be easily accessed by right-clicking on the timeline.

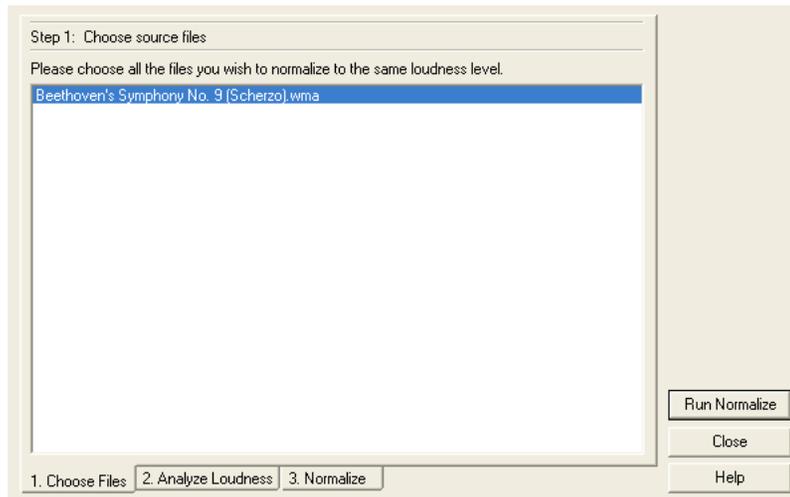
Group Waveform Normalize

Group Waveform Normalize allows you to normalize the volume of multiple open waveforms using a three-screen batch process. (When you normalize a waveform, the loudest part of the waveform is set to a specified amplitude, thereby raising or lowering all other parts of the same waveform by the same amount.) If the volume is raised as part of the normalization process, Adobe Audition can apply limiting to prevent clipping.

If you’re getting ready to master an audio CD, using Group Waveform Normalize is a great way to make sure that all tracks on the CD have a consistent volume.

1 Choose Files Tab

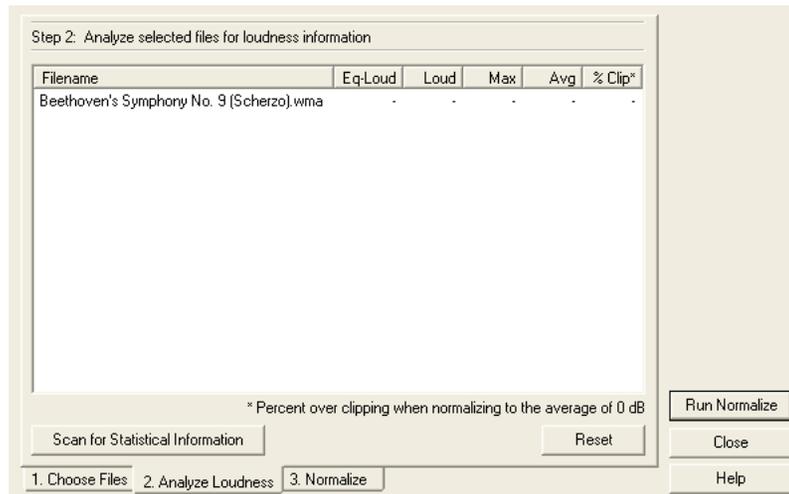
A list of all the currently open waveforms is shown. Choose the ones you wish to normalize by using the Ctrl and Shift keys to select multiple non-contiguous and contiguous files respectively. (This is similar to the way you pick multiple files in Windows Explorer.) You can also just click and drag to highlight a group of waves.



The Choose Files Tab of the Group Waveform Normalization Dialog

2 Analyze Loudness Tab

The waveforms you've chosen to normalize appear in this list. Press the Scan for Statistical Information button to have Adobe Audition analyze the current amplitude statistics for each listed waveform.



The Analyze Loudness Tab of the Group Waveform Normalization Dialog

These statistics are shown:

EQ-Loud This is the final loudness value with an equal-loudness equalization curve that takes into account frequencies the human ear is more or less sensitive to. If Use Equal Loudness Curve is chosen in the final Normalize step, then this is the value that will be used to determine how much to amplify the audio to normalize it.

Loud This is the final loudness value without equal-loudness equalization. This value will be used to determine how much to amplify the audio to normalize it if the Use Equal Loudness Curve is not checked in the final step.

Max This is the maximum RMS amplitude found. This value is based on a full-scale sine-wave being 0dB, and uses the window width given in the Advanced section of the Normalize tab.

Avg This is the total average RMS of the entire file. This value isn't used to normalize with because files can have more interesting dynamics that need to be accounted for.

% Clip This is the percentage of the waveform that would be clipped when normalizing. The value used to determine the amount of normalization is either the Loud or Eq-Loud value, depending on if the Use Equal Loudness Curve box is checked in the Normalize tab. If limiting is used, then actual clipping won't occur, but instead the louder portions of audio will be limited to prevent actual clipping.

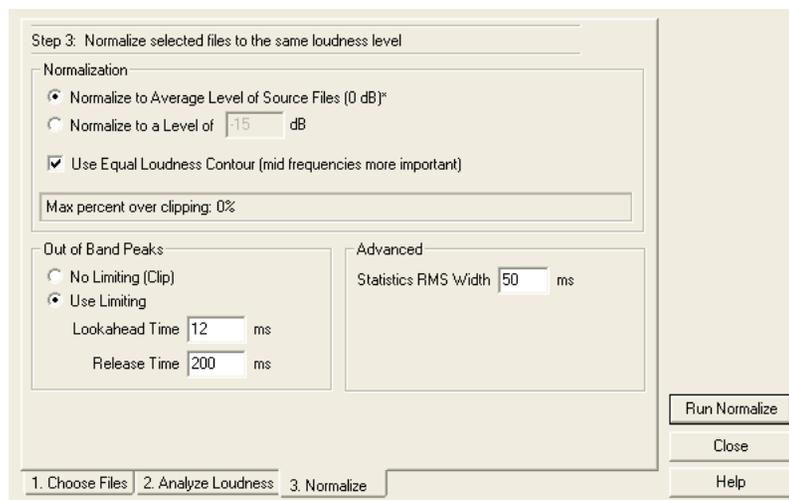
Generally, you don't want this value to be more than 5%. Anything higher than this may result in audible artifacts in the louder portions of audio.

Reset Press this button to clear all of the normalization statistics for the files in the list. You can select any subset of files and clear just those.

💡 Double-click on any of the files in this list to see a more detailed account of statistics including: a complete RMS Histogram, which shows the relative amounts of audio at each loudness level, and a Clipping Profile, which shows how much clipping will occur for each decibel of amplification.

3 Normalize Tab

This is the area where you specify how you'd like to have Adobe Audition normalize the listed waveforms.



The Normalize Tab of the Group Waveform Normalization Dialog

Normalization You can normalize to an average level, or to a specific level.

- Normalize to Average Level of Source Files normalizes each file using the average level of all of them.
- Normalize to a Level of lets you specify the normalization level (entered in decibels) for the whole batch of waveforms.

Note: Unlike Effects > Amplitude > Normalize, percentages aren't used with the Group Waveform Normalization function because it's not peak-based, but RMS-based. The peaks may not go above 25%, or may go over 150% and be the same loudness.

Use Equal Loudness Contour Check this box to have Adobe Audition use an equal loudness contour, where the middle frequencies are most important. You'd normally want to have this box checked. Since the ear is less sensitive to very high and very low frequencies, and much more sensitive to those frequencies between 2 kHz and 4 kHz, two different pieces of audio with the same RMS amplitude, but that contain different frequencies, will sound like they have differing volumes. Checking the Equal Loudness Contour box will ensure that the audio has the same perceived loudness regardless of what frequencies are actually present in the sound file.

Out of Band Peaks

These settings determine how Adobe Audition handles out of band peaks. When amplifying audio, the audio samples may go beyond clipping. If this happens, you can choose to just let it clip (and cause distortion) – or you can apply limiting to those areas so the audio doesn't actually clip (which is what's commonly done for the audio of TV commercials so they sound louder).

No Limiting (Clip) If this option is chosen, no limiting will be performed, which means you could end up with clipped (distorted) audio.

Use Limiting Choose this option to have Adobe Audition's Hard Limiter spring into action, if needed, to keep out of band peaks from clipping.

- **Look Ahead Time:** This is the amount of time (measured in milliseconds) that it will generally take to attenuate the audio before the loudest peak will be hit.

Note: If this value is too small, audible distortion effects may be heard. Make sure the value is at least 5 ms. (The default is 12 ms.)

- **Release Time:** This is the time (measured in milliseconds) that it takes for the attenuation to rebound back 12dB (or roughly the time it would take audio to resume normal volume if an extremely loud peak was encountered). Generally around 100 ms (the default) works very well, and will preserve very low bass frequencies.

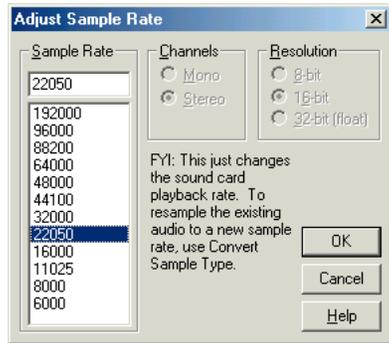
Note: If this value is too long, audio may just stay very quiet, and not resume normal levels for a while. (The default is 200 ms.)

Advanced Here's where you can modify the Statistics RMS Width, which is measured in milliseconds. The default is 50 ms.

Run Normalize After adjusting the parameters to your preferences, press the Run Normalize button to close the Group Waveform Normalization dialog and perform normalization on all selected waveforms.

Adjust Sample Rate

Use this command to bring up the Adjust Sample Rate window. It lets you play an audio file at any sample rate that is supported by your sound card. Keep in mind that this command doesn't let you actually convert the audio file – you'll need to go to Edit > Convert Sample Type for that. However, it does let you hear what the current waveform would sound like at a different sample rate.



Adjust Sample Rate Dialog

Choosing a higher sample rate results in Adobe Audition playing back the file at higher pitch (think “chipmunks”), while selecting a lower sample rate results in a lower-pitched sound (think “drunk Darth Vader”).

Sample Rate The sampling rate you choose determines the overall bandwidth (how many frequencies can be encoded within the audio signal). Higher sampling rates yield a wider bandwidth. You may enter any sample rate directly, or choose a common sample rate from the list.

Note that most sound cards support only certain sample rates. Certain combinations of sample rate, channels, and resolution may not be available on your system. To see the capabilities of your system, check Options > Settings > Device Properties. Although you can create and edit any sample type with Adobe Audition, your sound card may not be capable of playing it properly.

About Sample Rates

During the sampling process, an incoming analog signal is sampled at discrete time intervals. At each interval, this analog signal is momentarily “held” for observation and thus represents a specific, measurable voltage level. A mathematical conversion is used to generate a digital series of numbers that represent the signal level at that particular instant in time. Once the conversion has been made, it can be digitally stored or processed.

The sampling rate is defined as the number of samples (or snapshots) that are taken of an audio signal per second. Since sampling is tied directly to the component of time, a system’s sampling rate determines a system’s overall bandwidth – in other words, how many frequencies can be encoded within the audio signal. Higher sampling rates generally yield a better-quality waveform.

The most commonly encountered sample rates used for digital audio editing are:

- 11,025 Hz Poor AM Radio Quality/Speech (low-end multimedia)
- 22,050 Hz Near FM Radio Quality (high-end multimedia)
- 32,000 Hz Better than FM Radio Quality (standard broadcast rate)

- 44,100 Hz CD Quality
- 48,000 Hz DAT Quality
- 96,000 Hz DVD Quality

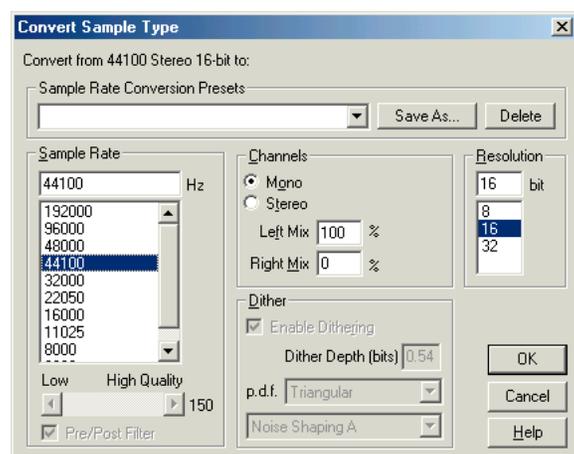
Non-standard sample rates can be entered directly into the numeric sample rate window.

Channels This setting is disabled in Adjust Sample Rate.

Resolution This setting is disabled in Adjust Sample Rate

Convert Sample Type

This function converts the sample rate, bit rate, and channel format of an audio file that's currently loaded in Adobe Audition into a new format type (such as 44 kHz/16-bit/stereo to 22 kHz/8-bit/mono). Unlike Edit > Adjust Sample Rate, Convert Sample Type directly processes the samples within the file, or re-samples the data, so that the audio retains the same pitch and duration as the original file.



Convert Sample Type Dialog

You can choose various levels of conversion quality. Changes in overall and relative L/R volume levels may be easily made when converting between mono and stereo formats. The Dither option allows you to reduce granulation noise when converting to lower bit rates. Various other options are available, letting you adjust the conversion process to suit your own particular needs. The following are the major components of the Convert Sample Type:

Sample Rate Conversion Presets Choose any saved presets that you've created from this drop-down list. Have a sample type that you convert to often? Use the Save As... button to add those settings to the Sample Rate Conversion Presets list. Press the Delete button to remove highlighted entries from the list.

Sample Rate The sampling rate you choose determines the overall bandwidth (how many frequencies can be encoded within the audio signal). Higher sampling rates generally yield better-quality waveforms. Enter any sample rate directly, or choose a common sample rate from the list. Keep in mind that most sound cards support only certain sample rates.

Low/High Quality Use this slider to adjust the quality of the sampling conversion.

Higher values retain more high frequencies while still preventing aliasing of higher frequencies to lower ones, but the conversion process takes longer. A lower quality setting requires less processing time, but results in certain high frequencies being “rolled-off”, leading to muffled-sounding audio. Usually values between 100 and 400 do a fine job for most conversion needs.



You should use high-quality settings whenever you downsample from a high sample rate to a low rate. When upsampling, a low-quality setting will sound almost identical to a high-quality setting.

Pre/Post Filter In general, you should enable this option for best results. During downsampling and upsampling, the appropriate filter will help keep false frequencies from being generated at the low end of the audio spectrum.

Channels Choose whether the new sample type will be Mono or Stereo by pushing the desired button. When you convert to Stereo from Mono or vice-versa, you’re given the option to adjust the mix of the left and right channels.

When converting from Mono to Stereo, you can choose the relative amplitude (measured in percentage) with which the original mono signal will be placed into each side of the new stereo signal. This way, you can place the mono source on the left channel only, the right channel only, or any balance point in between.

If you’re converting from Stereo to Mono, this percentage controls the amount of signal from the respective channel that will be mixed into the final mono waveform. The most common mixing methods are to use 50% of both left and right channels when converting to Mono, and 100% for both values when converting to Stereo.



To remove all or most of the lead vocals from many stereo music recordings, you can convert a stereo waveform to mono with a Left Mix of 100% and a Right Mix of -100%. Most vocal tracks are positioned in the middle of the stereo field in-phase, so converting the signal so that it’s out-of-phase will often greatly reduce or eliminate the vocal track’s level.



When converting a stereo recording of monaural material (that is, the exact same contents are on both channels) to a Mono waveform, try setting either the Left Mix or Right Mix to 100% and the opposite channel to 0%. This often results in a cleaner sounding mono recording, especially if the original recording was made or played back on a tape recorder with improperly aligned heads.

Resolution Use this control to choose the bit resolution to convert to. When converting to a lower resolution, the Dither options are enabled.

Dither Use Dither to reduce noise and distortion to figures below their current levels through the addition of small amounts of white noise. Although a small amount of noise is introduced into the “circuit”, the result is far preferable to the increased distortion that you would otherwise be listening to at low signal levels.

Note: *When converting to lower resolutions, dithering allows you to hear sounds that would otherwise be masked by the noise and distortion limits of 8-bit audio (or quieter than the 16-bit limit provides when converting from 32-bit.)*

If Dither isn't checked, quiet audio passages will just fade in and out of a disruptive, "choppy" sound that resembles falling rain or static. Whether or not dithering is used depends on the audio that's being converted and, of course, your preferences.

Enable Dithering Check this box to activate Dithering; uncheck the box to turn Dithering off.

Note: If Dithering isn't enabled, Adobe Audition truncates instead. This means that bits that aren't used are simply chopped off and discarded. This gives a "crackly" effect that fades in and out on very quiet audio passages.

Dither Depth (bits) This sets the bit amount of Dither to be applied. Generally, values of 0.2 to 0.7 give the best results without adding too much noise. Note, however, that as this value is lowered, other unwanted harmonic distortion noise will appear. (You can usually get away with lower values for the Dither Depth when you use Noise Shaping.)

p.d.f.: The "probability distribution function" controls how the dithered noise is distributed away from the original audio sample value. Five p.d.f. types are available from the drop-down list: Rectangular, Triangular, Gaussian, Shaped Triangular, and Shaped Gaussian.

The Rectangular function means there's an equal chance that the noise value added will lie anywhere between +1 and -1 (i.e., the likelihood of a value of -0.8 being chosen is the same as that of 0.2, or 0.3, or any value between -1 and +1).

The Triangular function chooses random numbers that are generally closer to 0 than to the edges -1 or +1 (i.e., the chance of 0 being chosen is twice as great as the chance of choosing 0.5 or -0.5). All dithering distribution functions can linearize the quantization noise, meaning that the noise that's heard doesn't depend on the frequency of the dithered audio, thus no harmonic distortion appears. The SNR Loss is measured against the undithered case. If modulation is present, the audible noise floor rises and falls depending on the amplitude of the signal, so generally one doesn't want this either.

p.d.f.	SNR Loss	Modulation
Rectangular	3dB	Yes
Triangular	4.8dB	No
Gaussian	6dB	Negligible
Shaped Triangular	4.8dB	No
Shaped Gaussian	6dB	Negligible

Usually a Triangular p.d.f. function is a wise choice, because it gives the best tradeoff between SNR, distortion, and noise modulation.

Noise Shaping Select one of the several noise shaping presets (known as "noise shaping curves") to have Adobe Audition determine the placement when moving noise to different frequencies. The same amount of noise overall is present, but you can choose less noise to be placed at one frequency at the expense of placing more noise at another. You may also specify that no noise shaping is used.

Note: Different curves result in different types of background noise. The type of curve to use depends on the source audio, final sample rate, and bit resolution. By introducing noise shaping, you may be able to get away with lower Dither depths to reduce the overall background noise level, without introducing a lot of unwanted harmonic noise.

Curve	Sample Rate
Noise Shaping A	44.1 kHz or 48 kHz
Noise Shaping B	44.1 kHz or 48 kHz
Noise Shaping C1	44.1 kHz or 48 kHz
Noise Shaping C2	44.1 kHz or 48 kHz
Noise Shaping C3	44.1 kHz or 48 kHz
Noise Shaping D	44.1 kHz or 48 kHz
Noise Shaping E	44.1 kHz or 48 kHz
Noise Shaping E2	44.1 kHz or 48 kHz
44.1KHZ	44.1 kHz
48KHZ	48 kHz
96KHZ	96 kHz

Note: In general, there are no really good noise shaping curves for audio at 32kHz or lower. With audio at those sampling frequencies, feel free to try the different curves to see if they help at all, and just choose the one that sounds the best.

Chapter 7: View Menu – Edit View

The View Menu displays options that relate to changing the display or viewable area when you're in the Edit View Mode. For many of the items, a checkmark is shown next to their menu entries to designate that they're active.

Multitrack View

Access this command to go to Adobe Audition's Multitrack View, where you can digitally mix up to 128 tracks to one or more sound cards.

 *Pressing the leftmost button in the Edit View's File toolbar takes you to Multitrack View mode. The face of the View button represents the target View that the system will "jump to" when pressed. For example, when you're working in Edit View, this button's appearance changes to let you know that if you push the button, you'll go to the Multitrack View.*

 *You can also use the F12 key to quickly switch between Multitrack and Edit View.*

Waveform View

Adobe Audition offers two ways to visualize waveform data. Waveform View is the default. Data is shown as vertical spikes that represent relative volume, while time is visualized horizontally.

Spectral View

Spectral View is the other method Adobe Audition utilizes to represent waveform data. Spectral View makes it easy to view the frequency components to an audio file over time. This mode is handy for spotting which frequencies are most prevalent throughout the waveform. The greater a signal's amplitude component within a specific frequency range, the brighter the displayed color will be.

Colors range from dark blue (which means that next to no amplitude components exist in this frequency range) to bright yellow (frequencies in this range are high in amplitude.) Lower frequencies are displayed near the bottom of the display, while higher frequencies are visualized from the middle-to-top part of the scale.

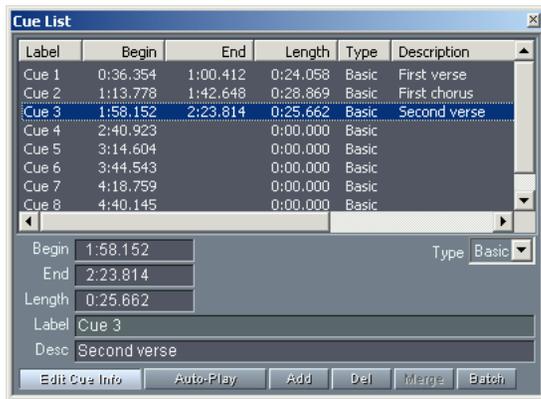
Show Organizer Window

This command lets you see Adobe Audition's Organizer Window. This handy, tabbed window allows you to easily open and close files, quickly send waveforms to the Multitrack Display, choose effects with ease, and more.

Note: *More information about the Organizer Window can be found in the "Looking at the Work Area" chapter.*

Show Cue List

The command opens the Cue List window, which is a dockable window. This means you can reposition it at one of several places in the Adobe Audition interface. Or, you can even detach the Play List window and have it float above Adobe Audition’s main window.



Cue List Dialog with Edit Cue Info Mode Active

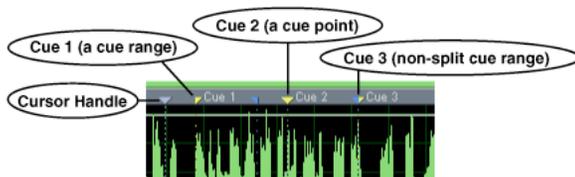
An unlimited number of cues may be defined and saved in the Cue List for later recall, or for assembling in Adobe Audition’s Play List.

To close the Cue List window when it’s floating, click on the “X” control in its title bar. To get rid of the Cue List window when it’s docked, right-click on its grab bar and select Close from the pop-up menu. Or, you can simply uncheck the Cue Play List item on the Edit menu to close it in either docking or floating mode.

About Cues Cues are time offsets or locations defined within a waveform. They may be thought of as virtual markers.

Adobe Audition has four types of Cues. Each type can be a point or a range.

- A point refers to an exact cursor position within a waveform (for instance, 1:08.566 from the start of the wave).
- A range has both a start time and an end time (for example, all of the waveform from 1:08.566 to 3:07.379). If a Cue is a range, you can drag its beginning and end points to different times.



The upper part of the Wave Display showing the Cursor Handle and three Cue Handles

Here are the four cue types:

- **Basic:** Basic Cues are used to mark important sections of the waveform for later reference (for example, to remind yourself of an editing point). They're also used to specify stop and start positions for the Adobe Audition's Play List.
- **Beat:** Beat Cues are just like Basic Cues, but they're specifically for marking musical beats. If you use the Find Beats and Mark part of the Auto-Cue feature (Edit > Auto-Cue), Adobe Audition will mark each beat with a Beat Cue.
- **Track:** Track Cues are only used for burning CDs. They indicate a split in tracks for an audio compact disc.
- **Index:** Index Cues are also just used for CD burning. Indexes are markers within a CD track. (Some CD players offer controls for cueing indexes.) Also, the time between the track marker that begins a track and the first index in that track will show up on the player as "negative time".

All four Cue types can be ranges as well as points, although it doesn't really make sense for indexes to be ranges.

***Note:** Cues are preserved in the waveform when the file is saved, but only if the Save extra non-audio information box is checked in Adobe Audition's Save Waveform As dialog.*

Creating Cues and Ranges

To add a specific point in a waveform (say, three seconds from the beginning of the waveform) to the Cue List, left-click on the waveform at the exact place where you wish the cue to be. Then, do one of the following:

- Click on the Add button in the Cue List window.
- Press F8 to add a Basic cue, Shift+F8 to add a Track cue, or Ctrl+F8 to add an Index cue.
- Right-click on the Wave Display and choose Add to Cue List.
- Right-click on the Cursor Handle (the triangle at the top and bottom of the Wave Display directly over and under the play cursor) and choose Insert in Cue List, Insert CD Track Marker, or Insert CD Index Marker.

To add a specified range (for instance, all the audio from two seconds to five seconds, as measured from the first part of the waveform), just highlight the range in the Wave Display and do one of the following:

- Click on the Add button in the Cue List window.
- Press F8 to add a Basic cue, Shift+F8 to add a Track cue, or Ctrl+F8 to add an Index cue.
- Right-click on the Wave Display and choose Add to Cue List.
- Right-click on the Cursor Handle (the triangle at the top and bottom of the Wave Display directly over and under the play cursor) and choose Insert in Cue List, Insert CD Track Marker, or Insert CD Index Marker.



The easiest way to make a range is to highlight a selection in the waveform. Then add a cue, either through the Cue List or by pushing F8 (for a Basic Cue), Shift+F8 (for a Track Cue), or Ctrl+F8 (for an Index Cue).

Cue Handle Context Menu Right-click on a Cue Handle (a Cue-named triangle at the top and bottom of the Wave Display) to display the Cue Handle context menu.



The Cue Handle context menu

The Cue Handle context menu has these commands:

- **Go To Cue List:** Accessing this command displays the Cue List, or if the Cue List is already displayed, makes it the active window.
- **Change Cue Type:** Use this entry and its submenu to change a Cue from one type to another. A checkmark designates the current Cue type.
- **Make Range:** If the Cue is already a point cue, this command turns it into a range cue. The Cue Handle is split into two handles, one yellow and one blue. Drag either or both to set the beginning and ending times of the range.

Note: Adobe Audition automatically makes the second point of a cue range the end, even if you move the ranges around.

- **Make Point:** If the Cue is a range cue, this command turns it into a point Cue. The two parts of the range Cue Handles are merged into a point Cue Handle, with the start time of the range becoming the time for the point Cue.
- **Delete:** This command deletes the Cue, removing it from both the Cue List and the Wave Display.

Here are the major components of the Cue List window:

Column Bar The column bar runs along the top of the Cue List window. It behaves similarly to the column bar in a Windows Explorer window. Along with telling you the type of data in each column (Label, End, Begin, etc.), you can click on a column header to sort the data in that column. For instance, a left-click on the Length column header will place all of the items in the Cue List from shortest to longest. You may also select and slide a column header horizontally to rearrange the sequence of column headers.

Edit Cue Info Button When you press this button, fields used to modify the information in the highlighted cue appear: Begin (the cue's starting time), End (the cue's ending time), Length (the cue's length), Label (the cue's name), and Desc (the cue's description).

Left-click on the desired field to make it active, and then type in the new time or name. You can also use the drop-down list to pick the type of cue: Basic, Beat, Track, or Index.

To hide the cue info fields, press the Edit Cue Info button once more.

Auto-Play If the Auto-Play button is depressed, the cue will automatically play whenever the cue is selected in the Cue List. To deactivate the automatic play mode, click on the Auto-Play button to release it.

Add Press this button to add a Basic cue – either an exact point or a specified range – to the Cue List. You can also press F8 to add a Basic cue, Shift+F8 to add a Track cue, or Ctrl+F8 to add an Index cue to the Cue List.

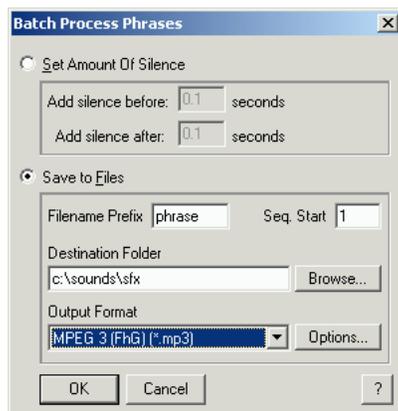
Del This button deletes all of the highlighted items in the Cue List. Just as in a Windows Explorer window, you can depress your Shift key as you click to select and highlight multiple contiguous files, or hold down the Ctrl key as you click to select and highlight multiple non-contiguous files.

Merge Click on the Merge button to combine highlighted entries in the Cue List into a single, combined cue that spans the earliest and latest locations in the two cues. Keep in mind that you're limited to merging just two ranges, but there are no limitations on the number of cue points you can measure.

Hold down your Shift key as you click to select and highlight multiple contiguous files, or depress the Ctrl key as you click to select and highlight multiple non-contiguous files.

Note: The new merged item inherits the name from the earliest item chosen. You lose the information entered in the Label and Desc fields for the subsequent merged items.

Batch If one or more Cue List range entries are highlighted, pressing this button brings up the Batch Process Phrases dialog.



The Cue List's Batch Process Phrases Dialog

Note: At least one of the cues must be a cue range in order for the Batch button to be active.

Here are the components of the Batch Process Phrases dialog:

Set Amount Of Silence Select this item to add silence between cue points in the currently shown waveform. Fill in the desired values (measured in seconds) in the Add silence before: and Add silence after: fields. Press the OK button to initiate the silence-adding process.

Save to Files Use this option if you'd like to split the audio between cue points in the active waveform to new files.

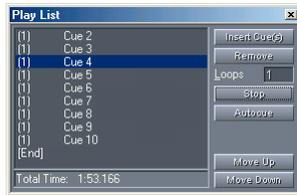
Filename Prefix Enter the desired Filename Prefix (such as "phrase") in the field provided. Adobe Audition automatically adds numbers after the prefix (phrase02, phrase03, etc.) as well as the correct extension based upon the output format you've chosen.

Destination Folder This is the folder where Adobe Audition will place your new “split” files. Press the Browse... button to open the Choose Destination Folder window.

Output Format Use this drop-down control to select the desired output format. If the selected format has options, the Options... button will be enabled, allowing you to select them.

Show Play List

Select this item to display Adobe Audition’s Play List window. The Play List is an arrangement of Cue List entries that you can play back in any order and loop a specified number of times. The Play List can include an unlimited number of entries.



Play List Dialog

The Play List window is a dockable window, allowing you to reposition it at one of several places in the Adobe Audition interface. Or, you can detach the Play List window and have it float above Adobe Audition’s main window.

To close the Play List window when it’s floating, click on the “X” control in its title bar. To get rid of the Play List window when it’s docked, right-click on its grab bar and select Close from the pop-up menu. Or, you can simply uncheck the Show Play List item on the Edit menu to close it in either docking or floating mode.

Here are the major components of the Play List window:

Show Cue List Click on this button to display Adobe Audition’s Cue List window. The Show Cue List button is then replaced by the Insert Cue(s) button.

Insert Cue(s) Press this button to add the currently highlighted selection(s) from the Cue List to the Play List. The selection is inserted before the currently highlighted Play List item, or at the end, if nothing is selected.

Note: The cues must be cue ranges, not just cue points, for this to work.

Remove Click on this button to remove the selected Play List item.

Loops In this field, enter the number of times you wish the selected Play List item to loop (play again). Each item in the Play List can loop a different number of times.

Play Pressing this button plays the cue ranges in the order listed, looping selections if specified. Play begins at the currently highlighted item in the Play List and goes to [End].

During playback the Play button turns into a Stop button.

Autocue This control plays the currently highlighted item in the Play List (or the first item if nothing is highlighted), and stops on the next item in the Play List. Thus, the next item in the Play List will be played every time Autocue is pressed.

Move Up Push this button to move the highlighted item in the Play List up one position.

Move Down Click on this button to move the highlighted item in the Play List down one position.

Show Transport Buttons

Check this item to display Adobe Audition's Transport Buttons, which are used to play, stop, pause, rewind, and fast forward the active waveform or session. The Transport Buttons window is dockable.

Note: More information on the Transport Buttons can be found in the "Looking at the Work Area" chapter.

Show Zoom Buttons

Select this item to display Adobe Audition's Zoom Buttons. The Zoom Buttons window is a dockable window, allowing you to reposition it at one of several places in the Adobe Audition interface.

Note: More information on the Zoom Buttons can be found in the "Looking at the Work Area" chapter.

Show Time Window

Select this item to show Adobe Audition's Time Window. This dockable window displays the current position of the playback cursor. The default display value is mm:ss:ddd (minutes:seconds:thousandths of a second), but this can easily be changed by right-clicking on the Time Window and choosing another option, such as Compact Disc, Samples, Bars and Beats, or one of several SMPTE values.

Note: More information on the Time Window can be found in the "Looking at the Work Area" chapter.

Show Sel/View Controls

When checked, the Sel/View Controls window is displayed. ("Sel" is an abbreviation for Selection.) This window shows the beginning and ending points, as well as the total length of both the selection and the section of the waveform that's currently viewed. The selection and display range are each shown using the current time display format.

Note: More information on the Sel/View Controls window (which is dockable) is in the "Looking at the Work Area" chapter.

Show Level Meters

Click on this option to display Adobe Audition's Level Meters, which appear by default near the bottom of both Adobe Audition's Edit View and Multitrack View windows. These meters display both record and playback levels. The Level Meters window is dockable.

Note: More information on the Level Meters can be found in the “Looking at the Work Area” chapter.

Show a Placekeeper

Select Show a Placekeeper to create a blank, dockable window called a “placekeeper”. Use it to more accurately define where you want a dockable window to be, especially for windows that are only useful when viewed in a certain aspect ratio.

Note: More information on Placekeepers can be found in the “Looking at the Work Area” chapter.

Display Time Format

The Display Time Format menu lets you choose between several time display readouts, including Decimal, Compact Disc, different versions of SMPTE, Samples, Bars and Beats, and Custom time code frame rates. Adobe Audition will then use the format chosen here when referencing time: the Time Window, the x-axis (time) ruler, time status boxes, Status Bar, etc.



Selectively cycle through the available time formats by double-clicking on the waveform ruler. In addition, right-clicking on the horizontal time waveform ruler or the main time display will pop-up a window that allows you to select between formats.

Decimal (mm:ss:ddd) This display type numerically indicates time in an mm:ss:ddd (minutes, seconds, decimal) format.

Compact Disc 75 fps This time format used is the same one utilized by audio compact discs, where each second equals 75 frames.

SMPTE 30 fps Time is displayed in the SMPTE format where 30 frames equal one second.

SMPTE Drop (29.97 fps) This display type numerically indicates time in the SMPTE Drop format of 29.97 frames-per-second.

SMPTE 29.97 Use this option to display time in the SMPTE Non-Drop format of 29.97 frames-per-second.

SMPTE 25 fps (EBU) Time is displayed using the standard European frame rate standard of 25 frames per second.

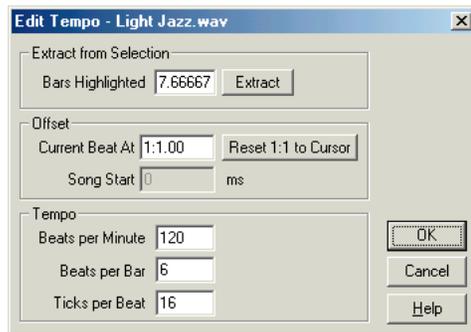
SMPTE 24 fps (Film) Use this SMPTE time format when working with film, where 24 frames equal one second.

Samples This display type indicates time numerically using the actual number of samples that have passed since the beginning of the edited file as a reference.

Bars and Beats This displays time in a musical measures format of bar:beat:ticks per beat. You can adjust the number of ticks per beat, tempo, and other properties in the Edit Tempo... dialog, which is described below.

Custom (XX frames/sec) This option will default to the time code setting that was last selected in View > Display Time > Define Custom Frames... For example, if a custom time code setting of 15 frames-per-second was chosen in a previous session, this option would show as “Custom (15 frames/sec)”.

Edit Tempo This command brings up the Edit Tempo window. Use this function to calculate the tempo or BPM (Beats Per Minute) value based on the currently highlighted selection. Adobe Audition uses the BPM value in displaying ruler information for the Bars and Beats time format. Also, any settings that follow the base time format will work in Bars and Beats mode as well (like Viewing Range).



Edit Tempo Dialog

Here are the major components of the Edit Tempo window:

- **Extract from Selection:** The number entered in the Bars Highlighted field specifies the number of bars highlighted in a selection, according to the Bars and Beats time format. If this number is incorrect (which it probably will be if you haven't yet defined the tempo), you can enter the correct value here to be used in extracting tempo information.
- **Extract:** Use this button to extract tempo information from the highlighted selection. Make sure you first correctly fill the Beats per Bar field below (4 for 4/4 music, 3 for 3/4, etc.), and enter the number of known highlighted bars. A push of the Extract button will then calculate the tempo and fill in the Beats per Minute and the Offset values.
- **Offset:** The information in the Current Beat At field is the bar/beat information for the left edge of the selection (or the current cursor point, if no selection is made). When extracting from a selection, Adobe Audition assumes that this is a downbeat. You can change the actual bar and beat number for the left edge of the selection (or the current cursor point). Modifying the Current Beat At figure updates the Song Start value based on the current tempo settings. Press the Reset 1:1 to Cursor button to change the value in the Current Beat At field to 1:1.00. Song Start is the number of milliseconds into the file before the measure 1:1.00 begins. You cannot edit this value.
- **Tempo**
 - Beats per Minute, as you'd expect, displays the number of beats that occur over a 1 minute interval. Calculate this value by using the Extract button.
 - Use the Beats per Bar field to assign the number of beats that form one measure. This number is usually the top number in the musical piece's time signature. For instance, enter 4 for 4/4 time, 6 for 6/8 time, or 3 for 3/4 time.
 - Ticks per Beat reflects the number of sections each beat is divided into, or the value displayed after the decimal point. You can set Ticks per Beat to any value from 2 to 3600. If you use 32 ticks for each beat, then a time setting of 4:2.16 would be the eighth note (half way) between beats 2 and 3, in common time.

- Define Custom Frames: Clicking on this option brings up the Adobe Audition Settings screen open to the General tab, where you may enter a new value (measured in frames/second) in the Custom Time Code Display field.

Vertical Scale Format

This menu lets you choose between several vertical scale formats for use in the y-axis vertical ruler, including Sample Values, Normalized Values, Percentage, and Decibels.

As with the Display Time Format, there are several ways you can change this format. Choose View > Display Scale Format from the menu, or you can selectively cycle through the available time formats by double left-clicking on the vertical scale waveform ruler. In addition, right-clicking on the vertical scale waveform ruler displays a pop-up window that will let you select between formats.

Note: When the wave display is set to Spectral View, the vertical scale is always in Hertz (Hz).

Sample Values This display type numerically indicates amplitude as the data's exact sample value of the data.

Normalized Values This display type numerically indicates amplitude on a normalized scale value that ranges from -1 to 1.

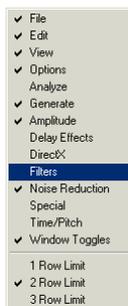
Percentage This display type numerically indicates amplitude on a percentage scale value that ranges from -100% to 100%.

Decibels This display type numerically indicates amplitude using a dB scale value that ranges from -Infinity to Zero.

Note: All of the displays can go to greater values for 32-bit audio if the audio goes above 0dB.

Toolbars

Use this menu to specify which, if any, groups of icons you'd like to display on Adobe Audition's Edit View toolbar. Checkmark the desired icon group(s) to display. If no items are checked, the toolbar area will disappear until you check at least one icon group.



Toolbars Menu

Options are also provided to limit how many rows of toolbar icons Adobe Audition displays at any one time: one row, two rows, or three rows.

Status Bar

Access the items on the Status Bar submenu to turn on Adobe Audition's status bar – which runs along the bottom of the Adobe Audition window – as well as decide which information fields you wish displayed there. You can also find this menu by right-clicking on the status bar.

To hide the Status Bar, right-click on it and select the Hide menu option. Or, uncheck the Show from the View > Status Bar submenu item.

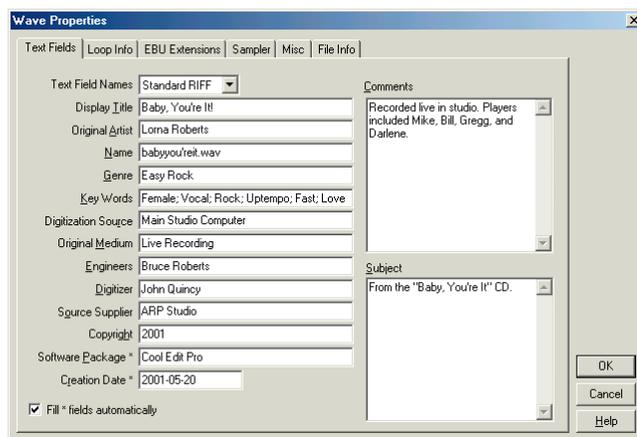
Note: More information on the Status Bar can be found in the “Looking at the Work Area” chapter.

Wave Properties

Access this command to bring up a tabbed window that lets you embed text and graphics, specify certain parameters, and get information about the active waveform.

Note: Make sure the Save extra non-audio information box is saved in Adobe Audition's Save Waveform As box in order for this additional data to be embedded in your waveforms.

Text Fields Tab Adobe Audition allows you to embed text-based data in Windows .wav files that use the RIFF LIST INFO and DISP type 1 formats, as well as standard ID3 tag information in .mp3 files. Provided that other audio editors support this information, this information remains with your audio file throughout its lifetime.



Wave Properties Dialog - Text Fields tab

Use the Text Field Names drop-down list to choose which field format you'd like Adobe Audition to display in the Text Field window: Standard RIFF, Radio Industry, or MP3 (ID3 Tag).

If you select Standard RIFF, these fields are available:

- **Display Title:** This should describe the sound, or if there are words in the waveform, the text. This field should be as short as possible, since it will be displayed in OLE objects and the like.
- **Original Artist:** Enter the name of the one who created the sound initially. Examples are: The Rolling Stones, George Jetson, or even your name.
- **Name:** The title of the waveform goes here.
- **Genre:** Specify the musical classification of the sound. For instance, you could put in Country, Rock, New Age, Voice, or Jazz.
- **Key Words:** If you plan to organize waveforms in software that supports searching off the Key Words field, then you'll want to include key words here. Key words should be separated by a semicolon followed by a space. For example: Guitar; Concert; Classic Rock; Eric Clapton...
- **Digitization Source:** Enter where the sound was digitized from, such as a cassette deck, a CD, an LP, or a microphone. You might want to describe the sound card used here as well.
- **Original Medium:** Describe where the sound came from originally. Examples: Live Band, Trumpet, Keyboard, Vocal...
- **Engineers:** This field is used to store the name(s) of the engineer(s) who worked on the file, or edited the file. Please separate names by a semicolon and a space. When someone new edits the file, they can add their name to the list.
- **Digitizer:** Who did the actual digitizing? They should put their name right here.
- **Source Supplier:** In this field, enter the name of the person or organization that supplied the original source material. You can use this space for the names of record companies or whoever supplied you with the source. Examples include Epic Records or, if recorded live, the name of the artist, such as Ann Wilson.
- **Copyright:** Any copyright information for this file should go here. Example: (c)2001 G. Willikers Corporation. All rights reserved.
- **Software Package:** The name of the software used to digitize and edit this file (such as Adobe Audition) goes here.
- **Creation Date:** This field should contain the date that the subject matter was created. The date should be in the format yyyy-mm-dd (year-month-day), using "0" as a place holder in single-digit values. For example, if the date the original recording was made was June 8, 2001, then it needs to be written as: 2001-06-08.
- **Fill * Fields Automatically:** If the Fill * Fields Automatically box is checked, the Software Package and Creation Date fields are automatically filled by Adobe Audition. If you don't want this extra information to be tagged with your wave files, simply un-check this box.
- **Comments:** This area is for making any comments you wish. Feel free to include any special effects or enhancements you made to any preexisting waves, so that the editing history can be tracked. Don't use line returns (i.e., don't hit the Enter key after completing a line), but end each sentence with a period. Sample entry: "It took 3 hours to get this recording right. Ted added reverb effects and noise reduction using Adobe Audition and it's ready to be mastered."

- **Subject:** Use this field to describe the contents of the file. Feel free to include a description of the instruments used, where someone can find the song recorded, etc. Line returns are OK, and are created by pressing Ctrl+J. Sometimes copyright information is placed here as well. For instance: “The shakuhachi of Japan.<Ctrl+J><Ctrl+J>The shakuhachi was developed in the 15th century from a Chinese end-blown flute, called the chiba.”

If you select Radio Industry, the fields in the Text Fields change to accommodate information for commercials and other types of audio files used by radio broadcasters. They include:

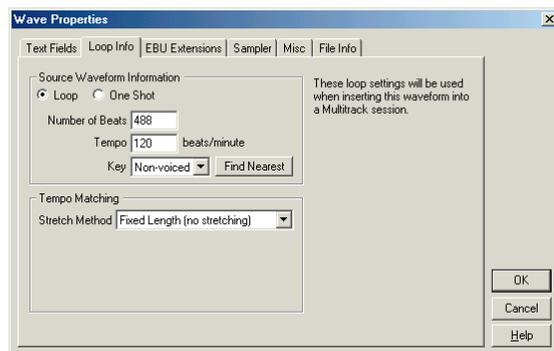
- **Description:** This is where you enter the name of the production, such as “Joe’s Chevrolet March Madness Sale” or “Kiss-FM Legal ID”.
- **Advertiser:** Place the name of the sponsor in the field. Examples include “First National Bank” or “In House Promo”.
- **Outcue:** A description of the ending (such as the last few words) of the production piece goes here. It might be “member FDIC”, “sale starts tomorrow”, or “(music up and cold)”.
- **Start Date:** In this field, type in the date the audio production is scheduled to begin airing.
- **End Date:** This is where you enter the date the audio production is supposed to end its run.
- **Intro Time (ms/begin):** The production piece’s introduction time, measured in milliseconds from the beginning of the audio file, goes here.
- **Sec Tone (ms/end):** Enter the time, measured in milliseconds from the ending of the audio file, where the secondary tone (a.k.a. the “trip tone”) appears. (This tone is used to automatically start the next audio element when used in an automation system.)
- **Producer:** Who produced this piece of audio? The name of the producer goes here.
- **Talent:** Enter the name of the person(s) who voiced this audio production, such as “Joe Jock”, “Vicki Voiceover”, or “Bill Baritone & Susan Soprano”.
- **Category:** This field is for the category this audio production belongs in. Examples include, “Auto Dealer”, “Financial Institution”, “Promo”, and “Stager”.
- **Agency:** Here’s where you enter the name of the advertising agency or production house.
- **Account Executive:** The name of the salesperson in charge of the advertiser’s account goes here.
- **Creation Date:** This field should contain the date that the subject matter was created. The date should be in the format yyyy-mm-dd (year-month-day), using “0” as a place holder in single-digit values. For example, if the date the original recording was made was June 8, 2001, then it needs to be written as: 2001-06-08.
- **Fill * Fields Automatically:** If the Fill * Fields Automatically box is checked, the Creation Date fields is automatically filled by Adobe Audition. If you don’t want this extra information to be tagged with your wave files, simply un-check this box.

- **Comments:** This area is for making any comments you wish. Feel free to include the name of any music, sound effects, or enhancements you made to any preexisting waves, so that the editing history can be tracked. Don't use line returns (i.e., don't hit the Enter key after completing a line), but end each sentence with a period. Sample entry: "Music came from Music Library Disc 17, cut 3, while the doorbell SFX came from Sound Effects Library Disc 3, cut 57. Compression and reverb were added using Adobe Audition."
- **Copy:** Enter the copy for the production here. You can type in the copy directly, or use Ctrl+V to paste in text from a Windows-based program such as a word processor or text editor via the Windows clipboard.

Selecting the MP3 (ID3 Tag) item from the Text Field drop-down menu activates these text fields, allowing you to tag the loaded MP3 file with ID3v1.1-compatible data:

- **Song Title:** The title of the musical composition (up to 30 characters) goes in this field.
- **Artist:** Enter the name(s) of the performer(s) here, up to 30 characters.
- **Album Name:** The title of the album is typed into this field. This title can have up to 30 characters.
- **Genre:** Select the musical category from the drop-down list of dozens of alphabetically-sorted choices.
- **Year:** The year (up to 4 digits) the song was recorded or released goes here.
- **Fill * Fields Automatically:** When this box is checked, the current year is automatically entered into the Year field.
- **Comments:** Enter any general text about the recording here, up to 30 characters.
- **Track Number:** If the recording came from an album, type in the number (up to two characters) of the track in this field.

Loop Info Tab



Wave Properties Dialog – Loop Info tab

Source Waveform Information

Loop Select this option to tell Adobe Audition that the file is a loop. If the file is inserted into Adobe Audition's Multitrack view, it will automatically have looping turned on for that block.

One Shot Choose this option if the file isn't a loop.

Number of Beats Enter the waveform's number of beats in this field. When you do so, the BPM (Beats Per Minute) is automatically calculated and entered into the Tempo box below.

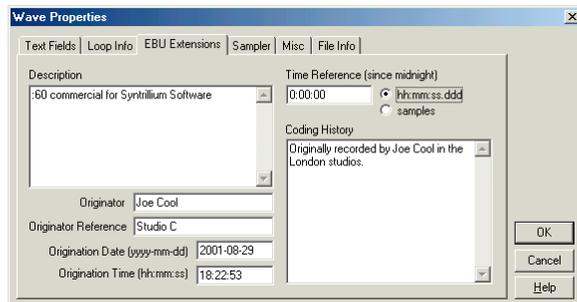
Tempo This is the number of beats per minute in the waveform. Enter this figure directly, or when you fill in the Number of Beats field above, the Tempo value is automatically calculated.

Key Use this drop-down menu to enter the waveform's key. This will cause the waveform to match pitch with a session that has a key specified when inserted into Adobe Audition's Multitrack View. Don't know the key? Press the Find Nearest button and Adobe Audition will scan the waveform to locate the closest match.

Tempo Matching Stretch Method Tempo matching determines how (if at all) the loop will stretch to match the session's tempo. Four options are provided:

- If the Fixed Length option is chosen, the file won't be stretched to match tempo.
- Select the Time-Scale Stretch option to stretch the file (just like Adobe Audition's Stretch effect) to match the tempo of the session. Options are provided for Quality (High, Medium, or Low), Frame Size (the number of splices per beat), and the percentage of Frame Overlapping.
- Resample will cause Adobe Audition to resample the loop to match the session's tempo, and as such, the pitch will be affected. High, Medium, and Low Quality options are available.
- The Beat Splice option actually tries to cut up the file at the beats to loop the file that way. This only works on loops that have very sharp (short duration) sounds, like drums. If the waveform already has beat markers, you may elect to use them. Otherwise, the Auto-Find beats option will be selected. If the default values of 10dB and 9 milliseconds aren't satisfactory, you can change them in the fields provided.

EBU Extensions Tab



Wave Properties Dialog – EBU Extensions tab

Description Up to 256 characters may be entered in this field to describe the audio file.

Originator Enter the name of the audio file's producer (up to 32 characters) in this field.

Originator Reference You can enter up to 32 characters of reference information here about the producer.

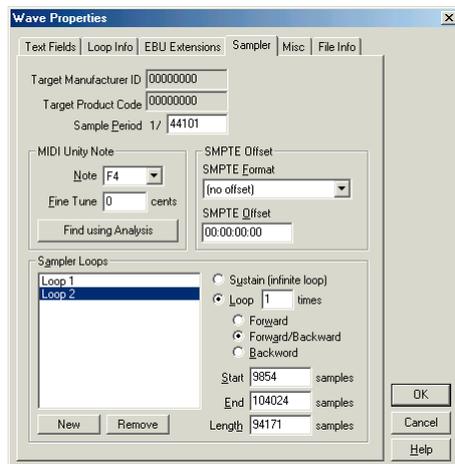
Origination Date (yyyy-mm-dd) This field should contain the date that the subject matter was created. The date should be in the year-month-day format, using “0” as a place holder in single-digit values. For example, if the date the original recording was made was June 8, 2001, then it needs to be written as: 2001-06-08.

Origination Time (hh:mm:ss) The time the audio file was produced goes in this field. The format is hour:minutes:seconds, with the hour represented in 24-hour (0 to 23) “military time” format (i.e., 10 p.m. is 22).

Time Reference (since midnight) This field contains the time code of the audio file, calculated since 12:00 a.m. Select from “hh:mm:ss.ddd” (hour:minutes:seconds:thousandths of seconds) or samples options.

Coding History This field is used to contain a description of all coding processes applied to the waveform. Adobe Audition doesn’t automatically add any information in this field. However, you’re welcome to enter a processing history (or any other text-based information) here manually.

Sampler Tab



Wave Properties Dialog – Sampler tab

Technical information relating to other devices, systems, or programs (such as synthesizer uploading and downloading software) can be directly imbedded within your .wav files using the View > Wave Properties > Sampler information window. As with the information in the View > Wave Properties > Text Fields window, this information should stay with your audio file through its lifetime.

Here are the major components of the Sampler window:

Target Manufacturer ID You can’t edit this information, but this field will show a proper value if a sampler has written a .wav file with this sampler information chunk present.

Target Product Code You can't edit this information, but this field will show a proper value if a sampler has written a .wav file with this sampler information chunk present.

Sample Period The sample rate (or within 1Hz of it) is automatically placed into this box. You can change the text in this field if you wish the sampler to think the data is at a different rate than it actually is.

Note: Use this control to select this "base" or "root" note on a sampler that the current audio file is to be assigned to. The audio file's original pitch will be preserved whenever this key is played on a sampler.

Fine Tune The actual tone can be entered in this text box in values as precise as 1/100th of a cent. Enter the number of cents above the Note that the tone actually is.

Find using Analysis Press this button to have Adobe Audition analyze the audio file to automatically determine the Note and Fine Tune values.

- If a sampler loop is selected in the Sampler Loops list, the frequency at the center of that loop is entered into this field for the Note and Fine Tune fields.
- If no loops are selected, the center of the entire waveform is used to gain the current note.

Note: This value can be off by a few hundredths of a cent, so manual adjustment after finding the note may be necessary. For instance, you may get G#4 at 99.99 cents, which would probably really be A4 and 0 cents.

SMPTE Format This field is used to enter in the SMPTE frame rate format for the currently loaded sample.

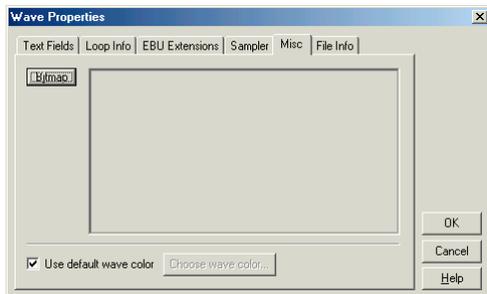
SMPTE Offset Use this field to enter the SMPTE trigger offset point for the currently loaded sample.

Example: An audio file for a film soundtrack that would need to be triggered at 45 minutes, 15 seconds and 29 frames, might have a frame rate setting of 30 frames-per-second with an offset of 00:45:14:29.

Sampler Loops New loops can be added by first highlighting the range area, and then going to this dialog and pressing the New button. If no area is highlighted, New can still be pressed to add the entire waveform as a new loop. The actual starting point, ending point or length may also be entered in directly by hand in the appropriate fields.

Samplers can usually play loops forward, backward, or back and forth and back again. Each loop can be looped a different number of times, or infinitely (as with a sustain loop, and the infinite loop would decay once the synth key is released). Currently this information is only saved in .wav files.

Misc Tab



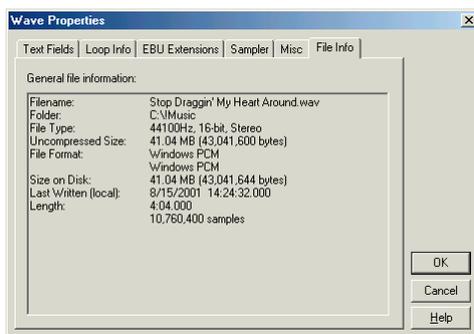
Wave Properties Dialog – Misc tab

Press the Bitmap button in this window to browse your system for a .bmp or .dib file that can be used to represent the audio file when it's displayed in OLE-compatible applications and in Explorer. For best results, choose a 32x32 pixel image.

If the Use default wave color box is checked, then the same wave color used for your current color scheme (the color of an unselected waveform in the Edit View wave display) is used.

Want to use another color? Uncheck the Use default wave color box and the Waveform Foreground Color dialog appears to let you select another color. (This box works similarly to the standard Windows color selector dialog.)

File Info Tab



Wave Properties Dialog – File Info tab

The File Info window displays non-editable file information about the active audio file. You can see the waveform's Filename, Folder, File Type, Uncompressed Size, File Format, Size on Disk, Date and Time Last Written, and Length.

Chapter 8: Effects Menu – Edit View

The Effects menu displays all the options that relate to waveform transformation (signal processing) functions when you're in Adobe Audition's Edit View mode. In other words, if you want to apply a “cool” effect to your audio, you'll be able to do it – and lots more – here.

Invert

This function simply inverts the waveform's samples, so that all positive offsets are negative and all negative offsets are positive. Inverting doesn't produce an audible effect, but it can be useful in lining up amplitude curves when creating loops or pasting. If applied on a stereo waveform, both channels are inverted.

Reverse

Select this command to have the current selection (or entire waveform, if no selection is made) play backwards by reversing the order of its samples. Reverse is useful for creating special effects.

Silence

This option silences out the selected range of the waveform. Unlike deleting or cutting a selection – which splices the surrounding material together – Silence leaves the duration of the selection intact, and simply zeros the amplitude within it.

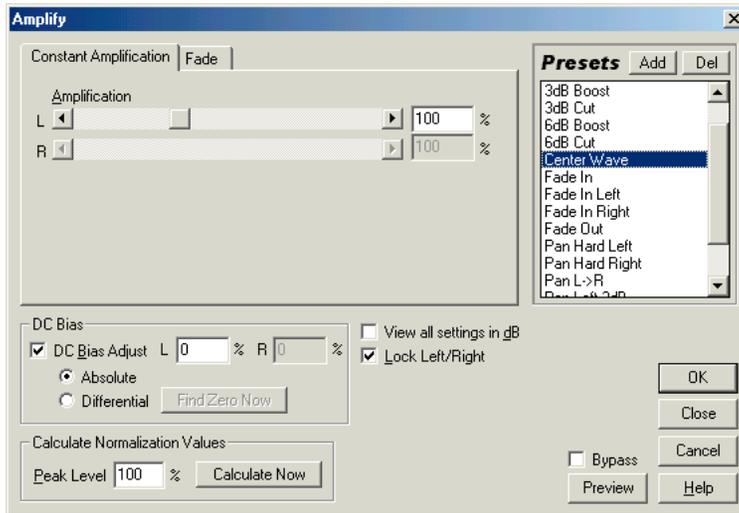
Note: You must select part or all of your waveform first in order for Silence to work.

Amplitude

Use the following commands in the submenu to alter the amplitude (volume level) of an audio file or selected range.

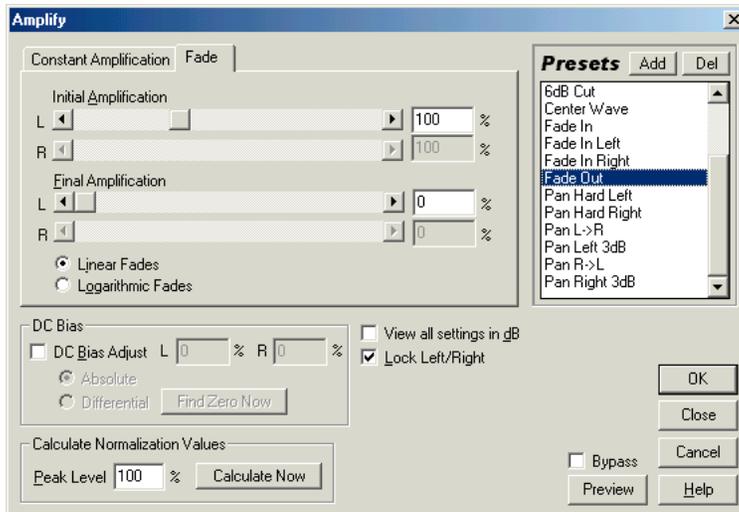
Amplify

The Amplify command provides the following options:



Amplify dialog – Constant Amplification tab

Amplification (Constant Amplification Tab) Move the slider(s) for the amount of volume change you wish to apply to the selection. Any figure over an amplification value of 100% or 0dB makes the selection louder; a figure below 100% or 0dB reduces the volume.



Amplify dialog – Fade tab

Initial Amplification (Fade Tab) Move the slider(s) for the amount of volume change you wish to apply to the beginning of the selection. Any figure over an amplification value of 100% or 0dB makes the selection louder; a figure below 100% or 0dB reduces the volume.



To achieve a fading in effect, choose an Initial Amplification value that's lower than that of the Final Amplification value.

Final Amplification (Fade Tab) Move the slider(s) for the amount of volume change you wish to apply to the end of a selection. Any figure over an amplification value of 100% or 0dB makes the selection louder; a figure below 100% or 0dB reduces the volume.



To achieve a fading out effect, choose a Final Amplification value that's lower than that of the Initial Amplification value.

Linear Fades (Fade Tab) Select this radio button if you want the waveform's sample values to be faded in an even, linear fashion, which will produce a smooth slope from beginning to end.

Logarithmic Fades (Fade Tab) This option tells Adobe Audition to use a logarithmic-style fade (also known as a "power fade"). This fades the amplitude of a signal at a constant rate, producing a steeper slope at one end of the fade (depending on whether you are fading in or out).

DC Bias Adjust Some recording hardware may introduce a DC bias, which results in the recorded waveform appearing to be above or below the normal center line in the Wave Display. Many waveform transformations require that the signal be centered, and checking the DC Bias Adjust box ensures new recordings are perfectly centered.

Choose the Absolute option to tell Adobe Audition the final DC value you want, as entered as a percentage in the L and R boxes. The main use for the option is to cancel out a DC that's not constant throughout the waveform. A super-low-cut filter is initiated to achieve this. Keep in mind that the actual amount adjusted varies with each sample. For instance, if you have a significant DC change in one area of the wave, at that boundary where the DC changes, the Absolute option will make all parts the same. (However, there will be a dip or peak right at the boundary point.)

Select the Differential option to analyze the entire selected area to get the DC offset, and adjust every sample by the inverse of that exact amount. The Find Zero Now button will be enabled. Press it to scan the waveform and automatically enter the correct L and R percentages.

To introduce a DC Bias by skewing the entire selected waveform above or below the center line, enter a positive or negative percentage. For example, a setting of 50% moves the entire waveform up halfway, and one of -50% moves it down halfway.

Calculate Normalization Values When you press the Calculate Now button, Adobe Audition scans the selection and automatically adjusts the amplification sliders to the volume level that will normalize the selection to the entered peak level. Click on the OK button to apply these adjustments.

Lock Left/Right With a stereo waveform loaded, the volume of the left and right channels may be individually adjusted. This lets you tweak the stereo balance or create cool panning effects. However, there are times when you want the amplitude of the left and right channels to be simultaneously lowered or raised. Check the Lock Left/Right box to make the scroll bars for the left and right channels to work in tandem and move together. Uncheck the Lock Left/Right box to adjust the channels separately.

View all Settings in dB When checked, amplification values are entered and shown in decibels; otherwise they're entered as a percentage of the original waveform.

Presets Adobe Audition comes with a nice selection of amplification presets including several fades and pans. More information on presets can be found in the “Looking at the Work Area” chapter.

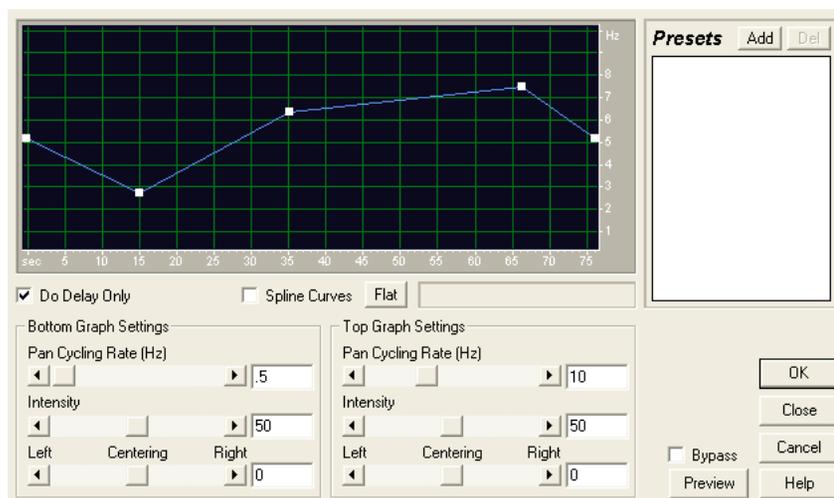
Preview Press the Preview button to audition the selected amplification setting before you apply it to the waveform. The selection is played over and over (“looped”) until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear amplification adjustments as the selection is looped. Check the Bypass box to hear the waveform without any changes in amplification for a quick “before and after” comparison.



Channel Mixer dialog

Binaural Auto-Panner

The Binaural Auto-Panner spatially locates the audio left and right, in a circular pattern over time. In order to spatially encode the signal, either the left or right channel is delayed so that the sounds will appear at each ear at different times, tricking the brain into thinking they are coming from either side.



Binaural Auto-Panner dialog

Frequency Graph Time is represented along the x-axis (horizontal edge), and frequency along the y-axis (vertical edge). As you go to the right of the graph, you are setting the frequency characteristics of the highlighted sample later and later in time. The settings chosen will vary between the low and high settings depending on where the graph dictates the signals should be. Choose the highest and lowest frequencies that are represented on the graph with the scroll bars located in Low/High Settings, below. The readout below the graph displays the current x, y position of your mouse.

Note: For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

Smooth Wave When checked, the audio that appears at the left and right channels will be smoothed out. The left and right channels will delay and un-delay following a smooth curve such that the delay difference between the left and right channels will follow a sine wave pattern, even though the brain will hear the audio as traveling around the head in a circle.

When Smooth Wave isn't checked, the net delays are the same, but are achieved by holding one channel constant (at no delay) while the other channel is delayed following half a sine wave. The boundary between holding non-delayed and delaying signal is discontinuous in that the dD/dt (difference in delay over time) jumps from zero to a positive delay value without hitting any values in-between. When Smooth Wave is checked, the dD/dt is always continuous, which will cause less noticeable distortion in either channel when heard independently.

Spline Curves Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use spline curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat Pressing the Flat button will reset the graph to its default state, removing all control points.

Low/High Settings There are two sets of settings for control of binaural frequencies. Low Settings all correspond to the lower part of the graph (points dragged near the bottom), and High Settings affect the top.

- Frequency: This is the binaural frequency that will be encoded into the final process.

 For special spatial panning effects, choose wave frequencies of 1Hz or less. A mono source (left and right the same) will appear to move from left to right and back at period of $1/\text{frequency}$. For example, a frequency of 0.1Hz will pan the audio in a "full circle" over the period of 10 seconds.

- Intensity: This controls the intensity of the binaural encoding. Higher intensities work well with lower binaural frequencies.
- Centering: You can trick the brain into thinking the binaural signal is coming from the left or right.

 Mixing a file that has been binaurally processed to the left with one that has been binaurally processed to the right (in the same frequency range within 2 Hz) has interesting effects.

Channel Mixer

The Channel Mixer enables you to alter the left/right balance of a stereo waveform. This lets you create new stereo mixes using the existing right and left channels as input sources. By recombining and inverting the channels, you can create some interesting stereo imaging effects. The default values are automatically set so that the waveform's original gain values will be unchanged. (In other words, the Channel Mixer can't make the waveform any louder than it already is.)

New Left Channel Adjust the slide bars to determine the percentage of the current left and right channels that will be mixed into the new left channel. For example, choose an L value of 50 and an R value of 50 to sum equally the current L and R channels to the new left channel after mixing. Enter an L value of 0 and a R value of 100 to make the new left channel contain the material of the original right channel.

New Right Channel Adjust the slide bars to determine the percentage of the current left and right channels that will be mixed into the new right channel. For instance, select an L value of 50 and a R value of 50 to sum equally the current L and R channels to the new right channel after mixing. Enter an L value of 100 and a R value of 0 to make the new right channel contain the material of the original left channel.

Invert Check the Invert box for either channel to invert that channel's phase polarity (i.e., the peaks will become valleys, and valleys will become peaks). Whenever both channels are inverted, there will be no perceived difference in sound. However, inverting only one channel places the channels out-of-phase and greatly changes the sound when played back (whereby the previously in-phase information is canceled out).

Presets Adobe Audition comes several Channel Mixer presets. More information on presets can be found in the "Looking at the Work Area" chapter.

Preview Press the Preview button to audition the selected Channel Mixer setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear Channel Mixer adjustments as the selection is continuously played. Check the Bypass box to hear the waveform without any changes in channel mixing for a quick “before and after” comparison.

 Try using the Channel Mixer's Vocal Cut preset to create your own karaoke backing tracks. It should remove most (if not all) of the lead vocal from many contemporary stereo music recordings.

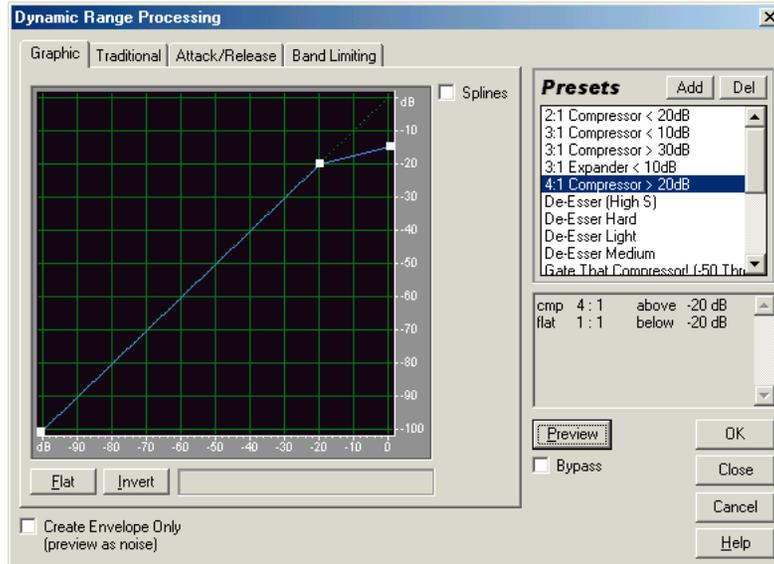
Dynamics Processing

The dynamics processor varies the output level of a waveform, based on its input level. This lets you limit or compress the dynamic range of a sample so that the perceived loudness is kept below a defined limit, or so that the waveform's overall dynamic range is kept at roughly the same level.

You can also expand or gate the signal so that low-level signals are reduced in level, thereby increasing the perceived dynamic range, or so that signals that fall below a certain threshold (i.e., noise) are eliminated.

Control of the Dynamics Processor is carried out through any of the four tabbed sheets:

Graphic Tab



Dynamics Processing dialog – Graphic tab

Graph The graph depicts input level along the x-axis (left and right) and the new output level along the y-axis (up and down). A line that flows directly from the lower-left to the upper-right (default) depicts a signal that has been left untouched, since every input value goes to the exact matching output value. Adjusting the shape of this line will adjust the input or output assignments, thereby altering the dynamic range.

For example, you can boost all input that has a level of around -20dB, leaving everything else unchanged. You can also draw an inverse line (a line from upper-left to lower-right) that will dramatically boost low amplitudes while dramatically suppressing high amplitudes (that is, all quiet sounds will be loud, and all loud sounds will be quiet).

Note: For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

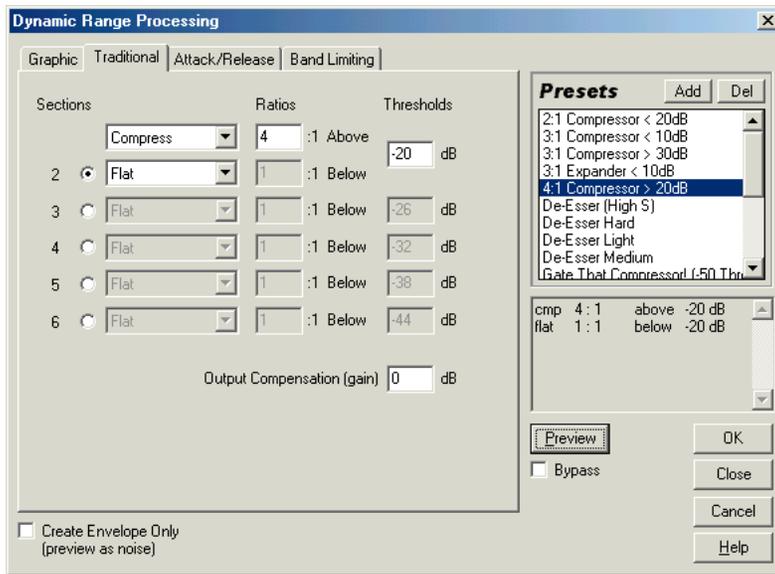
Splines Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use Splines, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat The Flat button resets the graph to its default state of an unchanged signal, removing all control points.

Invert The Invert button changes the graph to one that will function as the exact opposite. For example, if an effect function with a compressor characteristic is being displayed, pressing Invert changes the graph to one with the corresponding expander characteristic.

Note: For a graph to be invertible, it must have points in the two default corners (-100, -100 and 0, 0) and its output level must always increase from left to right (i.e., each control point must be higher than the one to its left).

Traditional Tab



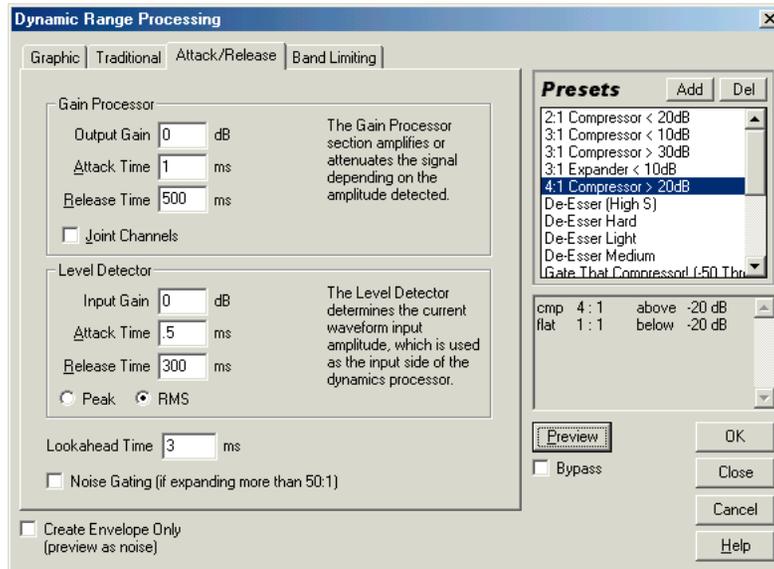
Dynamics Processing dialog – Traditional tab

The Traditional tab lets you enter ratios and thresholds numerically. You can input Compress, Flat, or Expand settings for up to six sections or stages, each with its own ratio and threshold setting. Simply enter the data to match the compressor you want. For example, to create a 3:1 compressor above -20dB, choose Compress from the drop-down list, enter 3 (for 3:1) in the Above line, and -20dB for the threshold. If you want to expand 2:1 below -20dB as well, choose Expand from the drop-down list, and enter 2 in the first Below line.

Note: Threshold settings must get lower in value as you move down the list.

Attack/Release Tab

Specify gain processing and level detection values using the fields in this area of the Dynamic Range Processing effect.



Dynamic Processing dialog – Attack/Release tab

Gain Processor The Gain Processor amplifies or attenuates the signal depending on the amplitude detected. Here are its components:

- Output Gain is a gain applied to the output signal. It's the last step performed on the audio.
- Attack Time is applied just before output. Attack time determines the time in milliseconds that it takes for the processed output signal to reach its specified output volume. If there's suddenly a quiet portion that drops 30dB, it will take "this much time" before the output actually drops to its corresponding volume level.
- Release Time is applied just before output. Release Time is the time it takes the end of a previous output level to reach the specified output volume. For example, where the Attack is the time it takes for the start of a pulse to reach the desired output volume, the Release is the time it takes for the end of the pulse to reach the desired level.

Note: If the sum of Attack and Release times is too short (less than about 20 milliseconds total), audible effects, such as a "vibrating" sound, can be heard at around 1000Hz/ms total. For example, if the Attack and Release times are each set to 5 ms (making 10 ms total), then a vibrating sound at 100Hz can be heard. Thus, a total value of about 30 ms is about as low as you can go without introducing these effects.

- Joint Channels In stereo, each channel can be compressed independently, sometimes causing the surrounding background noise to get louder on one channel at a time. This can sound strange. For example, a loud drum beat in the left channel will make the background noise sound louder in the right than in the left. If Joint Channels is checked, both channels are used to find a single input dB value, and both channels are amplified together by the same amount (thus preserving the stereo center-channel image). For instance, a loud drum beat on the left channel will also cause the right channel to be reduced in level by an equal amount.

Level Detector The Level Detector determines the current waveform input amplitude, which is used as the input side of the dynamics processor. Here are its components:

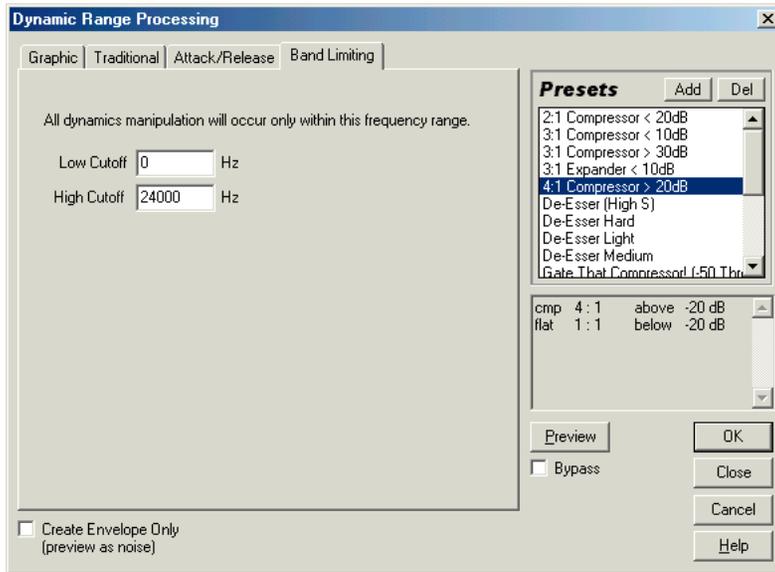
- Input Gain is added to the signal before it goes into the Level Detector (the section that detects the current level). This essentially “pushes” the graph up or down by the gain given.
- Attack Time is applied when retrieving the current amplitude information. Attack Time determines the time in milliseconds that it takes for the processed output signal to reach its specified output volume. If there is suddenly a quiet portion that drops 30dB, it will take “this much time” before the output actually drops to its corresponding volume level.
- Release Time is applied when retrieving the current amplitude information. Release Time is the time it takes the end of a previous output level to reach the specified output volume. For example, where the Attack is the time it takes for the start of a pulse to reach the desired output volume, the Release is the time it takes for the end of the pulse to reach the desired level.
- Peak mode is a graph interpretation method that is a little more difficult to use than RMS, and is a bit outdated. It equates to the RMS value times two. That is, if the RMS value is -20dB, then the equivalent peak value will be -40dB. This occurs because the RMS value calculated was mapped to a peak sample value for output. This method is available here for backward compatibility.
- RMS is a graph interpretation method that more closely matches the way people hear volume. This mode causes the output to be exactly the RMS amplitude that is specified in the graph. For example, a limiter (flat horizontal line) at -10dB causes the RMS amplitude of the result to average -10dB (where 0dB is a maximum amplitude sine wave without clipping).

Lookahead Time Lookahead Time is used to handle sharp spikes that may occur at the onset of a louder signal. At times, and for brief instances, these onsets can go beyond the limits of your compressor settings, which may be desirable in certain compression scenarios since it can enhance the impact of, say, a drum hit. However, this is obviously not desirable if you are using limiting in order to reduce the maximum amplitude of the audio.

Note: The spikes occur because it takes a little time to determine (the Level Detector’s attack value) and react (the Gain Processor’s attack value) to the current signal level, so Lookahead Time will actually cause the attacks to start before the audio gets loud, instead of right on top of the transient. Otherwise, with a Lookahead Time of 0, a spike will stay loud until all of the attack times have elapsed.

Band Limiting Tab

The fields in the Band Limiting tab allow you to specify a frequency range where all dynamics manipulation will occur.



Dynamic Processing dialog – Band Limiting tab

Low Cutoff This is the lowest frequency that dynamics processing will affect. You can define a band, or range, to which compression or expansion is applied, within the current frequency range. The Low Cutoff point is the bottom boundary in defining this band. For example, with values of 1000 for Low Cutoff and 5000 for High Cutoff, dynamics processing only affects audio in the frequency range of 1kHz to 5kHz.

 To use the entire frequency range of the source material, leave this setting at 0.

High Cutoff This is the highest frequency that dynamics processing will affect. The High Cutoff point is the top boundary in defining this band. For example, with values of 1000 for Low Cutoff and 5000 for High Cutoff, dynamics processing only affects audio in the frequency range of 1kHz to 5kHz. To use the entire frequency range of the source material, leave this setting at 0.

 To use the entire frequency range of the source material, this setting should be at 1/2 the current sample rate (24000 for 48kHz, 11025 for 22kHz, etc.).

Create Envelope Only Check this option to apply any dynamics processing, and have the result returned as an amplitude envelope. While it probably won't be much fun to listen to, this envelope can then be copied and used with Modulate option of Edit > Mix Paste to modulate it with another sound's amplitude. Of course, it's also useful for just seeing the amplitude envelope of your audio. When you press the Preview button with the Create Envelope Only box checked, you'll hear pink noise instead of the audio that's highlighted in the Wave Display.

 You might want to use this feature on a copy of your waveform instead of the original file.

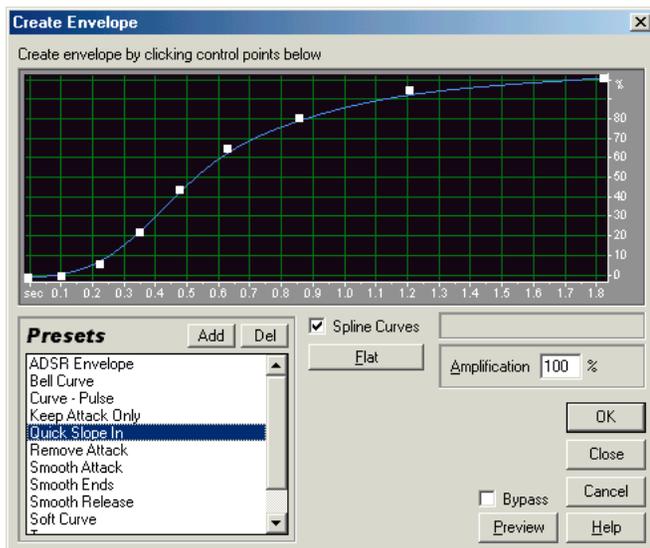
Presets Adobe Audition comes with several Dynamic Range presets, including some great-sounding compression and limiting settings. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the selected dynamic range processing setting before you apply it to the waveform. The selection is played over and over (“looped”) until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is looped. Check the Bypass box to hear the waveform without any changes in dynamic range for a quick “before and after” comparison.

 Use the RadioLimit, Fast Release, Boost preset to simulate the processed sound of a contemporary FM radio station.

Envelope

Envelopes give you control over which parts of your wave are amplified, and by how much. The top of the graph represents 100% (normal) amplification; the bottom represents full attenuation (silence). This function is handy when modifying tones that are generated with Adobe Audition, so as to create more realistic-sounding instruments and effects.



Create Envelope dialog

Graph The graph depicts time along the x-axis (left and right) and the new output level along the y-axis (up and down), with the blue line representing amplitude change.

Note: For more information about Adobe Audition’s graph controls (such as how to add and remove control points), see the “Looking at the Work Area” chapter.

Spline Curves Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use Spline Curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat The Flat button resets the graph to its default state by removing all control points.

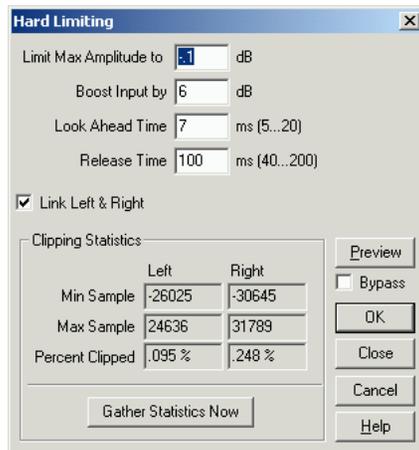
Amplification The value in the Amplification field specifies where the top of the graph is, measured in percentage. Then, when you draw your envelope curve, the top of the graph is whatever percentage you specify and the bottom is 0%. The default value of 100% means there will be no change in volume.

Presets Adobe Audition comes with several Envelope presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the selected envelope adjustment before you apply it to the waveform. The selection is played over and over (“looped”) until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear changes as the selection is looped. Check the Bypass box to hear the waveform without any changes in envelope processing for a quick “before and after” comparison.

Hard Limiting

Adobe Audition’s Hard Limiter enables you to drastically attenuate audio above a certain threshold, while leaving all audio samples below that threshold alone. For example, you can amplify a piece of audio well beyond where it would normally clip, and the Hard Limiter will soften the regions that would otherwise be clipping, ensuring that the maximum sample amplitude does not go above the given limit. This can be extremely handy if, for instance, you did a Mixdown to a 32-bit audio file, and only have a few places where the audio is clipping, but do not want to Normalize down the entire waveform (which would reduce the volume of it). Hard Limiting will bring down only those portions that are clipping (or would be clipping if converting to 16-bit).



Hard Limiter dialog

Here are the major components of the Hard Limiter window:

Limit Max Amplitude to This is the maximum sample amplitude that will be allowed.

 *To avoid clipping when working with 16-bit audio, set this value to no more than -0.1dB; if you set it to -0.5dB, you'll have a little more headroom for any future edits.*

Boost Input by Before performing the limiting, the audio may be pre-amplified first. To make a selection louder and ensure that no clipping occurs, you can use this to amplify it and soften the regions that would otherwise clip (similar to what is commonly done with the audio on TV commercials). Enter the desired amplification value, as measured in dB.

Look Ahead Time This is the amount of time (measured in milliseconds) that it will generally take to attenuate the audio before the loudest peak will be hit.

Note: If this value is too small, audible distortion effects may be heard. Make sure the value is at least 5 ms.

Release Time This is the time (measured in milliseconds) that it takes for the attenuation to rebound back 12dB (or roughly the time it would take audio to resume normal volume if an extremely loud peak was encountered). Generally around 100 ms (the default) works very well, and will preserve very low bass frequencies.

Note: If this value is too large, audio may just stay very quiet, and not resume normal levels for a while.

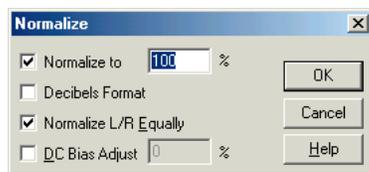
Link Left & Right If checked, both channels vary in loudness together. This preserves the stereo image of the source. To have the left and right channels vary independently, uncheck this option.

Gather Statistics Now Press this button to update the Clipping Statistics display. The Clipping Statistics fields show what percentage of the audio would clip if limiting were not performed. If you change any of the input parameters, you'll probably want to click this button.

Preview Press the Preview button to audition the selected hard limiting adjustment before you apply it to the waveform. The selection is played over and over (“looped”) until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear changes as the selection is looped. Check the Bypass box to hear the waveform without any changes in hard limiting for a quick “before and after” comparison.

Normalize

This command brings up the Normalize dialog, which enables you to set a desired peak level to which you want a file or selection raised. When set to 100% or 0db, normalization allows you to achieve the greatest amount of amplification that will not result in clipping (distortion). All parts of the wave are amplified at equal levels. For example, if the loudest peak of a waveform is at 80% and the quietest peak is at 20%, normalizing the file to 100% results in the waveform’s new loudest peak being 100%, and its new quietest peak being 40%.



Normalize dialog

Normalize to Enter the percentage of maximum to which you want to normalize. For example, choose 50% to compute values needed to amplify the selection no more than 50% of maximum (resulting in a 3dB attenuation from maximum output). Choose 100% (default) to apply the most amount of amplification possible without clipping.

Decibels Format With this box selected, the value in the Normalize to box is represented in decibels instead of percentage.

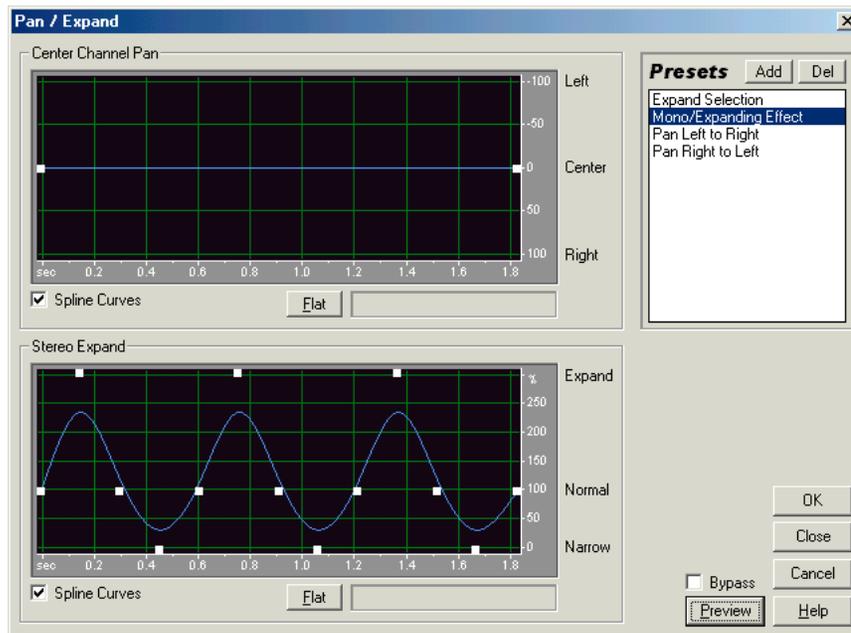
Normalize L/R Equally Check this box to use both channels of a stereo waveform in calculating the amplification amount. When it’s unchecked, Adobe Audition computes the left and right values in a stereo waveform separately, potentially amplifying one channel considerably more than the other. This could degrade the waveform’s stereo image.

DC Bias Adjust Some recording hardware may introduce a DC Bias, which results in the recorded waveform appearing to be above or below the normal center line. Check DC Bias Adjust and set it to 0% to center the waveform on the center (zero voltage) line. To skew the entire selected waveform above or below the center line, enter a positive or negative percentage.

 *If you’re planning to put normalized audio on CD, you might want to normalize the waveforms to no more than 96% as some audio compact disc players have problems accurately reproducing bits that have been processed to 100% (maximum) amplitude.*

Pan/Expand

The Pan/Expand effect is used to pan, or shift, the center channel of a stereo waveform. It also lets you expand or narrow the stereo separation of the left and right channels.



Pan/Expand dialog

Center Channel Panning makes use of the “surround” and “center” channels of a stereo recording, where the “surround” channel is the difference (L-R) of the two original channels, and “center” is the sum (L+R) of them. In this case, you can think of a stereo recording as having 4 channels (left, right, middle, surround), and this effect will pan these channels around. For example, you can pan hard left to get the old “center” channel coming out the left speaker, and the old “surround” channel coming out the right. This type of panning can provide realism to original stereo recordings.

Expanding works by subtracting out or adding differing amounts of left and right channel signal, so things occurring on the right or left are boosted or cut.

Both of these elements can be altered dynamically over time by using the respective graph.

Center Channel Pan This graph represents the pan position of the center channel of a stereo waveform over time. The graph’s x-axis (horizontal) represents the length of the waveform or selection, while the y-axis (vertical) represents the percentage of the pan from center. You can use the graph to position the center channel anywhere from hard left (-100%) or hard right (100%), with the corresponding surround channel moving right to left in the opposite direction. Use this for panning original stereo data more realistically than amplitude panning allows.

Stereo Expand This graph shows the expand level over time. This either amplifies (>100%) or removes (<100%) the differences between channels. The graph's x-axis (horizontal) represents the length of the waveform or selection, while the y-axis (vertical) represents the percentage of stereo expansion.

With some material, a stereo expanding effect may be achieved by increasing the differences between the left and right channels. The expansion level can vary over time for interesting effects (growing from a mono signal to a very wide stereo signal, for example).

Note: For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

Spline Curves Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use Spline Curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat The Flat button resets the graph to its default state by removing all control points.

Presets Adobe Audition provides several Pan/Expand presets. More information on presets can be found in the "Looking at the Work Area" chapter.

Preview Press the Preview button to audition the current Pan/Expand setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played. Check the Bypass box to hear the waveform without any changes in panning or expanding for a quick "before and after" comparison.

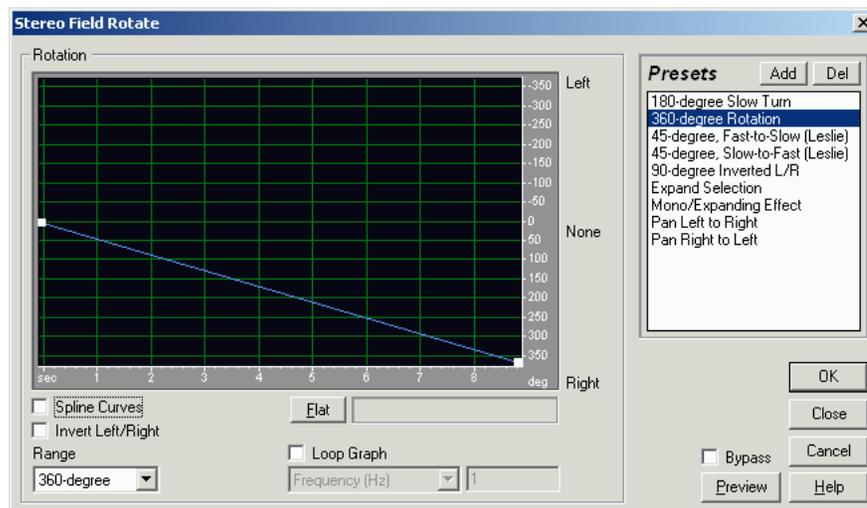
Automated When using Pan/Expand as a real-time effect in Multitrack View, it supports parameter automation.

- Check the Automated box in the Center Channel Pan section of the Multitrack View version of the Pan/Expand dialog to draw the Center Channel Pan curve using a yellow envelope on waveform blocks.
- Check the Automated box in the Stereo Expand section of the Multitrack View version of the Pan/Expand dialog to draw the Stereo Expand curve using a pink envelope on waveform blocks.

Note: You must have View > Show FX Parameter Envelopes in Multitrack View turned on in order to use parameter automation.

Stereo Field Rotate

This effect allows you to rotate the stereo field of an audio file. (Stereo field denotes where in space instruments or other sources are placed within the L/R image of a stereo waveform.) This is accomplished by manipulating the Stereo Field Rotation graph.



Stereo Field Rotate dialog

Note: If you specify a stereo field rotation of 180 degrees, both the left and right channels are 180 degrees out of phase. This means that the waveform is inverted. (This is the same as choosing the Invert effect from the Effects menu). At 90 degrees right, only the right channel is inverted. At 90 degrees left, just the left channel is inverted. In the range of -45 to 45 degrees, the results are identical to Adobe Audition’s Pan/Expand effect, panning the center channel left or right while panning the surround channel the opposite direction.

Rotation This graph represents the rotation of the stereo field over time. Use the graph to position the stereo field anywhere from hard left (the top of the graph) or hard right (the bottom of the graph) at any specific point in time. The x-axis (horizontal border) of the graph shows the waveform’s timeline, while the y-axis (vertical border) displays the number of degrees off center for the left and right channels.

Note: For more information about Adobe Audition’s graph controls (such as how to add and remove control points), see the “Looking at the Work Area” chapter.

Spline Curves Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use Spline Curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Invert Left/Right Check this option to reverse the way the graph goes, so an upward line spins the field clockwise instead of counter-clockwise.

 If you have Adobe Audition's Phase Analysis window (Analyze > Phase Analysis) open, you can actually watch the stereo field rotate. So, if your graph is drawn to rotate more and more to the right, you'll see the phase rotating clockwise (to the left). But, if you have Invert checked, that same graph will now rotate the phase image to the right. However, you'll actually hear the audio start panning to the right if your graph started at 0.

Flat The Flat button resets the graph to its default state by removing all control points.

Range Use this drop-down control to select the y-axis display range of the stereo rotation graph from 45 degrees to 360 degrees.

Loop Graph When this box is unchecked, the graphs are the same length as your selection, so the dynamic changes to your delay are graphed across the whole selection. If you check Loop Graphs, the graphs are only as long as specified, so they loop repeatedly over the course of the selection. Checking Loop Graphs also enables the drop-down control that lets you specify Period, Frequency, Tempo, or Total Cycles. You can either specify how long the graphs should be (Period), the how fast the cycle should move (Frequency), how many beats per minute the loop will have (Tempo), or how many times you want the graph to loop in the given selection (Total Cycles).

Presets Adobe Audition provides several Stereo Field Rotate presets. More information on presets can be found in the "Looking at the Work Area" chapter.

Preview Press the Preview button to audition the current Stereo Field Rotate setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played. Check the Bypass box to hear the waveform without any changes in stereo field rotation for a quick "before and after" comparison.

Automated When using Stereo Field Rotate as a real-time effect in Multitrack View, it supports parameter automation. Check the Automated box in the Multitrack View version of the Stereo Field Rotate dialog to draw the Center Channel Pan using a yellow envelope on waveform blocks.

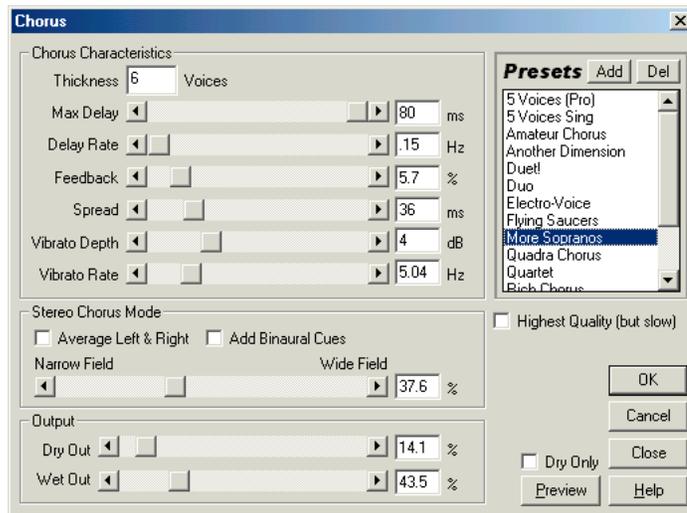
Note: You must have View > Show FX Parameter Envelopes in Multitrack View turned on.

Delay Effects

The options in this menu can be used to introduce a series of "repeats" or delayed echoes of the original input signal at regularly spaced or user-defined intervals. An audio file or selected range can be delayed in any of the following ways... in any of the following ways... in any of the following ways....

Chorus

The Chorus effect adds richness to a sound by making it sound as though several voices or instruments are being played at once. Chorus is a great way to add a degree of "presence" to an existing track. Use it to give a stereo effect to a mono sample (where left and right channels are identical), or to add harmony or "thickness" to a vocal track. You can also use chorusing to create some truly out-of-this-world special effects.



Chorus dialog

Adobe Audition uses a direct-simulation method of achieving the chorus effect, which means that each voice (or layer) is made to sound distinct from the original by slightly varying the timing, intonation, and vibrato. You can use the Feedback setting to add extra detail to the result.



You'll get better results if you convert mono files to stereo before applying a Chorus effect.

Chorus Characteristics

These are the characteristics used for each voice (layer) in the chorus. While the properties below apply to each voice, they represent ranges of random values, so each voice will be unique in each of these characteristics.

Thickness Thickness determines the number of voices that are simulated in the chorus effect. Enter the number of desired voices in this field. The final result may end up with an additional voice if the Dry Out setting is used (which mixes the original sample in with the chorused result).

Note: *The more voices you specify, the longer the effect will take to compute – but the result will have extra richness and fullness.*

Max Delay An important component of chorusing is the introduction of short delays (often in the 15-35 millisecond range) that vary in duration over time. Use this setting to limit the maximum amount of delay that will ever occur.

- If the setting is very small, all the voices will start merging into the original, and an unnatural flanging effect may be noticed.
- If it is set high, everything will start to sound “warbly”, rather like a tape that is being eaten by a cassette deck.

Delay Rate This parameter determines the time taken for the delay to cycle from its zero-to-maximum delay setting. Because the actual delay used varies over time, the pitch of the sample will increase or decrease over time, placing each voice slightly out of tune with the others (which is what gives the effect of a separate voice). For example, a value of 2Hz means the delay used could vary from no delay to the maximum delay and back twice each second (sort of a pitch vibrato at 2 times a second). Please note that this is only a maximum; if you set it to 2Hz, it may only go part way between zero and the maximum delay, and then start cycling back before reaching the maximum.

- If this setting is very low, the individual voices won't vary much in pitch at all.
- If it is set high, the voices may vary so quickly that "warbling" effects can be heard.

Feedback Use this setting to include a certain percentage of processed chorused voices back into the mix. This can give your waveform an extra echo or reverb effect.

Note: Be very careful with this setting. Just a little feedback (less than 10%) can give an extra richness to the effect (and even this depends on the delay and vibrato settings). If too much feedback is used, more traditional feedback noise will be heard, like loud ringing or other artifacts. These may get so loud that they will clip and destroy the signal. However, sometimes this can be a desired effect, as in the Flying Saucers setting, where it generates sounds that are reminiscent of warbly UFOs whizzing around your head.

Spread The spread setting gives an added delay to each voice, separating them in time by as much as 200 milliseconds (1/5th of a second).

- High spread values cause the separate voices to start at different times – the higher the value, the further apart the onset of each word may be, for example.
- With low values, the effect is of all the voices singing in unison. Depending on other settings, low values may also bring out some flanging effects, which may be undesirable if your goal is a realistic chorus effect.

Vibrato Depth Another property that varies with each voice is vibrato, which describes how the amplitude varies over time. This value determines the maximum variation in amplitude that will occur.

For example, a value of 5dB may alter a chorused voice by varying its amplitude by as much as 5dB louder or quieter than the original.

- If this setting is extremely low (less than 1dB) the vibrato may be unnoticeable unless the Vibrato Rate has been set extremely high.
- If Vibrato Depth is set too high, however, the sound may cut in and out, creating an objectionable warble. Natural vibratos occur around 2dB to 5dB.

Please note that this is a maximum depth – just as in the delay settings, the vibrato volume may not go as low as the setting indicates at times. This limitation is intentional, as it gives a more natural feel to the effect.

Vibrato Rate The Vibrato Rate determines the maximum rate at which vibrato will occur.

- With very low values, instead of a vibrato effect, the resulting voice will slowly get louder and quieter, like a singer that cannot keep his or her breath steady.
- If the vibrato rate is set high, the result can be very unnatural – more like a singer who had too much coffee.



With very high settings, you can achieve interesting special effects (as in the preset Another Dimension).

Stereo Chorus Mode

The Stereo Chorus Mode settings are only active when working with stereo files. They determine where the individual voices will be placed in the stereo field, as well as how the original stereo signal will be interpreted.

Average Left & Right Adobe Audition can either average the original left and right channels or keep them separate in order to preserve any stereo image that may already exist. If you leave the box unchecked, the stereo image will be preserved when processing a stereo source file. (For example, spatial binaural cues such as those that exist in reverberated audio or live stereo recordings will be preserved.) Note that there is no need to check this box if the sample was originally monophonic, as it will have no result – except that if the box is checked, Adobe Audition will process the sample more slowly.

Add Binaural Cues Check this option to add separate delays to the left and right outputs of each voice. This can make each voice appear to be coming from a different direction, but only when played through headphones. If the audio is meant to be played through speakers, this option should be turned off. In addition, when you add binaural cues, the volume of the right channel for a voice panned all the way to the left is still significant. If no cues are added, no output will be sent to the right channel. This is why greater separation can be heard when listening through speakers if binaural cues aren't used.

Narrow Field/Wide Field Slider Use this control to specify the stereo field of the Chorus Effect.

Stereo Field denotes where in space instruments or other sources are placed within the L/R image of a stereo waveform. A waveform that contains sound sources that are spread widely from left to right is said to have a wide stereo field, while those having images that are centered around the middle are said to have a narrow stereo field. In the case of Adobe Audition's stereo field Settings, the narrower the stereo field, the greater the chance that the chorused voices will be placed in the center of the L/R stereo image.

At a setting of 50%, all the voices are spaced evenly about a half circle from left to right. For example, if a thickness of 5 is used, each voice will be panned such that voice #1 will be to your left, voice #2 will be left-of-center, voice #3 will appear in the center, voice #4 will be right-of-center, and voice #5 will be on your right. At settings higher than 50%, the voices start migrating to the outer edges: voices to the left go farther to the left, and those to the right go farther right.

- If you use an odd number of voices, then there will always be one voice directly in the center.
- With an even number of voices, and very high stereo field settings, all voices are pushed either hard left or hard right.

Output

The final output can be a mix between the original input (dry) signal and the chorused (wet) signal. Ordinarily, both settings should be less than 100%; otherwise the overlaying of several voices may cause clipping.

Dry Out This setting determines how much of the unprocessed signal is mixed into the final output. If you set it to zero, Adobe Audition adds the original voice to the number of processed voices (which is determined by the Thickness setting).

 Keep the original signal near 100% and reduce the Wet Out to give the singer or instrument a “backup chorus”. Reduce the blend to 30% or so to blend the original signal with the processed chorus.

 Use Adobe Audition’s multitrack mixing capabilities to dynamically bring in and fade out the chorus. Add the chorus effect to a copy of the original audio and set this value to zero to create a chorus-only version of the original. In the mixer, insert both the original and the fully chorused versions. Use the volume envelope control to adjust the volume of the chorus over time, or just tweak the final amplitude of the background chorus with the track’s volume settings. This technique can be handy for emphasizing certain portions of the singing with a backup chorus.

Wet Out This setting determines how much of the processed signal is mixed into the final output. Generally, this value should be set lower whenever more voices are used. For example, with a Thickness setting of 3, a setting of 40% would be appropriate, but with Thickness at 10, 20% might be better. The best value will vary depending on the number of voices, and the desired stereo image field settings.

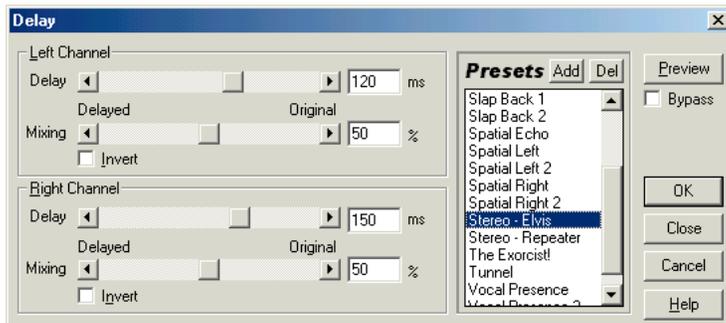
Presets Adobe Audition provides several Chorus presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Highest Quality (but slow) Check this box to make sure the chorusing effect is the best quality. However, this will slow down processing time both when you preview the effect and when you actually apply it.

Preview Press the Preview button to audition the current Chorus setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played. Check the Bypass box to hear the waveform without any applied chorusing effects for a quick “before and after” comparison.

Delay

Delay can be used to create single echoes, as well as a number of other effects. Delays of 35 milliseconds or more will be perceived as discrete echoes, while those falling within the 15-34 millisecond range can be used to create a simple chorus or flanging effect. (These effects won’t be as effective as the actual chorus or flanging effects within Adobe Audition’s Effects menu, as the delay settings will be fixed and won’t change over time.)



Delay dialog

By reducing the delay times further down into the 1-14 millisecond range, you can spatially locate a mono sound (a sound with the same information for both the left and right channels) so that it appears as though the sound is coming from the left or the right side, even though the actual volume levels for left and right of the wave are identical.

Delay These sliders allow you to adjust the desired delay for both the left and right channels up to plus or minus 500 milliseconds. Entering a negative number means that you can move a channel ahead in time instead of delaying it. For instance, if you enter 200 milliseconds for the left channel, the delayed portion of the affected waveform will actually be heard before the original part.

Mixing Move the Mixing sliders to set the percentage of Delayed (wet) signal and unprocessed Original (dry) signal to be mixed into the final output. A value of 50 will mix the two evenly.

Invert If this box is checked, the positive values of the selected waveform become negative. This means that cancellation will occur when you mix the inverted and original waveform. Inverting the delayed signal can be used for special effects, such as creating a quick-and-dirty comb filter.

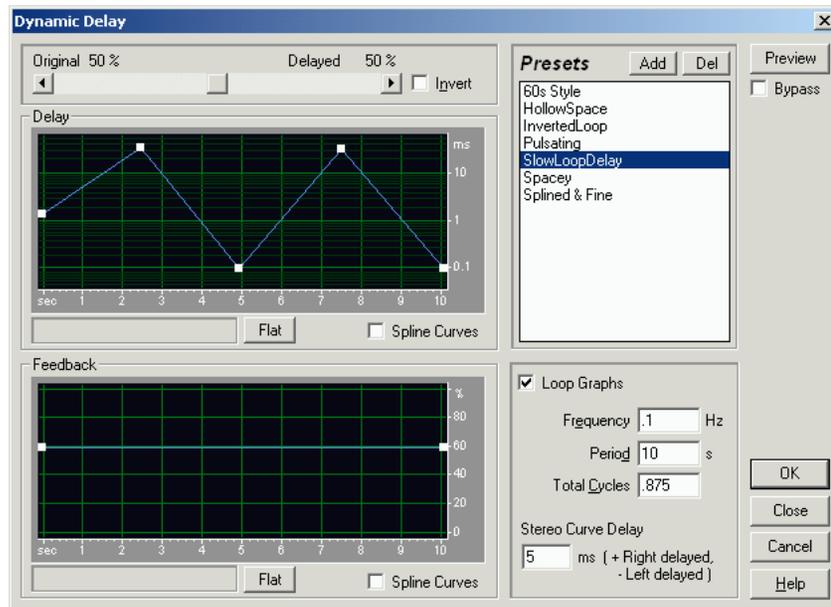
Presets Adobe Audition provides several Delay presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Delay setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played. Check the Bypass box to hear the waveform without any applied Delay effects for a quick “before and after” comparison.

 *Make sure there is enough silence at the end of the waveform for the delay trails to complete. If the delay abruptly cuts off before it fully decays, undo the Delay effect, add several seconds of silence using Adobe Audition’s Generate > Silence command, and rerun the Delay effect.*

Dynamic Delay

Dynamic Delay allows you to change the amount of delay over the length of the waveform. For instance, for the first five seconds of an audio file you can have a 2 millisecond delay, for the next 15 seconds you could have a 20 millisecond delay, for the following 10 seconds the delay could be at 7 milliseconds, and so on.



Dynamic Delay dialog

 *Dynamic Delay is especially cool when used as a real-time effect in the Multitrack View. If you add the dynamic delay (or Dynamic EQ, which has a similar principle) to the Multitrack View, you get a new envelope that determines the delay.*

Original - Delayed Slider Use this control to mix in the desired amounts of original and delayed audio. The more of the original signal you have in the mix, the less delay you'd have – and vice-versa.

Invert Check this box to invert the delayed signal. This causes the waves to cancel out periodically, instead of reinforcing the signal.

Delay Graph Use the Delay Graph to specify the amount of delay in the Dynamic Delay effect. The x-axis (horizontal edge) represents the timeline of the selection, while the y-axis (vertical edge) represents the length of delay (measured in milliseconds). Functionally, the Delay Graph works just like any other Adobe Audition graph control. For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

Spline Curves Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use Spline Curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Feedback Graph Use the Feedback Graph to specify the amount of feedback in the Dynamic Delay effect. The x-axis (horizontal edge) represents the timeline of the selection, while the y-axis (vertical edge) represents the percentage of feedback. Functionally, the Feedback Graph works just like any other Adobe Audition graph control. For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

Loop Graphs When this box is unchecked, the graphs are the same length as your selection, so the dynamic changes to your delay are graphed across the whole selection. If you check Loop Graphs, the graphs are only as long as specified, so they loop repeatedly over the course of the selection. Checking Loop Graphs also enables the Frequency, Period, Total Cycles, and Stereo Phasing fields. Frequency, Period, and Total Cycles are all ways of determining the length of the looped section. You can either specify the how fast the cycle should move (Frequency), how long the graphs should be (Period), or how many times you want the graph to loop in the given selection (Total Cycles). If you change one of these values, the other two reflect the change.

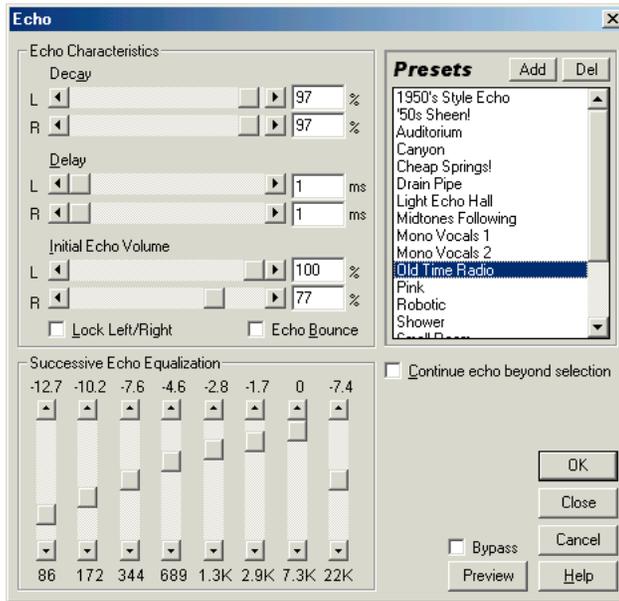
Stereo Curve Difference This setting is only effective when Dynamic Delay is used in Adobe Audition's Multitrack View. It's the number of milliseconds one channel is behind with respect to the other when following the envelope you draw in Multitrack View. Positive values will delay the right channel, while negative values will delay the left. The audio itself isn't delayed, just the curve used. This leads to a neat effect where the phasing sound of the delay lags in one channel with respect to the other.

Automated When using Dynamic Delay as a real-time effect in Multitrack View, it supports parameter automation. Check the Automated box in the Delay section of the Multitrack View version of the Dynamic dialog to draw the Delay curve using a purple envelope on waveform blocks.

Note: You must have View > Show FX Parameter Envelopes in Multitrack View turned on in order to use parameter automation.

Echo

This function adds a series of repeated, decaying echoes to a sound. (For a single echo, use the Delay function instead).



Echo dialog

Achieve effects ranging from a Grand Canyon-type “Hello-ello-llo-lo-o” to “boingy” drain pipe sounds by varying the delay amount. Change a room’s characteristic sound from one with reflective surfaces (yielding echoes that have a bright, shiny high-end) to one that is almost totally absorptive (very little reflected high-end sounds) by equalizing the delays.

 You can create striking stereo echo effects by using different Left and Right channel values for the Decay, Delay, and Initial Echo Volume controls.

Echo Characteristics The following characteristics can be applied to echoes.

- Decay is the falloff ratio of an echo. Each successive echo will trail off at a certain percentage less than the previous one. A decay setting of zero percent results in no echo at all, while a decay of 100 percent produces an echo that never gets quieter.
- Delay sliders are used to specify the number of milliseconds that is placed between each echo. For example, a setting of 100 milliseconds results in a 1/10th-second delay between successive echoes.
- Initial Echo Volume: Adjust these sliders to set the percentage of echoed (wet) signal to be mixed with the original (dry) signal in the final output.
- Lock Left/Right: With stereo waveforms, the left and right channels may be processed separately. When the Lock Left/Right box is checked, the Decay, Delay, and Initial Echo Volume scroll bars for the left and right channels move together, maintaining the same settings for each channel. Uncheck this box to adjust the channels separately.

- **Echo Bounce:** Select this option to make the echoes bounce back and forth between the left and right channels. If you want to create one echo that bounces back and forth, select an initial echo volume of 100% for the one side (left or right), and 0% for the other. Otherwise, the settings for each channel will bounce to the other, creating two sets of echoes on each channel.

Successive Echo Equalization This eight-band echo “quick filter” enables you to choose which frequencies are removed from the echo first. Each successive echo is passed back through the equalizer, allowing for control in simulating the natural sound absorption of a room. A setting of zero will leave the frequency band unchanged, while a slider’s maximum setting of -15 will decrease that frequency by 15db. And, since it would be -15dB for each successive echo, some frequencies will die out much faster than others.

Note: Unlike most other equalizer-like controls, you can’t increase the attenuation of frequencies with Successive Echo Equalization; you may only decrease them.

Continue echo beyond selection When this box is checked, the echo effect continues beyond the right-hand boundary of the waveform’s selected range. Sounds that extend beyond this boundary will not be affected. In other words, the selected range will be echoed, and this echo will naturally decay beyond the selection boundary, while unselected sounds will not be echoed. If the box isn’t checked, the echo effect will stop at the right-most waveform boundary visible in the waveform window. This means if the window is zoomed in, the echoing will stop before the range or file ends.

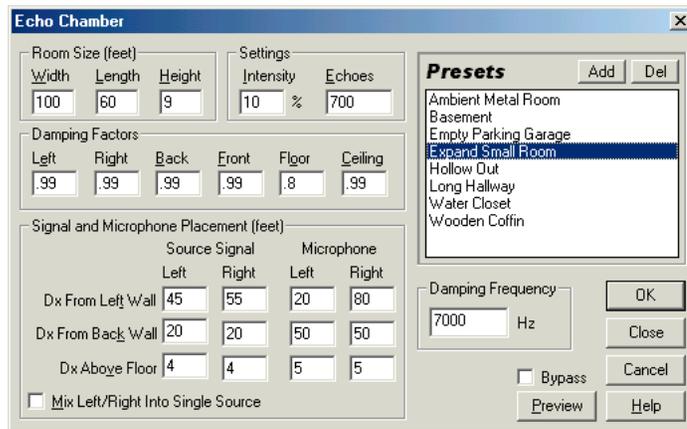
 *Make sure there is enough silence at the end of the waveform for the echo trails to complete. If the echo abruptly cuts off before it fully decays, undo the Echo effect, add several seconds of silence using Adobe Audition’s Generate > Silence command, and then rerun the Echo effect.*

Presets Several Echo presets come with Adobe Audition. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Echo setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Echo Chamber

The Echo Chamber function can simulate the ambiance of almost any room. Settings are provided for specifying your virtual room’s size and surface characteristics, along with the placement of virtual microphones. The number of echoes is adjustable, up to about 25,000. Keep in mind that the more echoes there are to calculate, the longer it will take for the function to complete.



Echo Chamber dialog

💡 One cool use for this function is to give a stereo effect to monaural audio by adding ambiance. Choosing a “left” microphone placement that is one or two feet away from the “right” microphone will simulate a listener’s ears and will give the effect of “being there”. (Try monitoring the result with headphones to experience the best effect.) Make sure you use Edit > Convert Sample Type to make the mono signal stereo that you can choose separate virtual microphone locations.

A spatial, stereo expansion effect may be created by placing the two microphone locations far apart, further apart in the settings than if you were playing them through speakers in real life. For example, if your stereo speakers are 6 feet apart, try placing the left and right virtual microphone settings 20 or 30 feet apart.

Room Size (feet) The length, width, and height of the virtual room, as measured in feet, are specified here. (There are approximately 0.3 meters per foot.) When entering source and microphone locations, they must lie between zero and the room’s width for the Dx From Left Wall parameter, and between zero and the room’s length for the Dx from Back Wall parameter. Room sizes can be as large as you like, but memory requirements will grow as room size increases. This means that some systems may run out of memory if very large room sizes are used.

Intensity This determines the percentage of amplitude of the direct (original) signal. Because echoes (picked up by the virtual microphone) add to the signal’s overall amplitude, you should always set Intensity to under 100% to avoid clipping. In fact, the more echoes there are, the lower the Intensity value should be set. For example, a setting of about 30% would be appropriate for 100 echoes, whereas 15% would be better for 1,000 echoes.

Echoes This is the number of actual echoes to produce. To achieve a nice reverberating ambiance effect, use at least 10,000 echoes. The more echoes that are generated, the “truer” the result will sound, but larger numbers of echoes require more processing time. 25,000 echoes produce a very realistic result

💡 In order to reduce wait time when testing out a virtual chamber’s size and overall room sound, you may want to generate only 100 or so echoes. Once you achieve the sound you want, you can undo the test effect and increase the number dramatically for the final production. A very fast system should be able to generate up to 100,000 echoes, depending on the virtual room size and available memory.

Damping Factors Use damping factors to describe the type of virtual room in which the audio is being played. These factors can simulate wall coverings, floor coverings, and other objects in the room that absorb sound. Granted, in real life, various objects absorb different frequencies, but in this simulation all frequencies are reflected equally.

For example, the fact that cement reflects high frequencies better than low ones isn't accounted for, but great effects can still be achieved (and they're much more realistic than those of basic Echo functions). The effect of speaker placement enhancing or canceling certain frequencies, though, is still accurate. A damping factor of 1.0 is the highest, simulating total reflectivity, while a factor of 0.0 is the lowest, which represents complete sound absorption (no reflecting surface).

 *To give more control over the environment, apply damping factors to any of the four walls, floor, and ceiling. If a wall has a damping factor of 1.0, it is totally reflective (rather like cement). If a wall has a very low damping factor, like 0.05, it will absorb most of the sound (like carpeting, heavy draperies, or sound proofing panels). You can also lower the damping factor of some of the walls to simulate other objects in the room, which effectively absorb some of the audio.*

Signal and Microphone Placement (feet)

- **Source Signal:** Enter the desired values of the Source Signal (your waveform or selection), measured in feet, in the Dx (Distance) from Left Wall, Dx from Back Wall, and Dx Above Floor fields. (When using a stereo waveform, you can have different values for both the left and right channels.) The signal will then simulate a single, non-directional point source, meaning that the sound will radiate outwards in all directions. The distance between the source and the walls affects which frequencies are enhanced and is crucial to the overall ambient effect.

 *Use the Dx (Distance) values in conjunction with Room Size values to determine exact 3D placement. If you use a value greater than the dimensions of the room (a Dx Above Floor of 21 feet in a room with a Height of 20 feet, for example), Adobe Audition will use the greatest possible value (20 feet in this case).*

- **Microphone Placement:** Enter the desired virtual microphone values, measured in feet, in the Dx (Distance) from Left Wall, Dx from Back Wall, and Dx Above Floor fields. Depending on whether the waveform is mono or stereo, up to two virtual mike pickups can be used. Each microphone is routed to its own respective channel of a stereo audio file. The resulting echoes are exactly what the microphone(s) would pick up in the room at the specified location. Place the virtual microphones in a stereo setting one foot apart to simulate human ears. The placement of and distance between the mikes gives the brain cues about the directions of each echo and the size of the room. Try listening with headphones to virtual microphones that have been placed far apart; this gives a very large “aural” or “spacey” feeling to the sound.

Note: *Make sure you always place the virtual microphone(s) sufficiently far away from the source. If the microphone and source are too close together, you will just hear the source and no echoes. This is analogous to placing your ear right next to the sound source, where you will hear the sound only (due to its loudness).*

Mix Left/Right Into Single Source When working with stereo audio, there are actually two source signals (left and right) that can be placed independently. In most instances, the stereo effect will be dramatically enhanced and sound fuller and often richer when this box is unchecked. This does, however, require twice as many calculations as a single audio source. If faster processing is desired, check this option, which will sum the left and right channels of a stereo waveform before routing it to the effects processor.

Damping Frequency The Damping Frequency value lets you specify an upper frequency limit of sound that will be reflected. For instance, if you set a damping frequency of 7000Hz, then frequencies above 7000Hz will be cut for each unit of time. Use lower values to get a “warmer” feel.

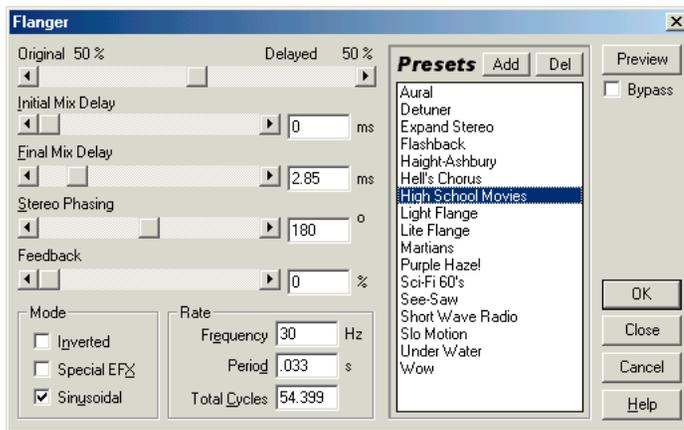
Presets Several Echo Chamber presets come with Adobe Audition. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Echo Chamber setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

 *Make sure there is enough silence at the end of the waveform for the echo trails to complete. If the echo abruptly cuts off before it fully decays, undo the Echo Chamber effect, add several seconds of silence using Adobe Audition’s Generate > Silence command, and then rerun the Echo Chamber effect.*

Flanger

Flanging was originally achieved by sending an audio signal to two reel-to-reel tape recorders, and then physically slowing down the reels of one machine. The resulting sound is a phase-shifted, time delay effect, characteristic of some of the psychedelic sounding recordings in of the 1960s and 1970s. (If you’ve ever listened to “Listen To The Music” by the Doobie Brothers, you’ve heard flanging in action.) Use Adobe Audition to create the same effect by slightly delaying and phasing a signal at predetermined or random intervals.



Flanger dialog

Original - Delayed Slider Use this control to adjust the mix between the amount of original (dry) and flanged (wet) signal. If the Original value is at 100%, the result is no flanging at all. If the Delayed signal is set to 100%, the result is a wavering sound, rather like a bad tape player. Portions of both signals must be present if you want to achieve the characteristic cancellation and reinforcement that occurs during flanging.

Initial Mix Delay This slider sets the point in milliseconds at which flanging will start behind the original. The flanging effect occurs by cycling over time from an initial delay setting to a second (or final) delay setting.

Final Mix Delay This control sets the point in milliseconds at which flanging will end behind the original. It refers to the flange effect's secondary (or final) delay setting, with the flange being cycled between this and the Initial Mix Delay at a specified rate.

Stereo Phasing This control lets you set the left and right delays at separate values, which are measured in degrees. For example, a setting of 180 puts the right channel at the initial delay value at the same time that the left channel is at the final delay value. This setting can reverse the initial/final delay settings for the left and right channels, thus creating a circular, psychedelic effect.

Feedback The Feedback slider determines what percent of the flanged signal is fed back into the flanger. With no feedback, the effect only uses the original signal. With feedback added, the effect uses a percentage of the affected signal from before the current point of playback.

Mode Three options are offered for mode:

- **Inverted:** Check this box to invert the delayed signal when flanging. This causes the waves to cancel out periodically, instead of reinforcing the signal. If the Original - Expanded mix settings are set at 50/50, the waves will cancel out to silence whenever the delay is at zero.
- **Special EFX:** Selecting this box mixes both the normal and inverted flanging effects. The delayed signal is summed while the leading signal is subtracted out.
- **Sinusoidal:** If this option is chosen, the transition from initial delay to final delay and back will follow a sine curve. Otherwise, the transition is linear, and the delays from the initial setting to the final setting are at a constant rate. With Sinusoidal checked, the signal is at the initial and final delays more often than it is between delays.



The Invert, Special EFX, and Sinusoidal options give you a lot of control over the flanging effect, so try experimenting with them to achieve the effect you want.

Rate The Frequency (measured in hertz), Period (measured in seconds), and Cycles (measured in cycles) settings refer to the rate at which the delay cycles between the initial delay and the final delay. Different settings can result in widely varying effects. For example, a setting of 0.5 cycles causes the effect to start with the initial delay, and end with the final delay. If a frequency of 4 is chosen, the flanging will cycle from the initial delay to the final delay and back again four times per second.

Presets Adobe Audition comes with several Flanger presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Flanger setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

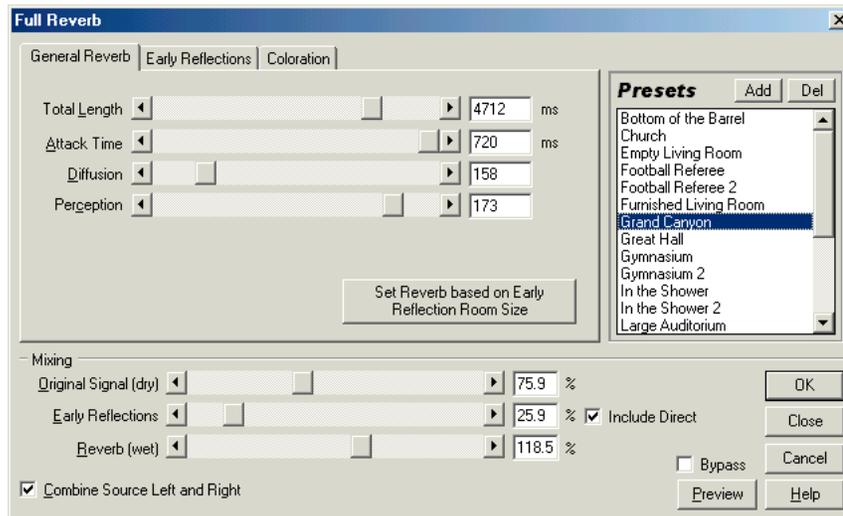
Full Reverb

Full Reverb, like Adobe Audition’s standard Reverb effect, is used to simulate acoustic space. It’s also convolution-impulse-based (like the standard Reverb), which means there are no ringing, metallic, or other artificial sounding artifacts. However, specific resonating can be achieved if desired (see Coloration).

Full Reverb has some unique features not found in other reverbs, like Perception to simulate room irregularities, and source location to place the “singer” off-center, and have the early reflections realistically model their position within rooms that have acoustically desirable dimensions that you can customize. Practically any wall surface material or other sound-affecting factors can be simulated by changing the reverb’s frequency absorption using a 3 band “parametric EQ” style interface (in the Coloration tab).

***Note:** Whenever any of the reverb characteristics are modified, a new impulse is computed (which can be several megabytes in size) to simulate the environment you’re specifying. This means it takes more CPU power to perform the reverberations, but they are much more natural sounding and easier to tailor. Because every parameter change (other than a mix level change) requires a new reverb “impulse” to be built, you may have to wait a few seconds after pressing Preview for the reverb generation process. Once built, the preview should generally run in real time, and subsequent previews or pressing OK won’t require rebuilding the impulse, nor will adjusting any of the mix level controls or the Include Direct checkbox.*

General Reverb Tab



Full Reverb dialog – General Reverb tab

Total Length This is essentially the time, measured in milliseconds, that it takes for the reverb to decay 60dB. However, depending on the Coloration parameters, certain frequencies may take longer to decay to 60dB, while other frequencies may decay much faster. Longer values give longer reverb tails, but also take longer to compute. The effective limit is about 6000 milliseconds (a 6-second tail). The actual tail generated is much longer than this in order to allow for decaying into the background noise level.

Attack Time This is the time, measured in milliseconds, that it takes for the reverb to build to its maximum amplitude. Generally, reverbs tend to build up over a short time span, and then decay at a much slower rate. Interesting effects can be heard with extremely long attack times (like 400 milliseconds or more).

Diffusion This setting controls the rate of echo buildup and how diffuse the echoes are. High diffusion values (above 900) give very smooth reverbs, without distinct “echoes” heard in them. Lower values will produce more distinct echoes since the initial echo density will be lighter, but the density will build over the life of the reverb tail.

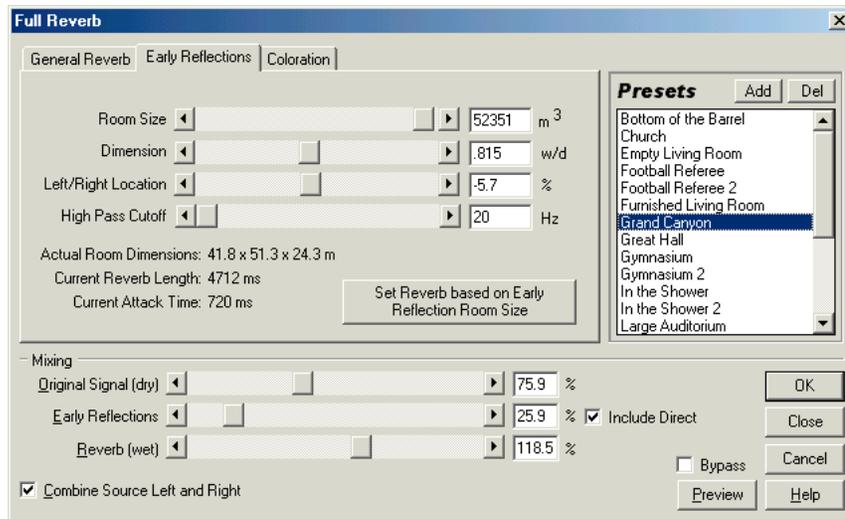
 *Very interesting “bouncy” effects can be obtained by using low Diffusion values and high Perception values. Using low Diffusion values, and somewhat low Perception values with long reverb tails give the effect of a football stadium or similar arena.*

Perception This value models irregularities in the environment (objects, walls, connecting rooms, etc.). Low values will do a smoothly decaying reverb without any frills. Larger values will give more distinct echoes (coming from different locations).

 *If a reverb is too smooth, it may not sound natural. Values up to about 40 will give just enough variation to the reverb to simulate small room variations.*

Set Reverb based on Early Reflection Room Size For convincing sounding reverbs, the tail length should be correlated with the feeling of room size given by the Early Reflections. Pressing this button will adjust the Total Length and Attack Time to roughly match the room size given in the Early Reflections tab. After these are set, you’re free to fine-tune the length and attack time to change the effect.

Early Reflections Tab



Full Reverb dialog – Early Reflections tab

Room Size This is the actual volume of the virtual room, as measured in cubic meters. The larger a room is, the longer the reverbs are for that room. Use this control to create virtual rooms from only a few meters to giant coliseums.

Dimension The ratio between the room's width (left to right) and depth (front to back) can be chosen with this slider. A sonically appropriate height is calculated and reported below after Actual Room Dimensions. Very narrow rooms simulate a person being at the other end of a long hallway. These rooms generally sound louder because of all the echoes reflected off the narrow walls. Extremely wide rooms are not very deep. If you want to simulate a person being at one end of the hall or the other in a very wide room, use the Location slider. The wider a room, the easier it is to hear the left/right location of the source. Generally, rooms with width-to-depth ratios between 0.25 and 4 provide the nicest sounding reverbs.

Left/Right Location The source can be placed off-center to produce a different set of incoming early reflection echoes. Checking the Include Direct box in the Mixing section below will also adjust the Original Signal to sound as if it is coming from the same location by delaying one of the channels. Very nice effects are possible with singers being very slightly off center, say, between 5% and 10% left or right.

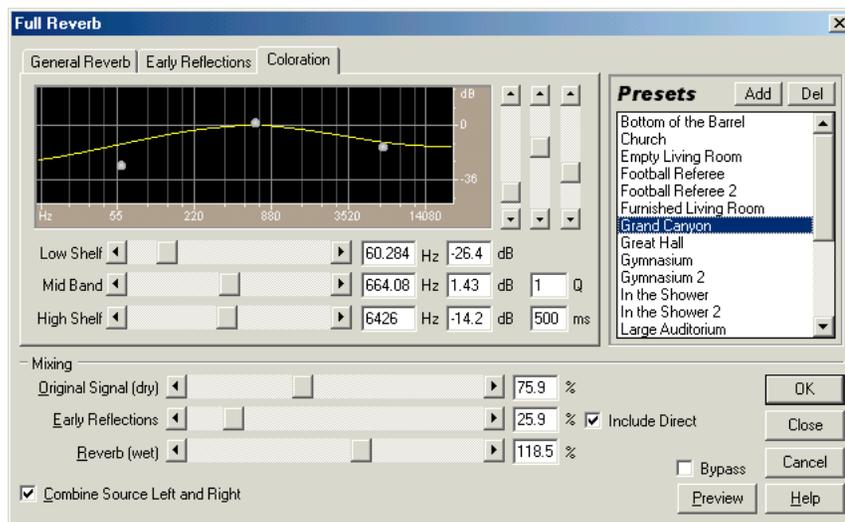
High Pass Cutoff Very low frequency (less than 100Hz) instruments (such as bass or drums) can get phased out for small rooms when the early reflections mix with the original signal. To prevent the loss of low frequency sounds, use this slider to specify a frequency above that of the sound you wish to keep.

- If the cutoff setting is too high, you may not get a realistic image of the room size.
- If the cutoff setting is too low, bass frequency sound may disappear when using small room sizes.

Good settings are generally between 80Hz and 150Hz.

Set Reverb Based on Early Reflection Room Size Once a room size and dimension are chosen, it is advisable to click this button to set the reverb length and attack time to match the current room size. This will make for a more convincing listening experience. While the reverb length and attack can be adjusted later, this provides a starting point for experimentation.

Coloration Tab



Full Reverb dialog – Coloration tab

Amplitude Sliders The three amplitude sliders to the right of the graph correspond to the low shelf, mid band, and high shelf bands, from left to right. A quick way to get a different sounding reverb is to just adjust these sliders up or down.

Frequency Sliders (Low Shelf/Mid Band/High Shelf) These are the frequency sliders directly beneath the graph. The cornering frequency for the shelves, or the center frequency for the mid band, can all be set to any frequency desired. For example, to “warm up” the reverb, you might want to lower the High Shelf frequency while also reducing its amplitude.

Q Box This value describes how wide the mid band’s affective area is frequency-wise. Lower values affect a wider range of frequencies (hence higher values will narrow the effective range).

💡 For distinct resonance, use values like 10 or higher. For just general boosting or cutting a wide range of frequencies, use lower values like 2 or 3.

💡 Sometimes it may be desirable to enhance some quality of the audio being processed, like to bring out a singer’s voice. Try boosting the frequencies just around the natural frequency of their voice to enhance resonance in that range – say, in the 200Hz to 800Hz range.

ms Box This is the time, specified in milliseconds, that it will take for the reverb to decay at each frequency following the coloration curve. For more colored reverbs, use lower settings (like 100 to 250). Values up to 700 work fine. Basically, the lower the value, the more the graph affects the resultant reverberation.

Mixing Section

Original Signal (dry) This slider adjusts the amount of Original Signal with respect to the other levels to give the feeling of distance between the listener and the source.

- Use a low signal level to sound far away.
- Use a high level (near 100%) along with low levels for the other sliders to give the feeling of being close to the source.

Early Reflections Since these are the echoes that first reach the ear, they give the listener a feeling for the overall room size and dimension.

- If the mixing for this is set too high though, the result will sound somewhat artificial.
- If the mixing for this is set too low the audio cues for room size will be lost.

Generally about half the volume of the original signal is a good starting point.

Reverb (wet) This controls the volume of the dense layer of sound usually associated with reverb. Like the Early Reflections volume, this slider should be fine-tuned to give a pleasing mix that gives the impression you are after. If it's too loud, the reverb sounds very unnatural. The balance between the reverb and the original signal gives the impression of distance, so increase the reverb volume with respect to the original signal to make the source sound further away.

Include Direct With a stereo waveform, checking this box will cause the original signal's left and right channels to be phase shifted (delayed) slightly to match the direction of the incoming early reflections. The Left/Right Location slider in Early Reflections determines the direction the audio will sound like it's coming from when listened to with stereo headphones.

Combine Source Left and Right When working with stereo audio, there are actually two source signals (left and right), that can be placed independently. However, this requires twice as many calculations as a single audio source. If you're after faster processing time, check the Combine Source Left and Right box. Adobe Audition then sums the left and right channels of a stereo source file before routing it to the effects processor. When unchecked, the original signal and its respective reverberated signal remain in its own channel.



If you know that both channels are identical (that is, if they originated from a monophonic sample), you should definitely check this option to minimize processing time.

Presets

Adobe Audition comes with several Full Reverb presets. More information on presets can be found in the “Looking at the Work Area” chapter.

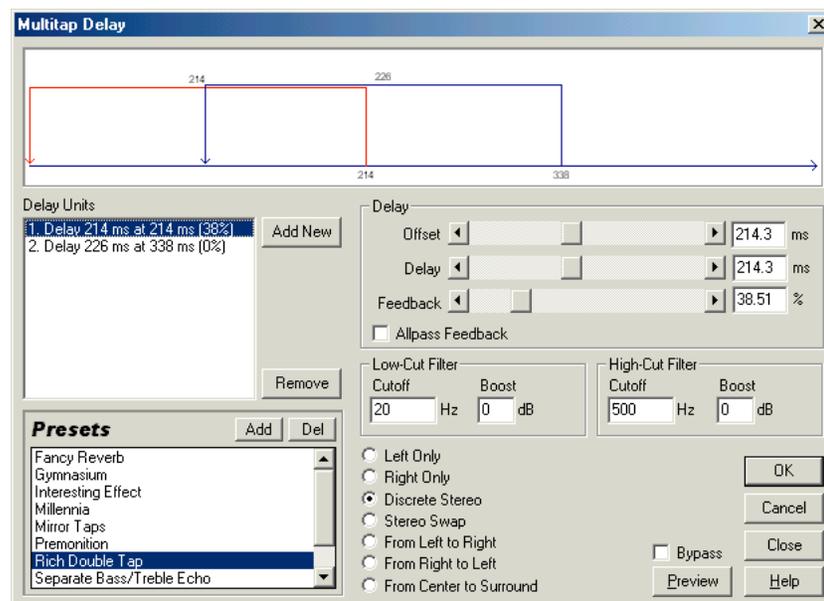
Preview

Press the Preview button to audition the current Full Reverb setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

 *Make sure that there is enough silence at the end of the waveform for the reverb trails to complete. If the reverb abruptly cuts off before it fully decays, undo the Full Reverb effect, add several seconds of silence using Adobe Audition's Generate > Silence command, and then rerun the Full Reverb effect.*

Multitap Delay

Multitap Delay can be described as a combination of Delay, Echo, Filter, and Reverb effects. Up to 10 delay units can be created, each with their own delay, feedback, and filtering settings.



Multitap Delay dialog

If one delay unit is placed inside another (as viewed in the chart above the controls) then the echo will occur more often than just once. As audio travels down the delay line (represented in the chart by the bottom horizontal arrow pointing to the right) portions at any point can be fed back into the delay line anywhere behind the given offset and at any feedback amount, with any high or low cut filter. With experimentation, you can achieve some very interesting effects.

Delay Units

This is a numbered list that displays the different Delay Units and their settings in the format: Delay [delay time] at [Offset] ([Feedback percentage]). Click on a delay unit in the list to adjust the Delay Settings sliders for that particular Delay Unit. The Add button creates a new Delay Unit with the current Delay Settings, and the Remove button deletes the currently highlighted Delay Unit.

You can create a maximum of 10 Delay Units. Each delay unit is represented in the graph above as a back-leading arrow starting at the Offset and going back the number of milliseconds stated under Delay. A single delay unit is much the same as the Echo function, but with a slightly different filtering setup. (It uses two sliding bands with variable cutoff points instead of 8 bands of filtering.)

Delay Settings

These are the characteristics used for the delay settings of each delay unit. Click on a different delay unit in the list (or add a new one) to bring up the Delay settings for that particular delay unit.

Offset Adjusts the point in the delay line from which Adobe Audition will take the audio. It is then mixed into an earlier point in the delay line, which will cause echoing. Keep in mind that it's the relative positions of the offsets of the delay units that make a difference, not their absolute position. For instance, if you have two delay units at offsets of 200 and 500, the resulting audio will sound the same if they were at 100 and 400. The difference between the offsets is what is important.

Delay Adobe Audition feeds audio back into the delay line after a certain delay. This slider adjusts that delay, in milliseconds. The result is an echo with a period of the delay given to be generated. With several delay units of varying delays added, the final echo pattern could become very complex. Very short delays give ringing or robotic sounding events. Longer delays give more distinct echoes.

Feedback •Is the percentage of the original signal to feed back into the delay line.

- If the feedback slider is set too high, definite ringing and true feedback will occur. The audio will get louder and louder until it clips and becomes distorted. However, sometimes you may want this effect, which is similar to the feedback you hear when a live microphone is set too close to a loudspeaker.
- If the feedback percentage is extremely low, then not very much of the original signal will be fed back into the loop. This results in a very subtle effect.

Allpass Feedback To help prevent the DC component from getting out of hand (the waveform tending upwards or downwards until it clips), enable the Allpass Feedback option. When turned on, audio from the destination of the delay loop is mixed back into audio from the originating delay offset. Instead of going one way (from the offset back a certain number of milliseconds), it also goes from the destination up to the source – a sort of forward feedback, or “feedforward”. This setting is handy when designing reverb effects.

Low-Cut and High-Cut Filters Audio being fed back into the delay line can also be filtered before going back in. With a low-cut filter, the low frequencies are reduced (or boosted) depending on the Cutoff and Boost settings. The High-Cut filter can also cut or even boost the high frequencies. This will make each successive echo filtered slightly differently, for interesting effects.

 When designing a reverb, it helps to cut some of the high frequencies to simulate absorption of the high frequencies by the surrounding walls.

- **Cutoff:** Frequencies below this setting are affected by the Low-Cut filter. Frequencies above this setting are affected by the High-Cut filter. Changes in the cutoff value affect the tone of the echoes, as more or less of the frequencies are affected by the filter.
- **Boost** is the amount of filtering to perform. Boost settings are usually negative, which means the audio is being cut or reduced in the affected frequency range.
 - Lower negative values result in more audio being cut.
 - Positive values result in boosted frequencies.

 When echoing, frequencies are generally not boosted. However, you can create interesting effects by entering positive values. Boosting a low-cut filter while reducing the feedback setting is identical to reducing a high cut filter and increasing the feedback setting.

Channel Buttons Several options are provided for use with stereo source audio:

- **Left Only:** The delay effect appears only on the left channel of stereo audio.
- **Right Only:** The delay effect appears only on the right channel of stereo audio.
- **Discrete Stereo:** The delay effect appears on both channels of stereo audio.
- **Stereo Swap:** The delay effect for the left channel is mapped to the right channel and vice-versa
- **From Left to Right:** The audio from the left channel is delayed and put onto the right channel
- **From Right to Left:** The audio from the right channel is delayed and placed onto the left channel.
- **From Center to Surround:** The audio from the center channel (the audio common to both left and right channels) is delayed and placed into the surround channel (the left channel inversed with respect to the right channel).

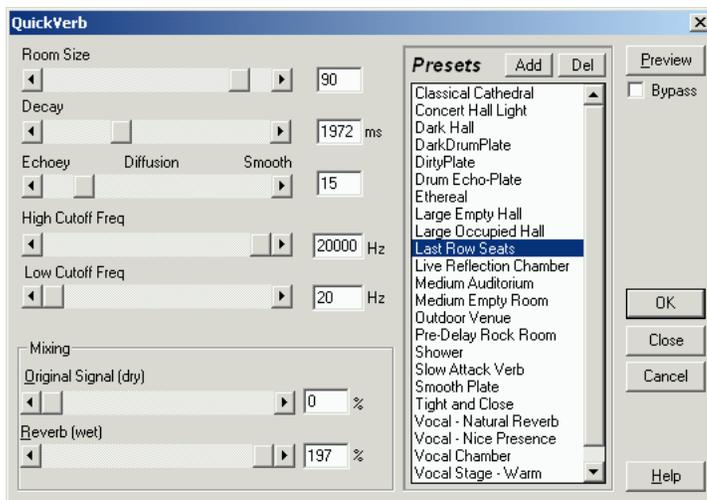
To spread out the effect in the stereo domain, at least one delay unit in the group should be cross-channel or single-channel only.

Presets Adobe Audition comes with several Multitap Delay presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Multitap Delay setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

QuickVerb

Like Adobe Audition’s two other reverb effects (Full Reverb and Reverb), QuickVerb allows you to add reverberation to your audio in order to simulate a different acoustic space. However, since it isn’t convolution-based like Full Reverb and Reverb (which require more power to run, thus increasing the processing load on your system) QuickVerb is faster to use. This means it can be altered more quickly and effectively in Adobe Audition’s Multitrack real-time environment, without the need for “locking” the effects to the track.



The QuickVerb dialog

Here are the major components of the QuickVerb dialog:

Room Size Use this slider to set the size of the room you wish to simulate, or directly enter a value in the text box to the right. Low numbers equate to a smaller room; use higher numbers to create a larger virtual space.

Decay The amount of reverberation decay is adjusted with this slider. You may also specify an exact value, measured in milliseconds, in the text box to the right.

Diffusion This setting defines the “texture” of the reverb. Move the slider to the left if you want a reverb with more echo characteristics, while positions right of center will produce a smoother reverberation with fewer echoes. A text box is provided so that you can enter a value directly.

High Cutoff Frequency This slider specifies the uppermost audio frequency at which the reverb effect can occur. You can enter the exact frequency in the text box to the right.

Low Cutoff Frequency The lowermost audio frequency at which the reverb effect can occur is set with this control. The exact frequency may be entered directly in the text box to the right.

Mixing The controls in this section determine the amount of reverb to mix in with the original audio source.

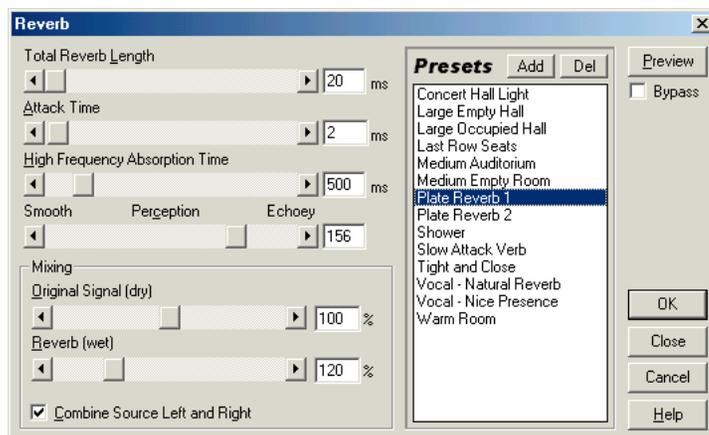
- Set the percentage of source audio to leave in the effect using the Original Signal (dry) slider, or type in the percentage directly in the text box to the right.
- Set the percentage of reverb to place in the effect using the Reverb (wet) slider, or type in the percentage directly in the text box to the right.

Presets Adobe Audition comes with several QuickVerb presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current QuickVerb setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Reverb

Reverb is an effect used to simulate acoustic space, and consists of both early reflections and echoes that are so closely spaced that they’re perceived as a single decaying sound. Reverb is different from the basic Echo effect in that the delays aren’t repeated at regularly spaced intervals.



Reverb dialog

Adobe Audition’s Reverb function can create a wide range of high-quality reverb effects. It can reproduce such acoustic or “ambient” environments as a coat closet, a tiled bathroom shower, a concert hall, or a grand amphitheater. These echoes can be spaced so closely together, and made to occur at such random times, that the signal’s reverberated “trail” will decay smoothly over time, and will have a warm and natural sound. Alternatively, initial early-reflection delays can be used to give a sense of room size, which can be either large or small, depending upon the initial delay times.

So what’s the difference between Adobe Audition’s Reverb and Full Reverb effects? Full Reverb is the newer of the two, and provides more options and better audio rendering. However, many folks still use and prefer the older Adobe Audition Reverb effect, so we’re happy to continue including it with new releases.



To simulate specific rooms that have both echoes and reverb, use Adobe Audition's Echo effect first to get the "size" of the room sound, and then use Reverb to make it sound more natural. This function can create a sense of spaciousness in a monophonic signal (one that has been recorded as or converted into a stereo audio file). Even a Total Reverb Length as little as 300 milliseconds can open up the perceived spaciousness of a dry sound (one that was recorded without any effects or reverb).

Total Reverb Length

This is the time, in milliseconds, that it takes for the reverb signal to trail off to infinity (about -96dB).

- Use values below 400 to create a small room environment.
- Use values between 400 and 800 to simulate medium sized rooms.
- Use values above 800 to simulate concert halls. For instance, setting Total Reverb Length to 3000 milliseconds will give you the reverb trails of giant amphitheaters.

Attack Time

The amount of time it takes for the reverb to gain full strength is known as the attack time. For short reverb times, the attack time should be smaller. In general, a value of about 10% of the total reverb time works well. However, you can create interesting and subtle effects by using longer attack times with shorter reverb lengths, or, conversely, by combining very short attack times with long reverb lengths.

High Frequency Absorption Time

In acoustic environments, higher frequencies tend to attenuate faster than lower frequencies. Use this slider to simulate this natural absorption so that the high frequencies are reduced in level during the reverb's decay time.

- Faster absorption times simulate rooms that are occupied and have furniture and carpeting, like night clubs or theaters.
- Slower times (especially over 1000 milliseconds) simulate rooms that are emptier, like gymnasiums or auditoriums, where higher frequency reflections are more prevalent.

Perception

This parameter gives subtle qualities to the environment by enabling you to change the characteristics of the reflections that occur within a room.

- With lower values, the reverb is "smoother" and doesn't have as many distinct echoes. In general, lower values (down to 0%) simulate smaller rooms.
- Higher values cause more variation in the reverb amplitudes and add more spaciousness by creating distinct reflections over time. Generally speaking, higher values (up to 60%) simulate large rooms.

 A setting of 100 and a reverb length of 2000 milliseconds or more creates interesting canyon effects.

Mixing

The Mixing section provides the following options:

Original Signal (dry) This setting determines how much of the unprocessed (dry) signal is mixed into the final output. In general, the more reverb you add, the lower the original signal volume should be. In most cases, a value of 90% or so should work well.

 To add spaciousness to an instrument, keep the dry signal higher, or at 100%. If you're trying to achieve a special effect with reverb, you might want to reduce the volume of the original signal. If the reverb is so great that audio begins to clip, try reducing both the dry and the reverberated signal strength.

Reverb (wet) This setting determines how much of the reverberated (wet) signal is mixed into the final effect output. In modern production, the wet amount should ordinarily be lower than the dry signal to add spaciousness to a track. However, you may want to increase this amount to simulate physical distance from the audio source (where reverb is heard in greater proportions to the original signal).

Combine Source Left and Right When working with stereo audio, there are actually two source signals (left and right), that can be placed independently. This does, however, require twice as many calculations as a single audio source, so you can check this option for faster processing.

- When this box is checked, Adobe Audition sums the left and right channels of a stereo source file before routing to the effects processor.
- When unchecked, the original signal and its respective reverberated signal remain in its own channel.

 If you know that both channels are identical (that is, if they originated from a monophonic sample), you should definitely check this option to minimize processing time.

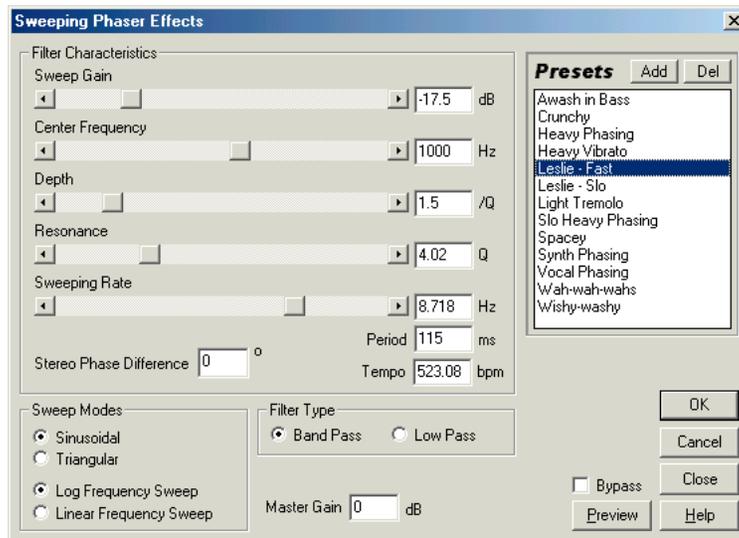
Presets Adobe Audition comes with several Reverb presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Reverb setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

 Make sure that there is enough silence at the end of the waveform for the reverb trails to complete. If the reverb abruptly cuts off before it fully decays, undo the Reverb effect, add several seconds of silence using Adobe Audition’s Generate > Silence command, and then rerun the Reverb effect.

Sweeping Phaser

Like the Flanger effect, a Phaser introduces a variable phase-shift to a split signal and recombines it, creating special effects popularized by guitarists of the 1960s. The Sweeping Phaser sweeps a notch or boost type filter back and forth about a center frequency.



Sweeping Phaser dialog

A phase is similar to a flange, except that instead of using a simple delay, frequencies are phase-shifted over time. When used on stereo files, the stereo image can be dramatically altered to create some truly interesting sounds.

Filter Characteristics

Sweep Gain This slider lets you apply variable amounts of Sweep Gain, which is the gain applied to the phased signal. Take care to avoid clipping when applying higher positive values

Center Frequency This is the frequency around which the phase will sweep. Frequencies closer to the middle of the dynamic range of the selected audio will produce more dramatic results.

Depth This setting determines the degree of phasing. The value is given as a Q value, which is a ratio of width to center frequency. Greater Depth settings cause the sweep to extend farther away from the center frequency in both directions (covering a greater frequency range), producing a wider tremolo effect.

Resonance This is the amount of phase-shift that is applied to the signal. You can think of it as a “strength” setting for the phase. The value is given as a Q value, which is a ratio of width to center frequency.

Sweeping Rate The Sweeping Rate is the speed at which the filter sweeps around the center frequency, covering the dynamic range specified by the Depth setting. Values are given in Hertz (Hz), or cycles per second, Period (milliseconds), and Tempo (beats per minute). You can adjust the rate with the slider, or enter a value directly into one of the text boxes. To have the sweep occur in time with a song, for instance, enter the BPM of the music, or a fraction of it. (Entering 240 for a song with a tempo of 120 would sweep in eighth notes, for example.)

Stereo Phase Difference This is the degree which the sweep interval is shifted between the channels of a stereo waveform. Values farther away from 0 or 360 cause the sweep to occur at increasingly distant intervals between the left and right channels. A value of 180 yields a complete difference.

Both positive numbers (from 1 through 359) and negative numbers (from -1 through -359) may be entered in this field. A low negative number (such as -45) is the same as a high positive number (315). In other words, 180 and -180 are the same value, 90 and -270 mean the same thing, and -5 and 355 refer to the same amount of degrees.

Sweep Modes These settings determine the shape of the filter sweep used.

- The Sinusoidal and Triangular options determine if the sweep will follow a sine or triangle curve. Triangle curves will tend to be sharper.
- The Log Frequency Sweep and Linear Frequency Sweep options determine if the sweep will be done in a logarithmic, constant fashion, or an even, linear fashion.

Filter Type With the Band Pass option selected, all variation is done around the frequency designated by the Center Frequency slider. If you choose the Low Pass option, the Sweeping Phaser effect will keep everything from the specified Center Frequency down.

Note: Most folks will want to use the Band Pass option, since it gives you the traditional sound of a sweeping phaser.

Master Gain This value adjusts the overall volume output. You can compensate for loss, or excessive gain, introduced by the effect by entering an overall master gain (measured in dB) to be applied to the resulting audio. The default value of 0dB represents no master gain adjustment.

Presets Adobe Audition comes with several Sweeping Phaser presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Sweeping Phaser setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

DirectX

We’d like to think that Adobe Audition has every audio effect any user will ever need, but some folks like to augment Adobe Audition’s processing power with DirectX (sometimes called ActiveMovie) audio plug-ins from other software companies. No problem! Any DirectX audio plug-in can be used from within Adobe Audition. When correctly installed, it will automatically appear in the Effects > DirectX cascading menu.

Using a DirectX plug-in works the same as any other Adobe Audition effect. Most plug-ins have a settings dialog where you can tweak the parameters of the effect before applying it to the selected region of the waveform. Some plug-ins may support real-time preview, or other particularities. You’ll need to consult the documentation provided by the plug-in manufacturer for any help with its proprietary features (that is, anything found in the plug-in’s Properties area).

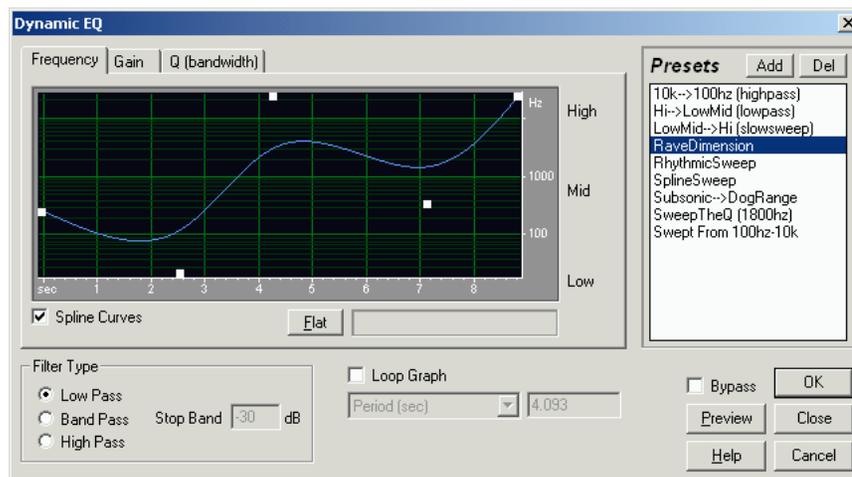
Filters

Use filters to modify a signal’s frequency content. The effects in this menu can be used to alter an audio file or selected range’s frequency content in many different ways.

Dynamic EQ

Dynamic EQ allows you to change the amount of equalization over the length of the waveform. For instance, the first part of an audio file can have a certain EQ profile (a mix of certain frequencies boosted or cut, as well as gain and bandwidth adjustments) while the middle and final portions of the audio file can have a completely different EQ profile. The Dynamic EQ dialog provides three tabbed areas for adjusting Gain, Frequency, and Q (bandwidth). Each area uses a graph control for adjusting parameters.

Note: For more information about Adobe Audition’s graph controls (such as how to add and remove control points), see the “Looking at the Work Area” chapter.



Dynamic EQ dialog

 *Dynamic EQ is especially cool when used as a real-time effect in the Multitrack View. If you add Dynamic EQ (or Dynamic Delay which has a similar principle) to the Multitrack View, you get a new envelope that determines the delay.*

Frequency Tab The graph control in the Frequency window is used to set the EQ frequency. The x-axis (horizontal) ruler represents the length of the selection, while the y-axis (vertical) ruler represents the frequency that’s boosted or cut.

Gain Tab Use the graph control in the Gain window to adjust the amount of amplitude or attenuation the Dynamic EQ effect uses. The x-axis (horizontal) ruler represents the length of the selection, while the y-axis (vertical) ruler represents the dB level that’s boosted or cut.

Q (Bandwidth) Tab The Q (bandwidth) window's graph is used to adjust the amount of Q used by the Dynamic EQ effect. The x-axis (horizontal) ruler represents the length of the selection, while the y-axis (vertical) ruler represents the frequency that's boosted or cut.

Spline Curves Check this box – which appears in each of the Gain, Frequency, and Q (bandwidth) tabbed windows – to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use spline curves, the line won't ordinarily travel directly through the control points. Instead, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat Press the Flat button – which appears in each of the Gain, Frequency, and Q (bandwidth) tabbed windows – to reset the curve to its default state of an unchanged signal, removing all Control Points.

Filter Type Three types of filters are offered:

- Low Pass passes the low frequencies while rejecting (or removing) high frequencies.
- High Pass passes high frequencies and removes low frequencies.
- Band Pass: This mode preserves a range of frequencies (that is, a band) while removing all other frequencies.

Loop Graphs When this box is unchecked, the graphs are the same length as your selection, so the dynamic changes to your equalization are graphed across the whole selection. If you check Loop Graphs, the graphs are only as long as specified in the Graph Duration field (which is enabled when Loop Graphs is checked), so they loop repeatedly over the course of the selection.

Presets Adobe Audition comes with several Dynamic EQ presets. More information on presets can be found in the "Looking at the Work Area" chapter.

Preview Press the Preview button to audition the current Dynamic EQ setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

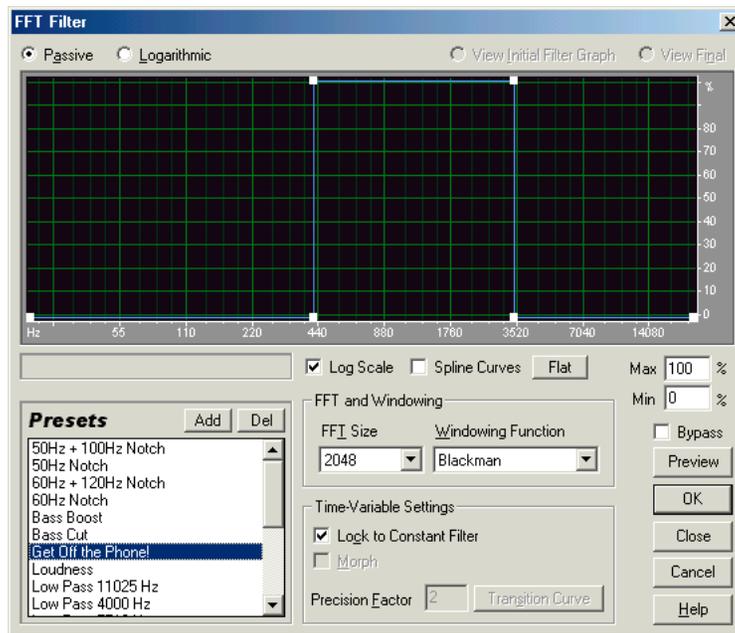
Automated When using Dynamic EQ as a real-time effect in Multitrack View, it supports parameter automation.

- Check the Automated box in the Gain section of the Multitrack View version of the Dynamic EQ dialog to draw the Gain curve using a yellow envelope on waveform blocks.
- Check the Automated box in the Frequency section of the Multitrack View version of the Dynamic EQ dialog to draw the Frequency curve using a pink envelope on waveform blocks.

Note: You must have View > Show FX Parameter Envelopes in Multitrack View turned on in order to use parameter automation.

FFT Filter

The graphic nature of the FFT (Fast Fourier Transform) Filter makes it easy to draw curves or notches for rejecting or boosting specific frequencies. Use it to create band-pass filters (which keeps only a certain band of frequencies), high-/low-pass filters (for reducing or eliminating the low or high-end frequencies), narrow-band pass filters (to simulate the sound of a telephone call), or notch filters (which eliminate very narrow frequency bands).



FFT Filter dialog

The noise level of the FFT Filter is lower than that of 16-bit samples, so when processing audio at 16-bit resolution or lower, there should effectively be no extra noise induced by the FFT Filter. For example, the stop band noise of a Blackman window will be below the -96dB mark.

 For optimal results, filter using 32-bit samples. If your source audio is 8-bit or 16-bit, try converting to 32-bit to do the filtering, and when done, you can convert back to the lower resolution with dithering. This will produce better results than processing at lower resolutions, especially if more than one transform will be performed on the audio.

Passive/Logarithmic Buttons Select Passive mode to express frequency changes (boosts or cuts) in terms of percentages, where a setting of 100% represents no change. Select Logarithmic mode when you want to express changes in terms of dB, where 0dB represents no change.

View Initial Filter Graph/View Final When the Lock to Constant Filter box isn't checked, you can choose both an Initial and a Final filter setting. Filtering will gradually go from the initial state to the final state. The rate at which the filter migrates from the initial to final settings depends on the Transition Curve settings.

Note: For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

Log Scale Choose this option to display the x axis (frequency scale) logarithmically rather than in linear fashion. When Log Scale is checked, the graph represents the frequency curve more closely to the way the ear hears sound.

- To do finer editing in low frequencies, leave Log Scale checked.
- For detailed high frequency work, or work with evenly spaced intervals in frequency, uncheck this option.

Spline Curves Check this box to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use spline curves, the line won't ordinarily travel directly through the control points. Instead, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat Press the Flat button to reset the curve to its default state of an unchanged signal, removing all Control Points.

Max/Min These boxes allow you to enter a maximum and minimum value for the graph's y axis. Adobe Audition displays the values in the ruler to the right of the graph. These settings affect the maximum boost or cut that the frequency curve represents.

FFT Size The FFT Size parameter specifies the size of the FFT to use (represented as a power of two), which can affect processing speed and quality. For cleaner sounding filters, use higher values. Recommend values are between 1024 and 8192.



Use lower values (512 or so) when tweaking your filter for faster Preview response. When you have the settings the way you want them, switch to a higher FFT size for better quality when processing.

Windowing Function The windowing function determines the amount of transition width and ripple cancellation that occurs during filtering, with each one resulting in a different frequency response curve. These functions are listed in order from smallest width and greatest ripples to widest width and least ripples.

The filters with the least ripples are also those that more precisely follow the drawn graph, and have the steepest slopes, even though they are wider, and pass more frequencies in a band-pass operation. The Hamming and Blackman filters give excellent overall results. Try different windowing functions if you these don't give you the effect you want.

Lock to Constant Filter Check this option to apply a constant filter to the waveform. Uncheck it to choose both an initial and a final filter setting. The rate at which the filter migrates from the initial to final settings depends on the Transition Curve settings.

Morph If Morph is checked, the transition from the initial filter settings to the final filter settings will actually “morph” from one to the other. If this is not checked, the settings simply change in linear fashion over time. For example, if you leave Morph unchecked and have a spike at 10kHz for the initial filter and a spike at 1kHz for the final filter, the spike at 10kHz will gradually decrease, and the spike at 1K will gradually increase over time. However, frequencies between 1kHz and 10kHz won’t be affected. If morphing is on, the spike itself will “ooze” from 10kHz down to 1kHz, passing many of the frequencies in between.

 *For a cool example of morphing, try choosing Passive mode, and set an initial curve with the first half at 100%, and the second half at zero. For the final curve, set the right 1/10th or so at 100% with the rest at zero. This selects high frequencies for the initial configuration, and low frequencies for the final configuration.*

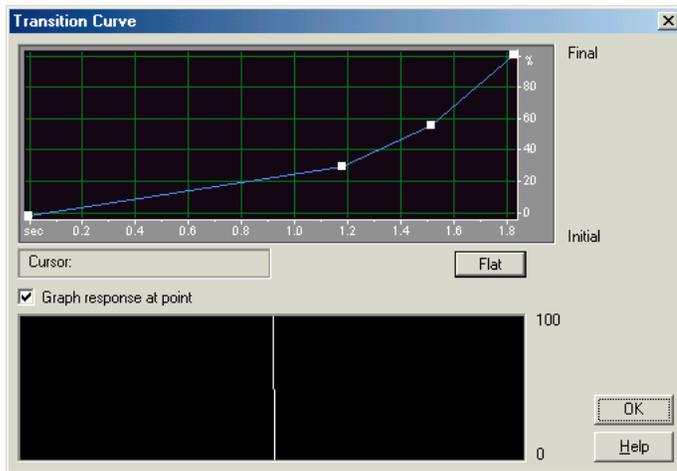
To get a nice blending from high to low, choose Morph to include all the frequency combinations between the two filters. To see exactly what’s happening as the filtering changes from the initial configuration to the final, push the Transition Curve button to view the actual settings that will be used over the duration of your selection.

Morphing is generally any technique used to transform one object into another. In the case of filter settings, it is a way to smoothly transform one setting (represented by a graph) to another by estimating all the possible combinations of the two settings. Over time, the first setting becomes the second. At some point, the setting will stop looking like the initial configuration, and start looking like the final configuration. In the filtering world, this means that frequencies between the ones selected to be filtered will also be filtered.

In migrating from the initial to the final filter configuration, the points in the “in-between” settings are just the average between the two settings. For example, a filter setting exactly between the initial and final would be the exact average of the initial and final filter settings.

Precision Factor This value determines how accurately you want to filter over time when separate initial and final settings are used. A larger number (low factor) means the filter settings will change roughly, or in chunks, from initial to final, while smaller numbers (higher factor) will make the transition much smoother. In any case, the higher the precision factor, the longer it will take to filter your selection, but the nicer it will sound.

Since the FFT function takes a large group of samples, and filters them all at once, the precision factor determines how many samples from the entire group will actually be saved. A factor of 2 means that half of the samples are saved, while a factor of 10 means that 1/10 of the samples are saved. Since there can be only one filter setting for the entire group of samples, you may want a more accurate (or smaller) setting if the EQ curve varies wildly over short periods of time.



FFT Filter Transition Curve dialog

Transition Curve Select this button to open the Transition Curve window. It displays a graphical representation of the transition from initial to final filter settings. The top graph shows time along the x axis (with the left representing the start of your sample, and the right side representing the end), and where in the transition you are allowed the y axis (with 0% representing your initial filter, and 100% representing your final filter). All points in between are a combination of your initial and final filter arrangements. The readout below the graph displays the current x, y position of your mouse.

In the Transition Curve window, select from the following options:

- The Flat button in the Transition Curve window resets the curve to its default state of an unchanged signal, removing all Control Points.
- Check this Graph response at point option to have the bottom graph update in response to your mouse position in the top graph. The bottom graph shows the filter at any given point in the transition (provided the Graph response option is checked), and corresponds directly to the position of the mouse in the top graph (click, or click and drag to update the view). Watch how the filter settings change as you move your mouse up and down in the top graph. Depending on the position you select, you can decide whether you want a morphing transition, or a linear transition. Each type of transition will give different filter settings for the points between your initial and final filter settings.

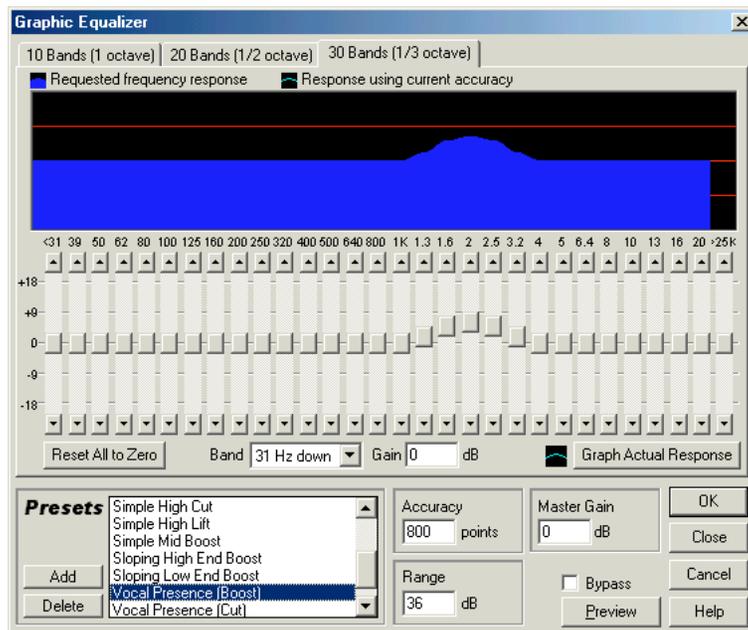
For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

Presets Adobe Audition comes with several FFT Filter presets. More information on presets can be found in the "Looking at the Work Area" chapter.

Preview Press the Preview button to audition the current FFT Filter setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Graphic Equalizer

With the Graphic Equalizer, you can boost or cut the signal at particular frequency bands, and have a visual representation of the overall EQ curve. Unlike the Parametric Equalizer, the Graphic Equalizer uses preset frequency band controls for quick and easy frequency equalization.



Graphic Equalizer dialog (30 Bands tab)

Adobe Audition's Graphic Equalizer is implemented as an FIR (Finite Impulse Response) filter, which maintains phase errors over the response curve, unlike IIR filters which can have phase error (often audible as a ringing quality). This means that to gain higher accuracy in the lower frequencies, a higher accuracy (the size of the FIR filter) must be used. The bands can be spaced at either one octave, one-half octave, or one-third octave intervals, with the appropriate Q setting such that if adjacent bands are boosted, the result has no drop outs at intermediate frequencies.

Bands Tabs The Graphic Equalizer bands are spaced at either one octave (10 band), one-half octave (20 band), or one-third octave (30 band) intervals. Select the appropriate tab to access the number of EQ bands you need. As you might expect, the 10-bands mode offers more general equalizing, while choosing 20 or 30 bands will let you zoom in on specific frequency ranges more precisely. Other than the number of bands, the controls in each tabbed Graphic Equalizer window are identical.

Reset All to Zero This button sets all sliders to 0dB so there's no equalization.

Band Use this drop-down list to select the band that's to be modified with the value entered in the Gain box to the right.

Gain This is where you enter the exact value for the gain (measured in dB) being used in the chosen band.

Graph Actual Response Because this is an FIR filter, the response may not actually match the desired equalization curve at lower accuracy levels. Press this button to see the actual response of the equalizer rendered in the window above the band sliders.

Accuracy Entering higher accuracy levels (longer FIR filters) in this text box will give better frequency response in the lower ranges.

Note: Higher values require more processing time, but you can use lower accuracy levels if you only want to equalize higher frequencies.



If you are equalizing very low frequencies, you should probably raise the accuracy. Values between 500 and 5000 points work well.

Range This field defines the range of the slider controls. Enter any value between 4dB and 180dB. As a comparison, standard hardware equalizers usually have a range of 30dB to 48dB.

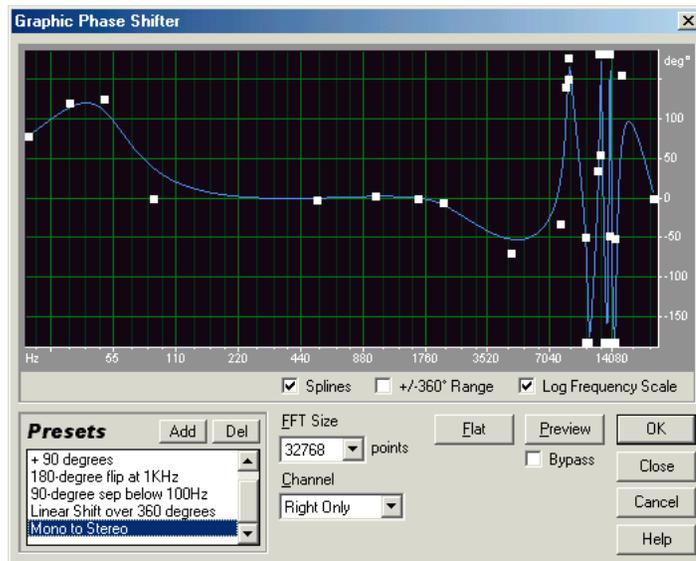
Master Gain After the EQ settings have been adjusted, the overall volume level might be too loud or too soft. You can compensate for this by entering an overall master gain to be applied to the resulting audio. The default value of 0dB represents no master gain adjustment.

Presets Adobe Audition comes with several Graphic Equalizer presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Graphic Equalizer setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Graphic Phase Shifter

The Graphic Phase Shifter lets you adjust the phase of a waveform by adding control points to a graph. The x-axis (horizontal edge) of the graph measures the waveform’s frequency content, while the graph’s y-axis (vertical edge) displays the degrees of phase to shift, with zero degrees representing no phase shift.



Graphic Phase Shifter dialog

Note: For more information about Adobe Audition’s graph controls (such as how to add and remove control points), see the “Looking at the Work Area” chapter.

💡 Most surround sound stereo decoders do different things with audio depending on the phase relationship. If you’re trying to take advantage of surround sound, try making just one channel 90 degrees out of phase. This would play the audio on all channels when surround-sound decoded. If you make the left channel 180 degrees out of phase (this is the same as just inverting one channel), it’ll play on the rear speakers when decoded.

You can create simulated stereo by creating a zigzag pattern that gets “zigzaggier” at the high end on one channel. Put two channels together that have been processed in this matter (using a different zigzag patterns for each) and the stereo simulation is more dramatic. (This would have the same effect as making a single channel twice as “zigzaggy”.)

💡 If you adjust the phase of one channel in just a small frequency band first, then use Flange or another Adobe Audition effect that depends on phase, you’ll get different results than with non-pre-phase-shifted audio.

Splines Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use Spline Curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. Use spline curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points). To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the Spline Curve will be to those points.

+/-360° Range When checked, the graph’s y-axis (vertical) ruler goes from +360 degrees (top) to -360 degrees (bottom).

Uncheck the box the have the y-axis ruler display values from +180 degrees (top) to -180 degrees (bottom).

Log Frequency Scale Check this option to represent the values of the graph's x-axis (horizontal) ruler logarithmically. You would use this mode to work at finer detail in the lower frequencies. Log scale is how the ear hears, so evenly spaced locations on the log scale represent what we hear as evenly spaced frequencies. Each octave occupies a fixed width on the display. If this option is unchecked, the values of the graph's x-axis ruler are shown linearly from 0Hz to the Nyquist. Octaves on the left take up much less width on the display than do octaves on the right.

Presets Several Graphic Phase Shifter presets are included. More information on presets can be found in the "Looking at the Work Area" chapter.

FFT Size Select the desired FFT size (measured in points) from this drop-down list. Higher sizes usually lead to more precise results, but take longer to process.

Channel Use this drop-down list to choose the channel(s) you want to apply the phase shift to.

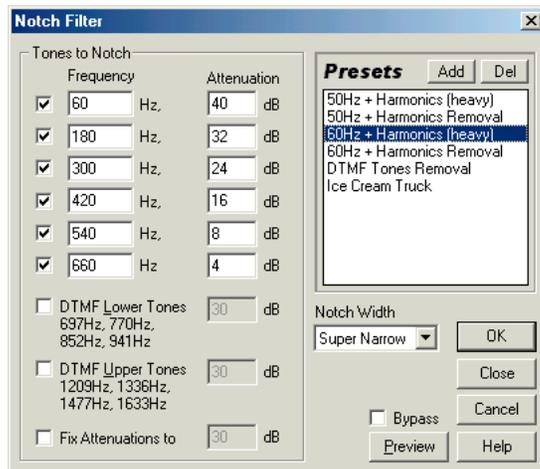
Note: Keep in mind that applying any phase shift to both channels simultaneously will result in a file that sounds exactly the same as it did before being processed when you're listening in stereo, even though it may be phase shifted all over the place. This is because we cannot hear phase directly. So for interesting effects, it's best to process a single channel.

Flat This button resets the graph so that a straight horizontal line appears at zero degrees. All control points are removed except for the ones at the end of the line.

Preview Press the Preview button to audition the current phase shift settings before you apply them. Check the Bypass box below to bypass the effect so that you can do a "before and after" comparison.

Notch Filter

This filter removes up to six user defined frequencies, in addition to standard telephone DTMF tones. Use this filter when you wish to remove very narrow band frequencies. For example, you can remove a 60Hz hum while leaving all 59Hz and 61Hz audio untouched.



Notch Filter dialog

Tones to Notch Here you may select and enter up to six frequencies to notch. Click in the checkbox to activate the box for entering a particular frequency (measured in Hz). You can apply different amounts of attenuation for each frequency by unchecking Fix Attenuations to at the bottom of the Notch Filter dialog. Then, enter the attenuation level (measured in dB) desired. Both the lower and upper DTMF tone sets (those used in the telephone system) can also be filtered as well. This is sometimes handy when preparing audio that could air on the radio. Just check which set of tones you wish to filter.

Fix Attenuations to This allows you to set attenuation levels.

- When this box is checked, all attenuation levels are the same.
- Uncheck the box to set individual attenuation levels for each of the frequencies.

Notch Width This drop-down control lets you select from three notch width presets. They range from Narrow for a second order notch filter, which can remove some side frequencies, to Super Narrow for a sixth order notch filter, which is very specific.

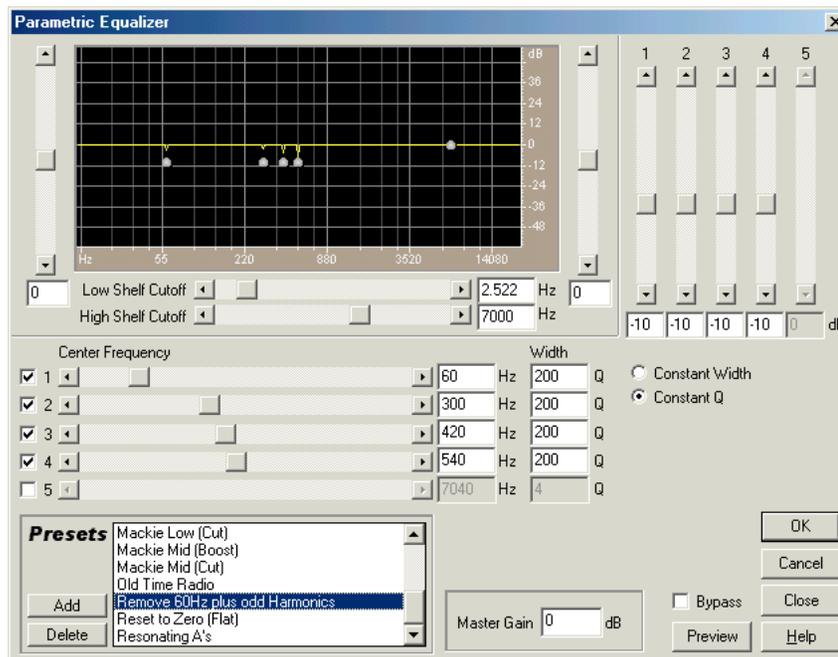
 *It's recommended that you use no larger than 30dB attenuations for a Narrow setting, no more than 60dB for Very Narrow, and no more than 90dB for Super Narrow. Any more attenuation for the given notch width will result in a wide range of neighboring frequencies being removed as well.*

Presets Adobe Audition comes with several Notch Filter presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Notch Filter setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Parametric Equalizer

Parametric equalizers are used for fine editing of frequencies within a sound. They differ from equalizers that offer a specific number of frequencies and Q bandwidths (which determines how many overall frequencies will be affected by a single, selected frequency control) in that they give you continuous control over the frequency, Q, and gain settings.



Parametric Equalizer dialog

For example, with the parametric equalizer, you can boost a broad range of frequencies that are centered around 1000 Hertz (1kHz), or you can boost the low frequencies while simultaneously inserting a 60 Hz notch filter. Adobe Audition's Parametric Equalizer uses IIR filters (second order), which are very fast and can give very precise resolution, even at the lower frequencies. For instance, it's easy to boost just the bass at 40Hz to 45Hz.

Graph The graph depicts frequency along the x-axis (left and right) and amplitude along the y-axis (up and down), with the curve representing the amplitude change at specific frequencies. Frequencies in the graph range from lowest to highest in a logarithmic fashion (evenly spaced by octaves).

Low Shelf The slider directly to the left of the graph determines the Low Shelf amplitude in dB. To increase or decrease the lows (bass) at any time, just adjust the slider, or enter a dB level in the box at the bottom of the slider.

Low Shelf Cutoff High and low pass filters cut the extreme high- or low-end of the audio spectrum. The low pass (so named because it allows the low frequencies to pass, while reducing the highs) is often used to reduce hiss, amplifier noise, and the like. To fine-tune the range of frequencies being boosted or cut, adjust the Low Shelf Cutoff slider located directly below the graph, or enter a specific frequency in the box to the right of the slider.

High Shelf The slider directly to the right of the graph determines the High Shelf amplitude in dB. To increase or decrease the highs (treble) at any time, just adjust the slider, or enter a dB level in the box at the bottom of the slider.

High Shelf Cutoff High and low pass filters cut the extreme high- or low-end of the audio spectrum. The high pass (so named because it allows the high frequencies to pass, while reducing the lows) is often used to reduce low-end rumble, hum, or other unwanted low-frequency sounds. To fine-tune the range of frequencies being boosted or cut, adjust the High Shelf Cutoff slider located directly below the graph, or enter a specific frequency in the box to the right of the slider.

Intermediate Bands You can place up to five intermediate bands into the EQ circuit. This gives you very fine control over the shape of the equalization curve. Check the box next to a slider to activate the band and its corresponding volume slider. The vertical sliders in the upper right of the Parametric Equalizer window control the amount of boost or cut. You can also enter boost/cut amount (dB) in the entry box below each slider. The horizontal sliders located below the Cutoff sliders control the center frequency at which the boost or cut will occur. A frequency may also be entered in the box to the right of the slider.

Width The Q or Width value (depending on the Constant Q/Width setting below) controls the width of the affected frequency band. With lower Q values (or greater Width values), a larger range of frequencies will be affected. If a Q value is very high (above 100), only a very narrow band will be affected, which is ideal for notch filters where only a particular frequency needs to be removed, like a 60-cycle hum.



You should be aware that whenever a very narrow band is boosted, it will tend to “ring” or resonate at the audio at that frequency. Q values of 1 to 10 are used most often for general equalization.

Constant Width/Constant Q This parameter describes a frequency band’s width. This can be a Q value (which is a ratio of width to center frequency) or an absolute width value in Hz. Constant Q is the most common setting, but you may want to use Constant Width if, for example, you want the length of ringing to be a constant, no matter which frequency is being boosted.

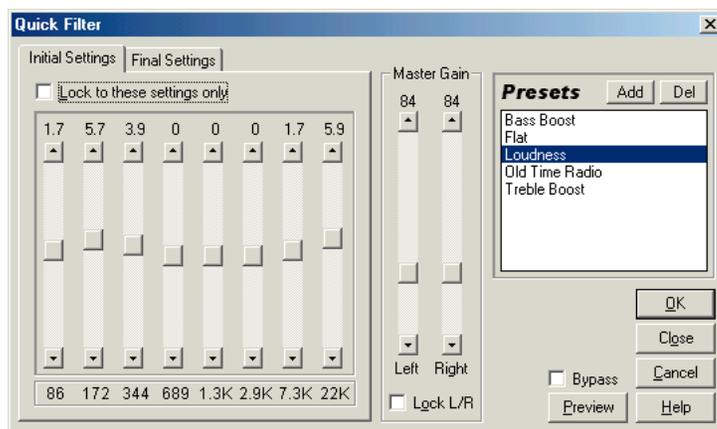
Master Gain After the EQ settings have been adjusted, the overall volume level might be too loud or too soft. This can be compensated for by entering an overall master gain (measured in dB) to be applied to the resultant audio.

Presets Adobe Audition comes with several Parametric Equalizer presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Parametric Equalizer setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Quick Filter

Quick Filter is an 8-band equalizer that can be easily customized to suit many filtering needs. It works much the same as a standard audio equalizer does, except that the bands act a bit differently. The highest frequency band will increase or decrease the high end, but it will also increase or decrease frequencies all the way down to the lowest band. It will, however, increase or decrease the high frequencies more than the low ones, of course. The effect is close to an equalizer, but not quite. Essentially, this is a quick and easy function for changing the tone of your sample to make it more pleasing to the ears.



Quick Filter dialog

Initial Settings When the Lock to these settings only box isn't checked, choose this tab to select the initial EQ settings.

Final Settings When the Lock to these settings only box isn't checked, choose this tab to select the final EQ settings.

Lock to these settings only Equalize or vary settings with this option.

- Check this box to equalize the entire selected range with the setting shown.
- Uncheck this box to select separate initial and final equalization settings, so that the selection can smoothly glide from the initial equalization setting to the final setting over the selected range. Click on the appropriate tab to change Initial and Final Settings.

Equalizer Bands Adjust these sliders to increase or decrease the frequency component specified beneath the slider. Amplitude is shown above each slider.

Master Gain This control adjusts the equalizer's overall level. Sliders are provided for both the left and right channels of stereo waveforms.



Many times when you increase the EQ frequencies of a waveform, the waveform's volume is increased as well, which could lead to clipping. Use the Master Gain sliders to reduce the level before applying the effect.

Lock L/R: This functions as a toggle to let you adjust channels together or separately.

- With Lock L/R checked, the scroll bars for the left and right channels will move together, maintaining the same settings for each channel.
- Uncheck Lock to adjust the channels separately.

Presets Adobe Audition comes with several Quick Filter presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Quick Filter setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

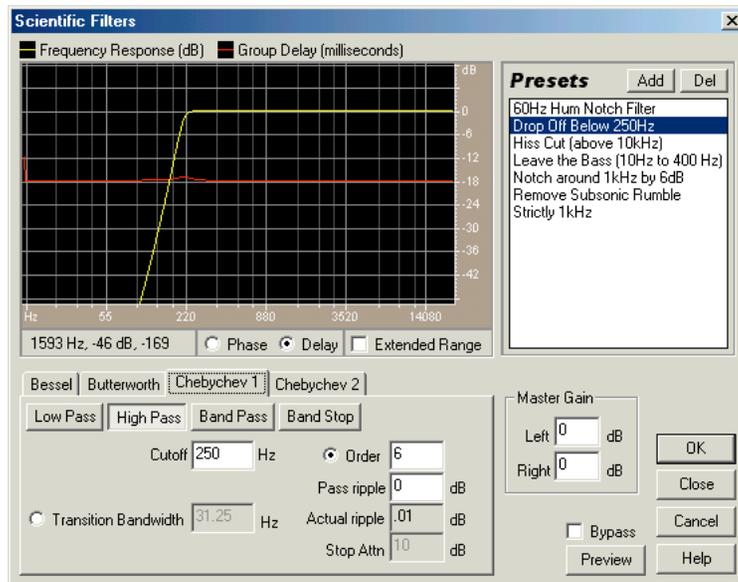
 *You can create some very interesting effects by changing the initial and final equalization settings so that they can “morph” from one curve type to another (such as having a “bass-y” curve change to a flat one at the introduction of a song).*

 *To produce a semi low-pass filter, set the higher frequency scroll bars to -30 to cut out higher frequencies. Similarly, you can create a high pass filter by reducing the lower frequencies.*

Note: Boosting any of the frequencies to very high values may lead to clipping.

Scientific Filters

Adobe Audition’s Scientific Filters provide high-order IIR (Infinite Impulse Response) filters for precise band pass, band reject (notch), or high or low pass filtering.



Scientific Filter dialog

The four most common types of higher order filters are available: Bessel, Butterworth, Chebychev 1, and Chebychev 2. Each one has different characteristics for how steep the transition bands are (at the cutoff points), and how much the filter attenuates. Butterworth is generally the best compromise between quality and desired precision.

Each filter is visualized on the Scientific Filters graph, with one line showing frequency response (measured in dB) and the other displaying either phase (measured in degrees) or group delay (measured in milliseconds), depending on whether the Phase or Delay option is selected. Increase the graph's display range by checking the Extended Range box.

Note: Some of the following functions are common to all four types of Scientific Filters, while others apply to just certain filters.

Low Pass This setting passes the low frequencies while rejecting (or removing) high frequencies. You must specify a cutoff point in order to determine the starting point at which the frequencies will be passed or rejected.

High Pass This setting passes high frequencies and removes low frequencies. You must specify a cutoff point in order to determine the starting point at which the frequencies will be passed or rejected.

Band Pass This mode preserves a range of frequencies (that is, a band) while removing all other frequencies. You must specify two cutoff points to define the edges of the band.

Band Stop Also known as a notch filter, Band Stop is the opposite of Band Pass, and will reject any frequencies within the specified range. You must specify two cutoff points to define the edges of the band.

Cutoff This parameter defines the frequency serving as a border between passed and rejected frequencies. It is at this point that the filter will switch from passing to attenuating or vice versa. In filters requiring a range (Band Pass and Band Stop), this serves as the lower frequency, while High Cutoff defines the high frequency border.

High Cutoff In filters requiring a range (Band Pass and Band Stop), this serves as the higher frequency, while Cutoff defines the lower frequency border.

Order The higher the order, the more precise the filter (with steeper slopes at the cutoff points, etc.). However, very high orders can also have high levels of phase distortion.

Transition Bandwidth Some filter types allow the specification of the width of the transition band. (Lower values have steeper slopes.) If you choose a transition bandwidth, the Order will be filled in automatically, and vice-versa. In filters requiring a range (Band Pass and Band Stop), this serves as the lower frequency transition, while High Width defines the higher frequency transition.

High Width In filters requiring a range (Band Pass and Band Stop), this serves as the higher frequency transition, while Transition Bandwidth defines the lower frequency transition.

Pass Ripple/Actual Ripple Some high-order filter types give you the ability to choose the maximum allowable amount of ripple. Ripple is the effect of unwanted boosting and cutting of frequencies near the cutoff point.

Stop Attn The Stop Band Attenuation control determines how much gain reduction is to be used when removing frequencies.

Master Gain After the filter settings have been adjusted, the overall volume level might be too loud or too soft. You can compensate for this by entering an overall master gain (measured in dB) for both the left and the right channels to be applied to the resulting audio.

Presets Adobe Audition comes with several Scientific Filter presets. More information on presets can be found in the “Looking at the Work Area” chapter.

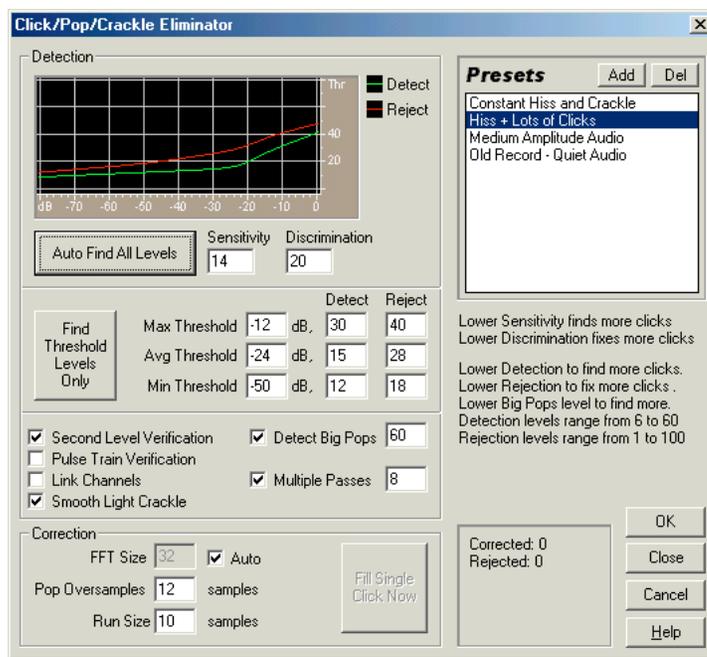
Preview Press the Preview button to audition the current Scientific Filter setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Noise Reduction

The effects in the Noise Reduction menu can be used to reduce background noise and general broadband noise without having an adverse effect on the overall quality of the originally-recorded signal.

Click/Pop Eliminator

The Click/Pop Eliminator works by searching for anomalies in the audio data that could be construed as clicks or pops (Detection), and then replacing or repairing the damaged location (Correction). If you want to clean up vinyl recordings before transferring them to CDs or another digital medium, Click/Pop Eliminator is the effect to use. It's more accurate than just cutting out the click, or replacing the data with a straight line. You can correct an entire selection or instantly remove a single click if one is highlighted.



Click/Pop Eliminator dialog

 Use a high zoom level to isolate the click.

 To speed up the repairing of single clicks, configure the F3 key to correct them when they are highlighted by first choosing the Click/Pop filter and pressing Fill Single Click Now. The F3 key then repeats the last action without bringing up the dialog box. You can also create a quick key in the Favorites menu for filling in single clicks.

 Use the Spectral View feature with the spectral resolution set to 256 bands and a Window Width of 40% to see the clicks in a waveform. See the Spectral Display area of Options > Settings > Display to adjust these parameters. Clicks will ordinarily be visible as bright vertical bars that go all the way from the top to the bottom of the display.



To hear all the clicks that were removed, save a copy of the original file somewhere, then Mix Paste it (overlap it) over the corrected audio with a setting of 100% and Invert enabled.

It may take a little trial and error to find the right settings, but the results are well worth it – much better than searching for and replacing each click individually. The parameters that make the most difference in determining how many clicks are repaired are the Detection and Rejection thresholds (the latter of which requires Second Level Verification). Making adjustments to these will have the greatest effects; you might try settings from 10 for a lot of correction, 50 for very little correction on the detection threshold, or 5 to 40 on the rejection threshold. The next parameter that affects the output most is the Run Size. A setting of about 25 is best for high-quality work. If you have the time, running at least three passes will improve the output even more. Each successive pass will be faster than the previous one.

Detection Graph

This graph shows the exact threshold levels to be used at each amplitude, with amplitude along the x-axis (left and right) and the threshold level along the y-axis (up and down). Adobe Audition will use values on the curve to the right (above -20dB or so) in processing louder audio, and settings towards the left in processing softer sections. There are separate, color-coded curves for detection and rejection.

Auto Find All Levels Using the values input in the Sensitivity and Discrimination fields as guides, the Auto Find All Levels command scans the highlighted area for clicks, and determines values for all of the Threshold, Detect, and Reject values. Five areas of audio are checked, starting at the quietest regions and moving up to the loudest.

Sensitivity Lower values entered here will find more subtle clicks. (Detected levels found when using Auto-Find All Levels will always be higher than this value.)



Try a value of 10 for lots of subtle clicks to 20 to set levels for fixing a few louder clicks.

Discrimination Higher values that are input in this field will fix fewer clicks, but will leave original audio alone more by not repairing clicks when there really aren't any.



Use values of 20 to 40 if there are relatively few clicks in the audio. Go as low as 4 if there are constant clicks.

Find Threshold Levels Only This function will automatically fill in the Max (Maximum), Avg (Average), and Min (Minimum) Threshold level settings.

Use this button to automatically set the threshold levels, before manually adjusting the Max Threshold and Min Threshold Detect and Reject values. (It's a good idea to settle on Max and Min settings first, because once they're in place, they usually don't need to be adjusted much.) After the Max Threshold and Min Threshold values have been set, enter a figure for the Avg Threshold level of about three quarters the way up between the Min Threshold and Max Threshold settings – for example, Max Threshold of 30, Min Threshold of 10, and then about 25 for Avg Threshold.

Once you've auditioned a small piece of repaired audio, you can go back and adjust the settings accordingly. For example, if a quiet part still has a lot of clicks, lower the Min Thresholds value a bit. If a loud piece still has clicks, lower the Avg or Max Threshold settings. The loud, average, and quiet portions of the audio are handled differently based on the detection and rejection settings.

Max, Avg, and Min Threshold There are three definable levels (Max, Avg, Min; given in dB) for which you can specify unique detection and rejection threshold settings. These levels are ordinarily set to the maximum amplitude, average amplitude, and minimum amplitude of the audio. For example, if your audio has a maximum RMS amplitude of -10dB, then you'd normally set the Max Threshold to -10dB. If the minimum RMS amplitude was -55dB then Min Threshold would be set to -55.

Press the Find Threshold Levels Only button to fill in these levels automatically with the maximum, average, and minimum levels. Generally, less correction is required for louder audio, as the audio itself will mask many of the clicks so that their repair is not necessary. Clicks are very noticeable in very quiet audio, so quiet audio will tend to require lower detection and rejection threshold levels.

Once the threshold levels are in place, assign appropriate Detect and Reject values to be used at each of the levels.

Detect This threshold setting, with possible values of 6 to 60, determines how sensitive the filter is in finding clicks and pops. Lower detection thresholds result in more clicks being found.

Start with a threshold to 35 for high-amplitude audio (above -15dB), 25 for average amplitudes, and 10 for low-amplitude audio (below -50dB). These settings will find most clicks, and often all of the louder clicks. If there's a constant crackle in the background of your source audio, try lowering the Min Threshold level more, or increasing the dB level to which the threshold is assigned. This value can be as low as 6, but if the setting is placed any lower, then the filter may remove sounds that are not actually clicks.

The more clicks that are detected, the more repair processing there is, which increases the possibility of distortion. With too much distortion of this type, the audio begins to sound flat and lifeless. If this happens, try setting the detection threshold rather low and clicking the Second Level Verification box, which will re-analyze the detected clicks and throw out (that is, leave unfiltered) the ones that are less likely to be clicks.



If you still hear clicks after filtering your audio, try lowering the detection threshold, but if this distorts the audio too much, try either increasing the threshold or enabling Second Level Verification.

Reject If the Second Level Verification box is checked, this value (which can be from 1 to 100) determines how many potential clicks (found using the Detection Threshold) are rejected. As with detection, lower settings result in more clicks being repaired. Increasing the rejection threshold will prevent more clicks from being repaired, as they may not be actual valid clicks.

The idea is to reject as many detected clicks as possible, but still remove all audible clicks. A good starting value for this setting is 30. If a trumpet-like horn sound has clicks in it, and the clicks are not being removed, try lowering this value to reject fewer potential clicks. If a particular sound or instrument is "fuzzing out" and getting distorted, then increase the rejection threshold to keep repairs at a minimum. (The fewer repairs that are needed to get good results, the better.)

Adobe Audition may interpret some valid audio as clicks, depending on the nature of the waveform in question. Some waveforms have sharp, albeit periodic, spikes throughout. The rejection algorithm looks to see if this click is isolated (meaning it's a real click) or has neighbors (meaning it's probably part of a desired waveform).

Second Level Verification Check the Second Level Verification box to start rejecting some of the potential clicks that were found by Adobe Audition’s click detection algorithm. In some types of audio, such as trumpets, saxophones, female vocals, and snare drum hits, some of the peaks in the natural waveform may be detected as clicks. If these peaks are corrected as clicks, the resulting audio may sound muffled. With Second Level Verification enabled, these peaks will be rejected, and not corrected after all. True clicks, on the other hand, will not be rejected.



Because this option slows down the restoration, you should first try correcting the audio with this option turned off, and only use it for sections that are very troublesome. See Rejection Threshold for more information.

Pulse Train Verification Use this option if too many clicks were detected because of a periodic waveform (like trumpet waveform peaks being considered clicks). If this option is used, some valid clicks may not be repaired. Try cleaning up audio without using this option first, and only use it if the clicks are very stubborn.

Link Channels If checked, audio from both channels is simultaneously analyzed for possible clicks. If a click is found in one channel, the likelihood that a click will be detected in the other channel is increased.

Smooth Light Crackle This option will smooth out one-sample errors when detected, and can have the effect of removing more background crackle. If this makes the resulting audio sound thinner, flatter, or more tinny, leave Smooth Light Crackle turned off.

Detect Big Pops With some audio, large unwanted events (such as those more than a few hundred samples wide) may not be interpreted as clicks by the normal detection algorithm. Check this box if you want Adobe Audition to remove these large clicks as well. Valid values are 30 to 200.

Note that a sharp sound like a loud snare drum hit can have the same characteristic as a very large pop, so you should only enable this option if you know the audio has very large pops (like a recording from a record with a very big scratch in it). If this option is enabled, and drum hits sound softer, then the onset of the drum is being considered a big pop and is being “repaired”. If this happens, try increasing the threshold a bit until the obvious pops are fixed.

Should you find that obvious, loud pops are not being fixed, then enable the Detect Big Pops option, and use settings from about 30 (which will find pops that are “kind of quiet”) to 70 or so (which will only find the very loudest pops).

Multiple Pass Some clicks may be too close together to be repaired effectively, so going over the audio with another pass will often catch those clicks that weren’t found the first time around. If this option is checked, then any number of passes (up to 32) will automatically be performed. Adobe Audition will perform fewer passes if no more clicks are found and all detected clicks have been repaired. Generally, about half as many or fewer clicks are repaired on each successive pass. When using multiple pass mode, a higher detection threshold may lead to fewer interventions in repairing, and increase the quality of the result somewhat while still removing all clicks. In the same vein, higher rejection thresholds may lead to better results.

Correction

Set the following correction options.

FFT Size Is the size of the FFT (Fast Fourier Transform) that will be used to repair the clicks, pops, and crackle. In general, it's a good idea to just let the program itself decide on good FFT size choices by checking the Auto box. However, you may wish to override the automatic setting and give a specific FFT size (from 8 to 512) to use for click repair when processing some types of audio. A good starting point is an FFT Size of 32. If the clicks are still quite audible, try increasing this value to 48, then 64, etc. The higher the FFT Size, the slower the correction will be, but with potentially better results. However, if the value is too high, there may be some extra unwanted low frequency distortion, which sounds "rumbly".



If you're repairing clicks one at a time using the Fill Single Click Now button, then a high FFT size (128 to 256) should work rather nicely.

Pop Oversamples Is used to give the clicks a little extra buffer space when being repaired. When a potential click is found, its beginning and end points are marked as closely as possible. The Pop Oversamples value (which can go from 0 to 300) will expand that range, and consider more samples to the left and right of the click as actually being part of the click.

If clicks aren't being corrected fully enough (for example, if clicks become quieter but are still evident), try increasing the Pop Oversamples value. Start with a value of 8, and increase it slowly to as much as 30 or 40. Audio that doesn't contain a click shouldn't change very much if it's corrected, so this buffer area should remain mostly untouched by the replacement algorithm.

Increasing the Pop Oversamples value will also force larger FFT sizes to be used when the Auto box is checked. This setting may remove clicks more cleanly, but if set too high, audio will start to distort where the clicks were removed.

Run Size Sets a ceiling on the potential distance between two clicks. Adobe Audition's click repair algorithm may not work well on two consecutive clicks that are extremely close together. The Run Size setting (which can go from 0 to 1000) determines that any two clicks closer together than the number of samples specified in the Run Size box will be treated as one.

A good starting point would be to set this value somewhere around 25 (or half the FFT size if you don't have the Auto box next to FFT Size checked). If the Run Size value is too large (over 100 or so) then the corrections may become more noticeable, as very large blocks of data are repaired at once. If you set the Run Size too small, then clicks that are very close together may not be repaired completely on the first pass.

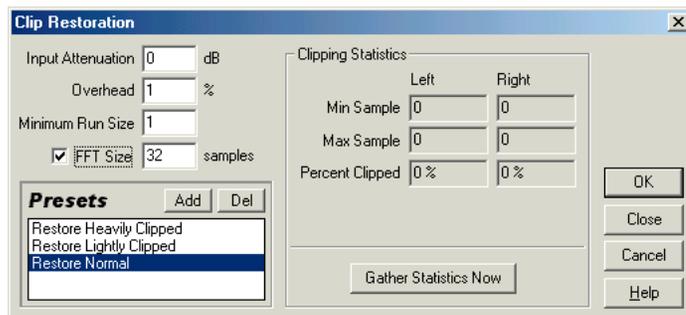
Fill Single Click Now You can correct a single click by highlighting it and pushing the Fill Single Click Now button. This is the "manual override" for click restoration. If the Auto box is checked next to FFT Size, then an appropriate FFT size will be used for the restoration based on the size of the area being restored. Otherwise, settings of 128 to 256 work very well when filling in single clicks. Once a single click is filled, the F3 (Repeat Last Command key) key will be set to repeat this action. So, to fill in another single click in highlighted audio, just press F3.

Previously Corrected/Rejected After repairing a click, re-enter the Click/Pop/Crackle Eliminator window to see the number of previously corrected clicks in the lower right hand corner of the dialog. This area will also show the number of rejected clicks that would have been corrected if the Second Level Verification box was not enabled.

Presets Adobe Audition comes with several Click/Pop/Crackle Eliminator presets. More information on presets can be found in the "Looking at the Work Area" chapter.

Clip Restoration

Adobe Audition’s Clip Restoration filter can remove the evil occurrences of clipping from your waveform. It does this by actually “filling in” the clipped segments of the waveform.



Clip Restoration dialog

Clipping is a phenomenon that occurs in digital audio when the amplitude value of a signal exceeds the maximum level that can be represented by the current bit resolution (i.e., 256 steps in 8-bit audio). This can happen when your source recording levels are simply too high. Clipping causes the signal to distort, and appears in the waveform display as a “chopping-off” of the top or bottom of the waveform. The audible result is a static-like distortion.

Note: Don’t use DC Bias Adjust on any audio you wish to run the Clip Restoration filter on, because DC bias adjusted material modifies the clipped area and makes it unrecognizable as being clipped. (You can do a DC adjust after de-clipping, of course.)

Note: You can monitor clipping occurrence during recording or playback by watching the level meters under the waveform display. If clipping occurs, the box(es) on the far right of the level meters will light up.

Input Attenuation This parameter, measured in dB, specifies the amount to amplify the signal by before processing.

Overhead This is the amount of play there is, measured in percentage, for detecting clipped regions. A value of zero will consider only perfectly horizontal line-type clipping to be “clipped.” A value of 1% will allow the horizontal line to vary up to 1% away from maximum amplitude.



Values of about 1% work best since they catch almost everything, which leads to a more thorough repair.

Minimum Run Size This value specifies the length of the shortest run of clipped samples to fix. A value of 1 will fix all samples that seem to be clipped, while a value of 2 won’t fix a clipped sample if it’s not followed or preceded by another clipped sample.

FFT Size Set an FFT (Fast Fourier Transform) Size, measured in samples, if your audio is severely clipped due to, for example, too much bass. In this case, you want to try and estimate the higher frequency signals in the clipped areas. Using the FFT Size option in other situations may help with some types of clipping. (Try a setting of 40 for normal clipped audio.) However, in general you can leave the FFT Size box unchecked. When FFT Size is disabled, Adobe Audition uses spline curve estimation.

Clipping Statistics These fields show the minimum and maximum sample values found in the current selected regions, as well as the percent of samples clipped based on that data.

Gather Statistics Now Pressing this button will fill in the Clipping Statistics fields with information for the current selection or file.

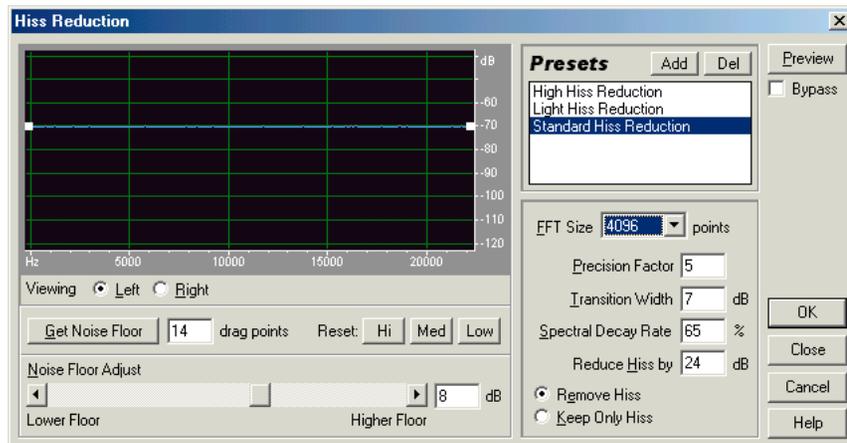
Presets Adobe Audition comes with several Clip Restoration presets. More information on presets can be found in the “Looking at the Work Area” chapter.

To restore clipped audio while retaining amplitude:

- 1 Convert the file to 32-bit for more precise editing.
- 2 Run Clip Restoration with no attenuation.
- 3 Run Hard Limiter with no Boost, but a Limit of about -0.2dB to bring the clipped audio back into normal range.
- 4 Convert back to 16-bit, if needed.

Hiss Reduction

Use Hiss Reduction to literally carve out annoying hiss from cassette recordings, record albums (after using Click/Pop Removal), or microphone recordings. It removes all audio in all frequencies that are below a certain threshold (which is generally the noise floor or hiss level). If audio has a constant background hiss, then that hiss can be “noise gated” or removed completely. Any audio in any frequency band louder than the cutoff level will remain untouched.



Hiss Reduction dialog

 To get rid of other types of noise, try Adobe Audition's Noise Reduction effect.

To use Adobe Audition's Hiss Reduction:

- 1 To use, choose an FFT (Fast Fourier Transform) Size first.
- 2 Then either reset the noise floor graph to Hi (High), Med (Medium), or Low, depending on the estimated noise level of the recording. Or, find a section relatively free of audio that contains mostly hiss and click Get Noise Floor.
- 3 Finally, choose how much to reduce hiss by, and try a few times reducing hiss from a short sample with different noise floor modifier settings.

When you're satisfied with the results, process the entire area you want to remove hiss from.

Graph Area The Hiss Reduction graph area depicts what the noise floor looks like, with frequency along the x-axis (left and right) and amplitude, or noise floor, along the y-axis (up and down). Thus, the line on the graph represents where the estimated noise floor exists for each frequency in the source audio. This information will be used to determine if audio is hiss or desirable audio data.

The actual value used when performing hiss reduction will be a combination of the graph and the Noise Floor Adjust slider, which is used to shift the estimate noise floor reading up or down for fine tuning.

Note: For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

Viewing Left/Right Select the Left button to view the left channel of the waveform in the Hiss Reduction graph. The Right option allows you to display the waveform's right channel.

Get Noise Floor This is the most powerful feature of Hiss Reduction. If a good estimate of what the noise floor looks like (i.e., how it's shaped) is graphed, Adobe Audition's Hiss Reduction algorithm can more effectively remove just hiss while leaving regular audio untouched.

Optimally, you should click on Get Noise Floor with a section of just hiss highlighted in your waveform. This will design a graph that most accurately reflects the noise floor. If no such area can be found in the waveform, select an area with the least amount of music or other desired audio. For best results, choose an area of audio that has the least amount of high frequency information. (If viewing in spectral view, the area will not have any activity in the top 3/4ths of the display.)

After the noise floor has been identified, you may need to skew the graph on the left-hand side (representing the lower frequencies) downward some, so as to make the graph as flat as possible. If music was present at any frequency, the dots on the graph around that frequency will be higher than they should be, and should be moved down.

Drag Points Enter the number of drag points (also known as control points) you want Get Noise Floor to add to the Hiss Reduction graph.

Reset (Hi, Med, Low) These three Reset buttons will clear the noise floor graph. Press Hi to assume the noise floor is at -50dB (if hiss is very loud). Press Med to set the floor to -70dB (average hiss), or Low to set the floor to -90dB (if there isn't much hiss at all). After pressing a Reset button, fine tune the noise floor setting using the Noise Floor Adjust slider until the appropriate amount of hiss reduction and quality level is achieved.



For quick, general-purpose hiss reduction, a complete noise floor graph isn't always necessary. In many cases, you can simply reset the graph to an even level and manipulate the Noise Floor Adjust slider.

Noise Floor Adjust Use this slider to specify the level of the noise floor.

- The more the slider moves to the left, the lower the noise floor will be (up to -40 db).
- The more the slider moves to the right, the higher the noise floor will be (up to 40 db).

FFT Size Specify a transform size. In general, sizes from 3000 to 6000 work best.

- Lower FFT sizes (2048 and below) will have better time response (less swooshing before cymbal hits for example) but poorer frequency resolution, making things sound hollow or flanged.
- Higher FFT sizes (12000 and above) may cause swooshing effects, reverb, and long drawn out background tones, but be very accurate in the frequencies processed.

Precision Factor This value determines the accuracy of hiss reduction in the time domain and affects how quickly spectral components below the previous hiss level decay. (See Spectral Decay Rate.) Usual settings range from 7 to 14.

- Larger values generally produce better results, but at slower processing speeds. Values higher than 20 or so will not ordinarily improve quality any further.
- With lower precision factors, you may get a few milliseconds of hiss before and after the louder parts of audio; with higher values, this is reduced.

Transition Width This setting, measured in dB, produces a slow transition in hiss reduction instead of dropping off suddenly from no reduction to the reduced hiss level. Values of 5 to 10 usually achieve good results.

- If the Transition Width value is too small, other background artifacts may be heard, such as little “tinkles” in the background.
- If the Transition Width value is set too high, some hiss may remain after processing.

Spectral Decay Rate When audio above the noise floor is encountered, more audio in the same frequency band is assumed to follow. With low Spectral Decay Rates (measured in percent), less audio is assumed to follow, and the carving function will cut more closely in time to the frequencies being kept. Values of 40% to 75% work best.

- With Spectral Decay Rates too low, background bubbly effects might be heard, and music may start to sound artificial.
- With Spectral Decay Rates too high (above 90%), unnaturally long tails and reverbs might be added.

Reduce Hiss By When audio components are found below the given noise floor settings, they are reduced up to this amount, measured in dB.

- With lower reduction amounts (below 6dB) not as much noise is removed, but the original audio signal stays relatively undisturbed.
- With higher levels (especially above 20dB) dramatic hiss reduction can be achieved, but the remaining audio may start to be distorted.

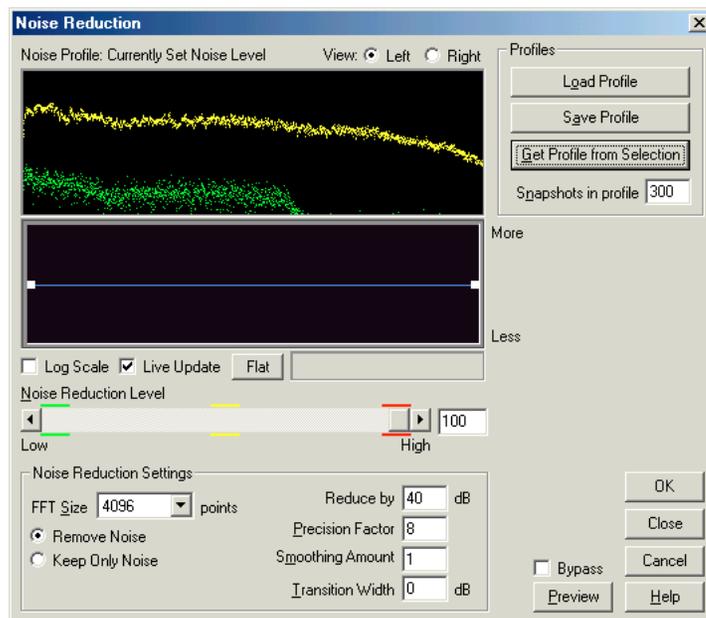
Remove Hiss/Keep Only Hiss: For normal operation, choose Remove Hiss. If for some reason you want to extract the hiss only, choose Keep Only Hiss. All the audio will be removed, leaving only hiss.

Presets Adobe Audition comes with several Hiss Reduction presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Hiss Reduction setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Noise Reduction

Adobe Audition’s Noise Reduction function can dramatically reduce background and general broadband noise with a minimal reduction in signal quality. It can also remove tape hiss, microphone background noise, 60 cycle hum, or any noise that is constant throughout the duration of your waveform.



Noise Reduction dialog

The amount of reduction required depends upon the type of background noise, and the allowable loss in the quality of the signal that is to be kept. In general, you should be able to attain increases in Signal to Noise ratios of 5dB to 20dB. (Noise is reduced 21dB and signal 1dB for example). Before you can perform noise reduction, you must load or get a noise profile first. This is done by pressing the Load Profile or Get Profile from Selection buttons.

Note: If hiss is what you want to remove, try Adobe Audition's Hiss Reduction effect instead.

View Left/Right Click to view either the left channel noise profile or the right channel noise profile. The amount of noise reduction is always the same for both channels. To perform separate levels of reduction on each channel, edit the channels individually instead of both channels at once.

Noise Profile Plot The yellow points in the profile plot represent the amount of noise reduction that will be performed at any particular frequency. Move the Noise Reduction Level slider up or down to move the entire yellow region up or down.

Load Profile Press this button to load any previously saved noise profile. You can load any .fft file that Adobe Audition has saved.

Note: A noise profile can only be used on a sample of the same type as when saved. In other words, a 44K stereo 8-bit sample is not compatible with a 22 kHz mono 16-bit profile. Bear in mind, however, that because noise profiles are so specific to the recording environment of the waveform in question, even if the sample types are compatible, a profile for one type of noise is likely to not produce good results when used for another type of noise. Even if the audio samples were recorded with the same microphone, if the recording environment is different, the type of background noise could be different.

Save Profile Once the noise level is set, you can save the noise profile as an .fft file. This file will contain information on sample type, FFT (Fast Fourier Transform) size, and three sets of FFT coefficients: one for the lowest amount of noise found, one for the highest amount, and one for the power average.

Get Profile from Selection Press this button to extract a noise profile from the highlighted selection. Try to select a portion of the waveform that has no important signal in it (i.e., it only has background noise), and then press Get Profile from Selection. Adobe Audition will gather the statistical information about the background noise, after which you can remove all noise of this type from your waveform.

Note: If the selection used for gathering the noise level is too small, then the Get Profile from Selection button won't be enabled. Reduce the FFT Size or select a larger section of noise to sample.



If you can't find a longer noise section, use Copy and Paste to create one. Highlight and copy the short noise sample. Then, paste this new instance at the end of original highlighted sample. Repeat this process until the sample has been extended long enough for the Get Profile from Selection button to be enabled. You can then perform as many Undos as necessary to remove the extra samples you've pasted.

Snapshots in profile The number in this field describes how many snapshots of noise to take in the highlighted interval when the Get Profile from Selection button is pressed. The larger this number, the more accurate the statistical data is. A value of 64 is usually sufficient.

You'll notice that using very small numbers of statistical samples will greatly affect the quality of the various noise reduction levels. With more samples, a noise reduction level of 100 will likely cut out more noise, but also cut out more original signal as well. However, a low noise reduction level with more samples will also cut out more noise, but likely will not disrupt the intended signal.

Reduction Graph Use the Reduction Graph to set the amount of noise reduction at certain frequency ranges. For example, if you need noise reduction only in the higher frequencies, adjust the chart to give less noise reduction in the low frequencies, or alternatively, more reduction in the higher frequencies.

The graph depicts frequency along the x-axis (left and right) and the amount of noise reduction (in percent) along the y-axis (up and down). If the line is set to Flat (which is accomplished by pressing the Flat button), then the amount of noise reduction used is based on the noise profile exactly. The readout below the graph displays the frequency and adjustment percentage at the current position of your mouse.

Note: For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

Log Scale You can view the chart of the noise profile in either linear or logarithmic fashion.

- Check Log Scale to divide the graph evenly into 10 octaves.

- Uncheck Log Scale to divide the graph linearly, with each 1000kHz (for example) taking up the same amount of horizontal width.

Live Update Select or deselect this option:

- When you check this option, the noise profile plot updates as control points are moved around on the graph.
- With Live Update unchecked, the profile plot is only updated when a control point is placed in its new location.

Noise Reduction Level Use this slider to manually adjust the amount of noise reduction (measured in percent) to be applied to the waveform or selection. You may also enter the desired amount directly in the text box to the right of the slider.

***Note:** Depending on the original waveform and the type of noise removed, high noise reduction levels can sometimes cause the remaining audio to have a “flange-y” or “phase-y” quality. If this happens undo the effect and try a lower setting.*

Noise Reduction Settings Set the following options:

- **FFT Size:** This parameter causes the most drastic changes in quality. It determines the number of individual frequency bands that are analyzed. The noise in each frequency band is treated separately, so the more bands you have, the finer frequency detail you get in removing noise. For example, if there’s a 120Hz hum, but not many frequency bands, frequencies from 80Hz on up to 160Hz may be affected. With more bands, there’s less spacing between bands, so the actual noise can be detected and removed with more precision. However, with too many bands, time slurring occurs, which can make the result sound reverberant or “echoey” (with pre- and post-echoes). So the tradeoff is frequency resolution vs. time resolution, with lower FFT sizes giving better time resolution and higher FFT sizes giving better frequency resolution. Good settings for FFT Size range from 4096 to 12000.
- **Remove Noise/Keep Only Noise:** For normal operation, choose Remove Noise. If for some reason you want to extract the noise for other purposes, choose Keep Only Noise. All the audio will be removed, leaving only noise.
- **Reduce by:** Lowering this value (measured in dB) may help reduce some bubbly background effects. Settings between 5dB and 100dB are recommended.
- **Precision Factor:** This value affects distortions in amplitude. With values of 3 or less, the FFT is performed in giant blocks that aren’t very continuous between the blocks. This means that after each block is processed, there can be a drop or spike in volume at the interval between blocks. Values of 5 and up work best. Beyond 10 or so, there is no noticeable change in quality-only in the time it takes to compute. Try using 5 or 7, as odd numbers are best for symmetric properties.
- **Smoothing Amount:** This takes into account the standard deviation, or variance, of the noise signal at each band. Bands that vary greatly when analyzed (such as white noise) will be smoothed differently than constant bands (like a 60 cycle hum). Generally, increasing the smoothing amount (up to 2 or so) will reduce the “burbly” background artifacts at the expense of raising the overall background broadband noise level.

- **Transition Width:** This setting determines how sharp the division is between what is considered noise and what should be kept. For instance, with a Transition Width of zero, a sharp, noise gate-type curve is applied to each frequency band. If the audio in the band is just above the threshold, it stays; if it's just below, it's truncated to silence. Conversely, you can specify a range over which the audio will fade to silence based upon the input level. For example, with a transition width of 10dB, and a cutoff point (scanned noise level for the particular band) of -60dB, then audio at -60dB would stay the same, at -62dB it would be reduced some (to about -64dB), and so on until audio at -70dB would be removed entirely. Again, if the width is zero, then audio just below -60dB is entirely removed, while audio just above it would remain untouched. Negative widths simply go about the other side, so in the above example, a -10dB width would have ranged from -60dB to -50dB.

Preview Press the Preview button to audition the current Noise Reduction setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

 You can reduce the noise incurred by the sound board's circuitry during recording--just record a second of silence before whatever you want to record and tell the noise reducer to remove the sound of that silence in order to further reduce the noise level by up to 10dB.

Note: Noise reduction works best on 16-bit and 32-bit samples. Although it will work with 8-bit audio, it's impossible to get the noise level to less than about -45dB – if even that. Noise at -45dB is very audible, as owners of 8-bit sound cards can attest. You can get much better results by converting to 16 bits first, reducing the noise, and downsampling back to 8 bits, than by reducing noise in 8-bit samples directly.

 You'll get better results with Noise Reduction if the original signal is centered. To center a signal, highlight it and choose the Center Wave preset from the Effects > Amplitude > Amplify function. Centering the wave adjusts the DC offset to zero. If the wave isn't centered, you may hear audible clicking in quiet parts after processing. Because centering takes out all frequencies below about 16Hz, it's completely safe to do without any ill side effects.

 You can generate interesting effects by choosing valid "foreground" audio as your profile "footprint" rather than background noise or hiss. For example, within a vocal line, you can select the vowel "O" to be used as the profile. Processing this vocal file (remember, it's best to process a copy of the original file) will then reduce or eliminate the "O" sounds, thereby creating a rather wild effect.

 Great effects can be generated by setting the noise level to some valid signal component in the waveform, and not the background noise. Whatever frequencies are present in the highlighted selection when Set Noise Level is chosen will be removed when the reduction level is set to 100.

Here's the most common way to use Adobe Audition's Noise Reduction function:

- 1 Record or open a piece of music that has tape hiss or other background noise.
- 2 Select a short range (at least one half second) that contains only the noise that you'd like to reduce.
- 3 Choose the Effects > Noise Reduction > Noise Reduction option and click on Get Profile from Selection to sample the noise.
- 4 Click on Close.

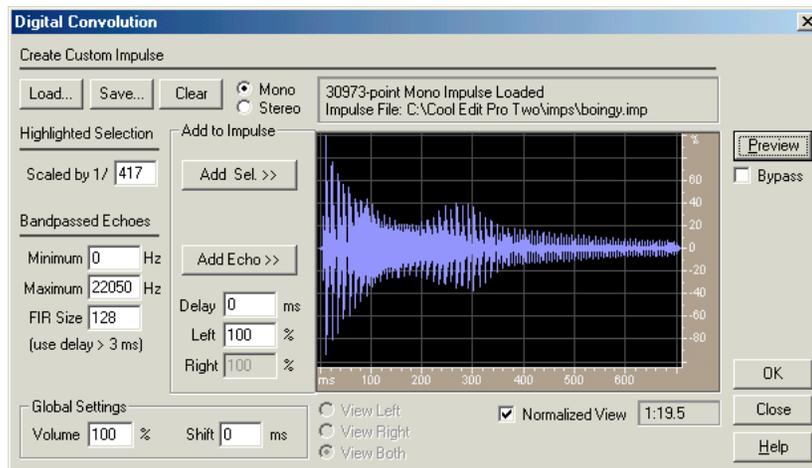
- 5 Select the entire file (or the portion you want to noise-reduce).
- 6 Go to Effects > Noise Reduction > Noise Reduction again.
- 7 Click on OK to perform the noise reduction.

Special

The options in this menu can be used to introduce some processing effects that are both innovative and wild-n-wacky. Have fun and experiment.

Convolution

Convolution is the effect of multiplying every sample in one wave or impulse by the samples that are contained within another waveform. In a sense, this feature uses one waveform to “model” the sound of another waveform. The result can be that of filtering, echoing, phase shifting, or any combination of these effects. That is, any filtered version of a waveform can be echoed at any delay, any number of times.



Convolution dialog

For example, “convolving” someone saying “Hey” with a drum track (short full spectrum sounds such as snares work best) will result in the drums saying “Hey” each time they are hit. You can build impulses from scratch by specifying how to filter the audio and the delay at which it should be echoed, or by copying audio directly from a waveform.

With the proper impulses, any reverberant space can be simulated. For example, if you have an impulse of your favorite cathedral, and convolute it with any mono audio (left and right channels the same) then the result would sound as if that audio were played in that cathedral. You can generate an impulse like this by going to the cathedral in question, standing in the spot where you would like the audio to appear it is coming from, and generating a loud impulsive noise, like a “snap” or loud “click”. You can make a stereo recording of this “click” from any location within the cathedral. If you used this recording as an impulse, then convolution with it will sound as if the listener were in the exact position of the recording equipment, and the audio being convoluted were at the location of the “click”.

Note: An “impulse” is the data by which every other sample in your waveform will be multiplied. If the impulse is a single sample of a full volume “tick”, then the convolution of that impulse with any audio data will just be that audio data itself. If that “tick” is at half volume, then the original audio data will be reproduced at half volume.

If there are several ticks, descending in amplitude over time, such as one tick every 100 milliseconds, and half as loud as the previous tick, then the result of convolution with some audio will be that sound echoed with 100ms between each echo, and each echo at half the volume of the previous echo.

Note: To send any portion of unprocessed “dry” signal back out, simply add a full spectrum echo at 0 milliseconds. The Left and Right volume percentages will be the resulting volume of the dry signal in the left and right channels.



Another interesting use for convolution is to generate an infinite sustained sound of anything. For example, one singing “aaaaaah” for one second could be turned into thousands of people singing “aaaaaah” for any length of time by using some dynamically expanded white noise.



To get a feel for Convolution, load up and play with some of the sample Impulse files (.imp) that come with Adobe Audition. You can find them in the /IMPS folder inside of the folder where you have installed Adobe Audition, or on the Adobe Audition CD.

Create Custom Impulse Use these options to create new impulses.

- **Load:** Press this button to load a previously saved impulse (such as the ones that come with Adobe Audition.) The sample rate of an impulse will affect the outcome of convolution. For example, if an impulse was created at 44100Hz, and reloaded and used on a 22050Hz file, then everything will be stretched out 2:1. All filtered echoes will be at half the frequency, and all delays will be twice as long.
- **Save** saves an impulse for later recall.
- **Clear** clears the impulse completely.
- **Mono/Stereo:** Mono impulses work with either mono or stereo data (the left and right channels will be convoluted with the same impulse). Stereo impulses convolute the left and right channels separately.

Highlighted Selection Any audio can be added to an impulse directly. The scaling factor to use when adding a highlighted selection to an impulse determines its volume. By default, Adobe Audition automatically estimates a good starting value for you. Lower this value to increase the amplitude of the impulse.

Bandpassed Echoes Use these options to add echo at any frequency to the impulse.

- **Minimum:** When adding bandpassed echoes, this is the lower cutoff frequency of the echo. For example, to echo just the range from 500Hz to 1000Hz, enter 500Hz for the minimum value.
- **Maximum:** When adding bandpassed echoes, this is the upper cutoff frequency of the echo. For example, to echo just the range from 500Hz to 1000Hz, enter 1000Hz for the maximum value.
- **FIR Size:** This parameter sets the size of the FIR filter to use to generate the filtered echo.

***Note:** Adobe Audition recommends a minimum delay (below the FIR entry box) when you add this echo. If you use a smaller delay than that suggested, the echo may contain more frequencies than you want. You can ignore this delay for full spectrum echoes, as they are just single sample ticks in the impulse.*

Add to Impulse These settings let you add features to impulses.

- **Add Sel:** This setting adds the currently highlighted selection to the impulse at the delay and left/right volumes specified. You can add as many selections of actual audio as you like.

***Note:** You can make any audio data part of an impulse by first highlighting the audio and then using the Add Sel. >> button. Ordinarily, you should first scale down any such selection to a lower volume; otherwise the convolution will come out extremely loud.*

- **Add Echo:** Pressing this button will add the bandpassed echo to the impulse at the delay and left/right volumes specified. You can add as many echoes as you like.

***Note:** To add a tick at any volume, enter the volume percentages after the Left and Right settings, and the Delay at which the tick should appear. This will cause an echo of the specified volume to occur at the given delay after convolution. Besides just echoes, you can add filtered versions of echoes by entering the minimum and maximum frequencies to echo.*

To echo all frequencies outside the range, add a full spectrum echo (e.g. from 0Hz to 22050Hz) at a specific delay, then add another echo at the same delay, but with different minimum and maximum values, and the opposite Left and Right percentages (e.g. at -100% instead of 100%).

- **Delay:** This parameter sets the amount (in milliseconds) by which samples are delayed. For pre-echoes, place at least one echo that is full spectrum (Minimum=0Hz, Maximum=22050Hz) at a larger delay (like 1000ms). Then any echo placed before 1000ms will be a pre-echo.
- **Left :** This setting specifies the left volume (in percent) at which to make the echo or selection being added to the impulse.
- **Right:** This setting specifies the right volume (in percent) at which to make the echo or selection being added to the impulse.

Global Settings These settings effect how the given impulse will be used during convolution.

- **Volume:** If the convoluted audio comes out too soft or too loud, enter an appropriate value to amplify or attenuate the audio.
- **Shift:** This value is generally set to one-half the FIR Size when building impulses from scratch, to compensate for the delay incurred when the minimum delay used was only one-half the FIR size. Adjust this shift setting if you find that the convoluted audio is migrating too far to the right with respect to the original audio.

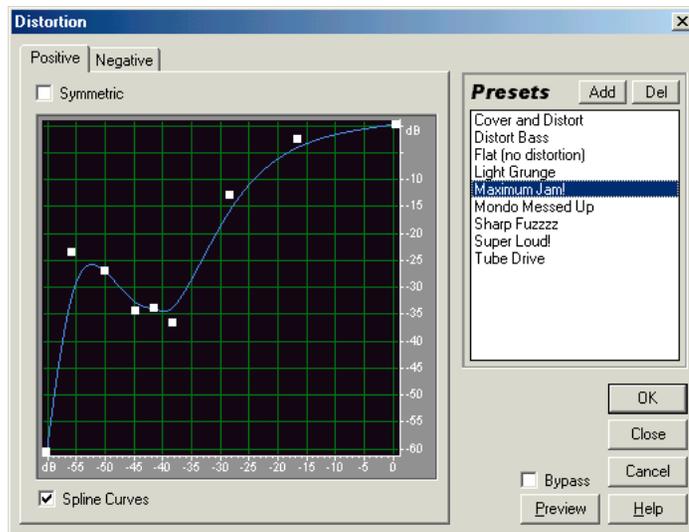
View Settings These settings affect the impulse display.

- Left: Select this button to enable display of the impulse for the left channel only.
- Right: Click this button to enable display of the impulse for the right channel only.
- View: Both Selecting this button enables display of the impulse for both the right and left channels.
- Normalized View: If this option is checked, the impulse’s amplitude is displayed so that it exactly fits in the display vertically.

Preview Press the Preview button to audition the current Convolution setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Distortion

Use this function to create effects such as a blown car speaker, a muffled microphone, and an overdriven amp. Have fun making your audio sound really, really bad! (Of course, it’s great for adding fuzz to guitar licks to get that “authentic” heavy metal sound.)



Distortion dialog

Positive/Negative Tabs You can specify separate distortion curves for positive and negative sample values. Select the tab corresponding to the distortion curve you want to display. If the Symmetric box is checked, these tabs become inactive (and are marked Symmetric).

Copy from Positive Press the Copy from Positive button (available in the Negative tab) to copy the positive curve to the Negative window.

Symmetric If Symmetric is checked, the positive and negative curves are identical. To specify separate Positive and Negative curves, uncheck Symmetric and choose the appropriate tab.

Distortion Graph This graph depicts the input sample value in dB along the x-axis (left and right) and the output sample value in dB along the y-axis (up and down). A line that flows directly from the lower-left to the upper-right (which is the default) depicts a signal that's been left untouched, since every input value goes to the exact matching output value. Adjusting the shape of this line will adjust the input to output assignments. The readout below the graph displays the current input and output sample values that are under your mouse pointer.

Note: For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

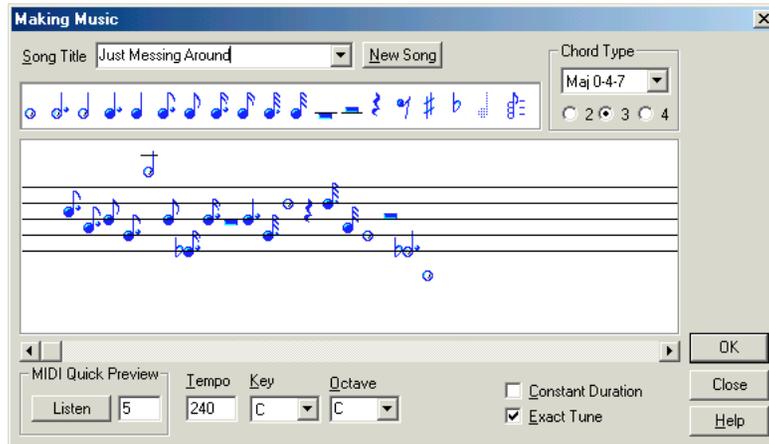
Spline Curves Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use Spline Curves, the line won't ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Presets Adobe Audition comes with several Distortion presets. More information on presets can be found in the "Looking at the Work Area" chapter.

Preview Press the Preview button to audition the current Distortion setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Music

Adobe Audition's Music function can use any highlighted selection as a "voice" to synthesize music or harmonize a wave using a particular chord. While this function is far from a complete MIDI authoring studio, it provides a quick and simple way to put a sample to music.



Music dialog

Before you bring up the Music dialog, you must first highlight the range of the current waveform you wish to use as a quarter note. If no range is selected, Adobe Audition will use the data on the clipboard. Keep in mind that the clipboard data will be filled with your sample automatically once music is generated. Thus, selecting music a second time will automatically use your last sample.

Song Title If you wish to keep a sequence of notes you've created, give it a name in the Song Title box. You can choose previously saved songs from the drop-down list of song titles you've created. The actual song data is saved in the file SONGS.INI in your Adobe Audition data folder.

New Song Press this button to clear the current song/note sequence.

Notes Simply drag the notes and rests you desire to the staff. To sharpen or flatten a note, drag the sharp or flat symbol on top of the note you wish to transpose. If you want to clear a sharp, flat, or chord from a note, use the faded looking quarter-note object, and drop it on the note you wish to bring back to normal.

You can move notes up or down after they have been placed, or pick them up to insert in a new position. To remove a note, pick it up and drop it off away from the bar. Use the horizontal scroll bar to work on individual portions of the song at a time. You can scroll to write a piece as long as 256 notes.

Chord Type Use the chord selection drop down list to determine the minor or major chord type and voicing you would like to use when entering a chord. You can choose to make a chord out of 2, 3, or 4 voices, and then choose the chord type from the list. Then, pick up the chord object (the 3 notes on top of each other) from the Notes area and drop it on a note above. The note you drop it on will be the starting note of the chord, and the other notes will automatically appear above it in the right ratios.

Staff Drag notes from the Notes area, and place them here. The note location (A, E-flat) will determine the playback pitch for your sample.

Listen If you have MIDI playback capabilities, you can listen to a preview of your sequence before actually applying it. Play begins at the leftmost note visible on the staff (which means play begins at the position you are scrolled to), and continues on to the end of the song. The music is played through channels 1 and 13 for Extended and Base level compatibility. Choose the desired instrument by typing its MIDI instrument number in the box to the right.



You can record music played by the Listen preview button. Simply start Adobe Audition's recording process, then go into the Music dialog and press Listen. When the song is done, hit Cancel, and then press Adobe Audition's Stop control to terminate recording. Important: Make sure your sound card's mixer is set correctly for recording the MIDI channel.

Tempo The tempo is given in quarter notes (beats) per minute. Your sample's length is the length of a quarter note. If your note is longer than the period determined by the tempo, then the notes will overlap.

Key Choose a key for your song from the Key drop-down list. Only standard major key signatures are listed, so for a minor key simply choose the relative major (B flat for C# minor, for example). The key of C is the default.

Octave This drop-down list lets you transpose the sequence by octaves. Choosing "C" plays the notes at normal transposition.

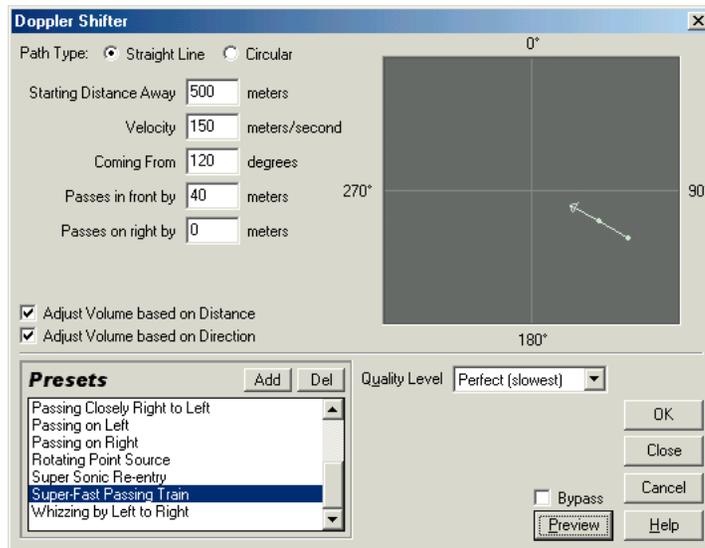
Constant Duration When you check the Constant Duration box, all notes will be the same length as the original sample, regardless of pitch. The operation that does this takes longer to calculate, but high-pitched notes will be the same length as lower pitched notes. If the Constant Duration box isn't checked, the note is created by directly stretching or compressing the original sample, resulting in higher pitches being shorter than lower pitches.

Exact Tune Check the Exact Tune box to tune your sample so that when played at A (above middle C), the frequency of your sample is at 440Hz. If the Exact Tune box isn't checked, your sample's original frequency will be played at A (above middle C).

Time/Pitch The effects in this menu to are used to alter the time (duration) and pitch (tonality) of an audio file or selected range.

Doppler Shifter

Use Adobe Audition's Doppler Shifter effect to create the "fly by" sounds that are most familiar from cars racing by or airplanes flying overhead. Let's use a car for our example.



Doppler Shifter dialog

The frequency of the noise from the car starts out at a high pitch and tempo, and lowers as it races past you. When the car is coming towards you, the noise it's making sounds like it's a higher frequency because each wave crest as it comes towards you is actually compressed by the car itself moving forward. The first crest leaves the car, and by the time the next one leaves, the car itself has moved forward, reducing the wavelength of the sound and raising its frequency. The opposite happens as it passes by you, and the waves are stretched out, resulting in a lower-pitched sound.

Path Type

Select one of the following:

- Select the Straight Line option to have effect's path visualized as a straight line on the graph in the right section of the Doppler Shifter window.
- Select the Circular option to visualize the effect's path as a circle on the graph in the right section of the Doppler Shifter window.

Depending on the path type you chose, you'll get a different set of options.

Note: Unlike many of the graphs in Adobe Audition's effects, the Doppler Shifter graph is non-interactive; you can't directly manipulate the items on the graph using your mouse. Instead, the graph simply displays what the Doppler effect will "look like" as you make adjustments to the effect's parameters.

Straight Line Options Set the following:

- Starting Distance Away: This value, measured in meters, is the virtual starting point of the effect.

- **Velocity:** The value entered here is the virtual speed at which the effect moves, measured in meters per second.
- **Coming From:** Enter the virtual direction (measured in degrees) that the effect is coming from.
- **Passes in front by:** Specify the number of meters you want to the effect to “pass in front” of the listener by.
- **Passes on right by:** Specify the number of meters you want to the effect to pass by the right of the listener’s perspective by.

Circular Options Set the following:

- **Radius:** This value, which is measured in meters, sets the circular dimensions of the effect.
- **Velocity:** The value entered here is the virtual speed at which the effect moves, measured in meters per second.
- **Starting Angle:** This value, measured in degrees, specifies the beginning virtual angle of the effect.
- **Center in front by:** Enter the number of meters where you want to the effect to center in front of the listener.
- **Center on right by:** Specify the number of meters where you want the effect to center on the right of the listener’s perspective.

Adjust Volume based on Distance With this box checked, the volume of the effect is automatically adjusted based upon the virtual distance values that have been specified.

Adjust Volume based on Direction Check this box to have the volume of the effect automatically adjusted based up the virtual direction values you’ve entered.

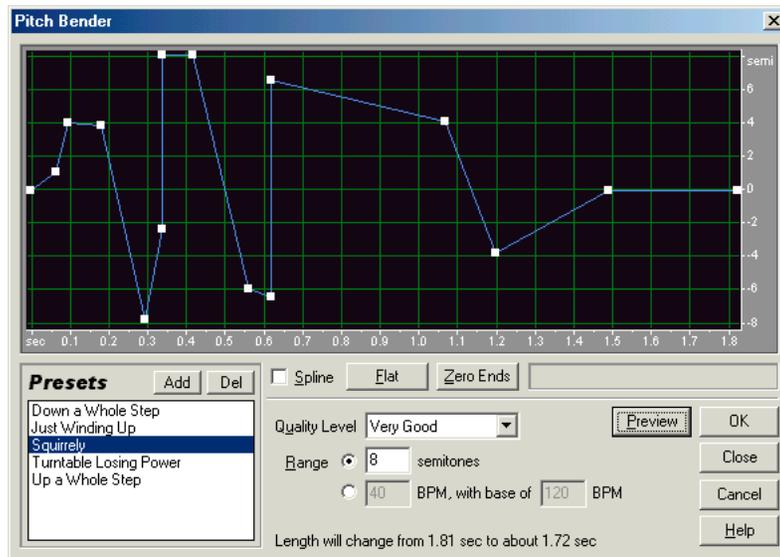
Quality Level This drop-down list lets you select from six different levels of processing quality. The lower quality levels will take less time to process, but the higher quality levels should produce more pleasing-sounding results.

Presets Adobe Audition comes with several Doppler Shifter presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Doppler Shifter setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition’s Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Pitch Bender

This effect varies the pitch of the source audio over time. Use the graph to “draw” your tempo to create smooth tempo changes or effects, such as that of a record or a tape speeding up or slowing down.



Pitch Bender dialog

Pitch Graph Use the graph to draw your tempo change over time. The x-axis (lower horizontal border) of the graph represents the sound file or highlighted section's time duration, while the y-axis (right vertical border) represents pitch, measured in semitones or beats-per-minute, depending on the Range button pushed. Higher points on the curve speed up the sound, while lower points slow it down.

Note: For more information about Adobe Audition's graph controls (such as how to add and remove control points), see the "Looking at the Work Area" chapter.

Spline Check the Spline box to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use Spline Curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve. To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points. Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat Press this button to reset the graph, with all points being set to no pitch bending.

Zero Ends If you're pitch bending the middle of a selection, it's good to have the endpoints be at the same rate as the surrounding audio. Pressing this button will set the endpoints to no pitch shift.

Quality Level Use this drop-down list to select the desired quality level. The higher quality levels will sound best, but will take longer to process. The lower quality levels have more unwanted harmonic distortion, but will process more quickly. Usually you won't notice any ill effects from harmonic distortion of the signal when using levels from Very Good on up. Aliasing will still occur however when pitch shifting up, but the higher quality levels will reduce the distortion effects tremendously when shifting the pitch down.

Range This sets the vertical scale of the graph (the y-axis). The range can be set as semitones (there are 12 semitones to an octave) or as beats-per-minute. Each format will give a different rate of pitch change for a perfect diagonal line on the graph. In semitones format, the pitch changes logarithmically up and down, while in beats-per-minute format, the pitch changes up and down linearly. The number of semitones to shift up or down can be entered. If doing beats-per-minute, a range as well as the base tempo needs to be entered. You may specify the exact tempo of your piece to change to different rates, but this isn't required.

Length Report At the bottom of the Pitch Bender dialog is a report of what the new length of the file will be. Keep in mind that when going down in pitch, it's easy to make very long files depending on how large a Range is selected.

Presets Adobe Audition comes with several Pitch Bender presets. More information on presets can be found in the "Looking at the Work Area" chapter.

Preview Press the Preview button to audition the current Pitch Bender setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Stretch

The Stretch effect changes the pitch (frequency) and/or tempo (duration) of the audio signal. It can also change either pitch or tempo without affecting the other. For example, you can use Stretch to transpose a song to a higher key, or to slow a passage down without changing the pitch. Pitch and tempo may also be varied over the length of the audio, giving the effect of slowing down and speeding up, or raising and lowering pitch.

<<Screenshot: TP-Stretch.bmp "File not found.">>
Stretch dialog

Constant Stretch/Gliding Stretch Tabs To perform a constant stretch (no variation in the amount of stretch) on the entire selection, select the Constant Stretch tab. To vary the stretch from one level to another, choose the Gliding Stretch tab. With Gliding Stretch, you can choose the initial stretching percent (less than 100% will compress the wave), and the final stretching percent. Specifying separate values for start and end stretches the waveform in a linear fashion from one ratio to another.

Initial/Final Sliders Adjust the desired slider to set the amount of stretch to be applied to the waveform. If you're using Gliding Stretch, sliders for both initial and final settings will be available. Changes in the sliders are reflected in each of the Ratio and Length boxes.

Ratio/Length Boxes You can enter a desired Ratio (in percentage) or final Length (in time) for the stretch. Entering one automatically changes the other. If the initial and final lengths are different, then the actual final length will be exactly $(\text{initial} + \text{final}) / 2$ when in Preserve Pitch mode.

Transpose Use this drop-down list to select musical transposition amounts. The corresponding numerical values are entered into the stretch sliders automatically. For example, to transpose your sound up one semi-tone (one half-step on a keyboard) choose 1# for one sharp.

Precision Three options for precision (overall faithfulness to the sound's quality) are provided: Low Precision, Medium Precision, and High Precision. When preserving pitch or tempo, the degree of precision that an audio file should be processed with is often balanced against the time that it takes to process that file. As you might expect, the tradeoff to having a precision factor that's set too high is that it uses up a great deal of processing power (and thus time). 8-bit or low-quality audio files can be processed in a short amount of time using the Low Precision setting, whereas a professionally-recorded audio file may require stretching using the High Precision algorithm. This will take longer, but the results can be worth it.



A quick way to determine the quality factor to use is to process a small portion at the lowest setting and then move up the scale until you've achieved the best balance of quality vs. processing time.

Stretching Mode Three Stretching Mode options are provided:

- **Time Stretch (preserves pitch):** Select this option to make the tempo or play speed shorter or longer, but keep the pitch the same throughout. Lower percentages slow down the tempo when Time Stretch (preserves pitch) is selected. Higher percentages increase the tempo when you choose Time Stretch (preserves pitch).

Note: This is the setting a radio or TV production person would use in the process of making a 33 or 28 second commercial exactly 30 seconds.

- **Pitch Shift (preserves tempo):** This setting raises or lowers pitch, while the tempo or speed of play remains the same. Higher percentages lower the pitch when Pitch Shift (preserves tempo) is selected. Lower percentages increase the pitch when you chose Pitch Shift (preserves tempo).

Note: This is the setting you'd use for making a voice sound deeper or higher, while not affecting the original playback speed.



Try using differing initial and final percentages to raise and lower the pitch without affecting the tempo.

- **Resample:** When you use this setting, both the pitch and tempo settings are affected. If using percentages above 100, the tempo will slow, while at the same time the pitch will lower. For percentages of 99 or lower, the tempo will speed up and the pitch will increase.

Pitch and Time settings Set the following options:

- **Splicing Frequency:** This setting determines the size of the chunk of audio data used when preserving pitch or tempo while elongating or truncating a waveform. The higher the value, the more precise the placement of stretched audio is over time. However, artifacts are more noticeable as rates go up. At higher precision, lower splicing frequencies may add stutter or echo. If the frequency is too high, it may sound "tinny" or like talking through a tunnel.

 *If the low precision mode is used, you can improve the quality of stretched mono-tonal (pure tone) samples by choosing a Splicing Frequency that's evenly divisible into the frequency of the sample. Use the Frequency Analysis window to find the sample's base frequency, then divide by an integer to get the Splicing Frequency. For example, if the tone was reported to be 438Hz, dividing by 20 gives 21.9Hz. Thus, using 21.9Hz as the Splicing Frequency will greatly improve the quality by reducing phase artifacts. For non-tonal or noisy samples, the Splicing Frequency doesn't matter as much.*

- **Overlapping:** When stretching or compressing, “chunks” of audio are overlapped with previously transformed chunks. This setting determines how much the current chunk will overlap with the previous and next ones. If you find you're getting a chorus effect when stretching, you can lower the overlapping percentage. However, doing so may produce a “choppy” sound. If so, adjust the overlapping value to strike a balance between choppiness and chorusing. Overlapping can be as high as about 400%, but you should only go this high for really high speed increases (like 200% or more).
- **Choose Appropriate Defaults:** When preserving pitch or tempo, check this box to have Adobe Audition automatically select good default values for the Splicing Frequency and Overlapping settings.

Presets Adobe Audition comes with several Stretch presets. More information on presets can be found in the “Looking at the Work Area” chapter.

Preview Press the Preview button to audition the current Stretch setting before you apply it to the waveform. The selection is looped until you press the Stop button. Since Adobe Audition's Previews are dynamic, this gives you the opportunity to make and hear processing adjustments as the selection is continuously played.

Refresh Effects List

Select this item to scan your system for new Adobe Audition effects and DirectX plug-ins.

Chapter 9: Generate Menu – Edit View

The Generate menu displays all the options that relate to waveform generation when you're in the Edit View mode.

Silence

This function generates a specified amount of silence (entered in seconds) from the current playbar cursor position. Any following audio is pushed back in time, thereby lengthening the waveform's duration. Silence will replace any selected area.

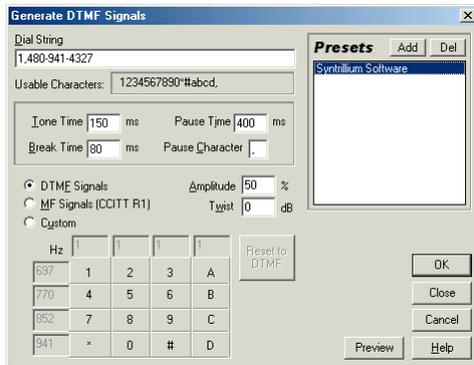
To use, just enter the desired amount of silence you wish to generate (measured in seconds) and press the OK button. Feel free to use decimals for partial seconds. For instance, entering .3 in the Silence Time field will generate exactly three-tenths of a second of silence.



To “silence out” a highlighted selection, use *Effects > Silence* instead.

DTMF Signals

This function generates Dual Tone Multi-Frequency (DTMF) signals (a.k.a. “touch tones”) used for dialing telephone numbers over phone lines that are capable of responding to touch tone signals. These signals are recommended internationally by the International Telegraph and Telephone Consultative Committee as the signals for pushbutton telephones.



Generate DTMF Signals dialog

Keep in mind that the DTMF signals generated by telephone pushbutton keypads are different from the Multi-Frequency (MF) tones generated by the telephone network to transmit information. Adobe Audition's DTMF Signals function can generate MF tones as well.

Dial String This is where you enter the phone number for which you want to generate tones. You may enter other characters such as the “*” and “#” symbols, as well as extra digits “a”, “b”, “c”, and “d”. Entering the pause character (defined in Pause Character below) inserts a pause of a defined length.

Tone Time Enter the milliseconds for which the tones will last. The standard time for DTMF tones is 100ms.

Break Time This is the number of milliseconds of silence between successive tones.

Pause Time When the pause character is used in the Dial String field, this is where you enter the length of the pause, measured in milliseconds.

Pause Character When you enter this character in the Dial String field, Adobe Audition interprets it as a pause, and will insert silence for the duration specified in the Pause Time field.

DTMF Signals With this option selected, Adobe Audition generates DTMF signals (normal push-button telephone type tones) using combinations of the frequencies 697Hz, 770Hz, 852Hz, 941Hz and 1209Hz, 1336Hz, 1477Hz, and 1633Hz.

MF Signals (CCITT R1) Choose this option to have Adobe Audition generate MF signals (tones that are internal to telephone networks) using paired combinations of the frequencies 700Hz, 900Hz, 1100Hz, 1300Hz, 1500Hz, and 1700Hz.

Custom This feature allows you to specify the combinations of frequencies to be used in generating signals. Select this option to enable the frequency entry boxes on the keypad chart below. Click in an entry box to change that frequency.

Amplitude This setting determines the volume level (in percent) of the tones generated, where 100% means maximum volume without clipping.

Twist “Twist” refers to how much louder the higher frequency tone is from the lower frequency tone. Enter a value (measured in dB) in the Twist field to increase the volume of the higher frequency tone accordingly.

Reset to DTMF Press this button to clear any custom frequency entries and replace them with the standard DTMF frequency combinations.

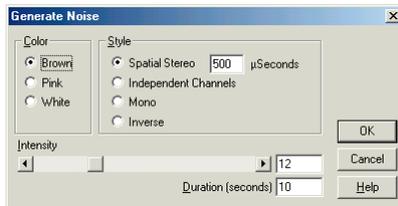
Presets This section allows you to recall and manage DTMF Signals presets. More information about Presets can be found in the “Looking at the Work Area” chapter.

 *The presets in this function save everything, including the dial string. To see how effective these tones are, try entering your favorite phone number to generate the tones for it. Then hold the receiver of your phone next to the speaker and play the wave. Your phone will dial the number you entered.*

Preview Press this button to audition the current DTMF Signals settings before you apply them to the active waveform.

Noise

This function generates random noise in a variety of colors. Traditionally, color is used to describe the spectral composition of noise. Each color has its own characteristics.



Generate Noise dialog

Adobe Audition inserts noise from the current playbar cursor position, pushing any following audio back in time, and thereby lengthening the duration of the waveform. Selecting an area and generating noise replaces the selection, deleting existing audio.

 Use noise as the basis for creating weird sound effects, for creating soothing sounds like waterfalls (perfect for use with Adobe Audition's Binaural Auto-Panner function), or for generating signals that can be used to check out the frequency response of a speaker, microphone, or other audio system component.

Color Adobe Audition has the following noise options:

- Brown noise has a spectral frequency of $1/f^2$. This means, in layman's terms, that there's much more low-end, and there are many more low-frequency components to the noise. This results in thunder- and waterfall-like sounds. Brown noise is so called because, when viewed, the wave follows a Brownian motion curve. That is, the next sample in the waveform is equal to the previous sample, plus a small random amount. When graphed, this waveform looks like a mountain range.
- Pink noise has a spectral frequency of $1/f$ and is found mostly in nature. It is the most natural sounding of the noises. By equalizing the sounds, you can generate rainfall, waterfalls, wind, rushing river, and other natural sounds. Pink noise is exactly between brown and white noise (which is why some people used to call it tan noise, but pink was more appealing). It is neither random nor predictable. It has a fractal-like nature when viewed. When zoomed in, the pattern looks identical to when zoomed out, except at a lower amplitude.
- White noise has a spectral frequency of 1. In other words, equal proportions of all frequencies are present. Because the human ear is more susceptible to high frequencies, white noise sounds very "hissy". Adobe Audition generates white noise by choosing random values for each sample.

Style Choose among the following styles:

- Spatial Stereo: If you choose this option, Adobe Audition generates Spatial Stereo noise by using three unique noise sources, and spatially encodes them to appear as if one is coming from the left, the other from the center, and the last from the right. When you listen to it with stereo headphones, your mind perceives sound coming from all around, not just in the center. To specify the distance from center of the left and right noise sources, enter a delay value in microseconds. About 900 to 1000 microseconds correspond to the maximum delay perceivable. A delay of zero is identical to monaural noise, where left and right channels are the same.
- Independent Channels: Select this option to tell Adobe Audition to generate noise by using two unique noise sources, one for each channel. The left channel's noise is completely independent of the noise of the right channel.

- **Mono:** If this option is chosen, Adobe Audition generates Mono noise by using a single noise source, with the left and right channels set equally to the same noise source.
- **Inverse:** With this option selected, Adobe Audition generates Inverse noise by using a single noise source (similar to the Mono option). However, this time the left channel's noise is exactly inverse of the right channel's noise. When you listen to it with stereo headphones, the effect is that of the sound coming from the center of your head instead of out in space somewhere.

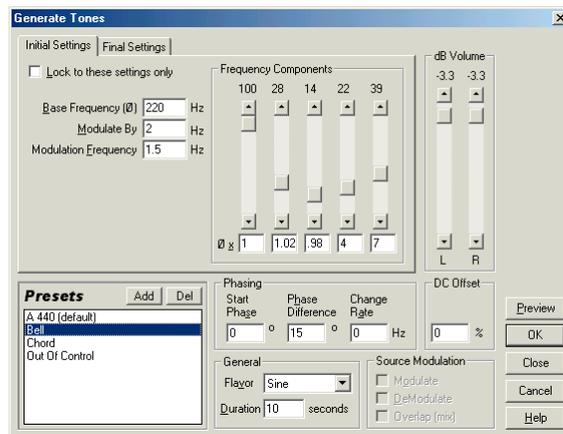
Intensity Adjust this slider to specify the noise's intensity, using a scale from 2 to 40. Or, you can key in the value numerically in the entry box to the right of the slider. With higher intensities, the noise becomes more erratic, and sounds harsher and louder.

Duration In this field enter the number of seconds of noise you want to generate.

Note: For very long periods of noise, it's faster to generate a shorter period (say, about 10 to 20 seconds) and delete excess noise at the beginning and end so that the waves are starting and ending at the midpoint. Then copy and loop (using Edit > Mix Paste) as many times as needed.

Generate Tones

Use this function to create a simple waveform, with control over numerous amplitude- and frequency-related settings. Generating tones is a great way to provide base audio to create impressive sound effects.



Generate Tones dialog

Initial Settings When the Lock to these settings only box is unchecked, choose this tab to select the initial tone settings. The tone generated will gradually go from the initial state to the final.

Final Settings When the Lock to these settings only box is unchecked choose this tab to select the final tone settings. The tone generated will gradually go from the initial state to the final.

Lock to these settings only Check or uncheck this box as desired:

- If checked, the overtones, base frequency, modulation, modulation frequency, frequency multipliers, and overtone intensities are constant; they do not vary over time.
- Uncheck this box to dynamically change the proportion of any overtone over time by choosing the initial and final proportions. You may also dynamically change the frequency multipliers, base frequency, modulation, and modulation frequency for interesting effects. Click on the appropriate tab to change Initial and Final settings.

Copy from Initial Settings This button in the Final Settings dialog copies the tone values from the Initial Settings window to the Final Settings window for editing.

Base Frequency This is where you enter the main frequency to be used for sound generation.

Modulate By This setting modulates the Base Frequency in pitch over a user-defined range. For example, a 100Hz setting modulates the original frequency by +/- 100Hz. In other words, a 1000Hz tone would modulate between 900Hz and 1100Hz.

Modulation Frequency This is the rate (times per second) at which the frequency modulates. Entering a value of 10, for instance, generates tones that seem to warble in amplitude at the rate of 10 times per second. The tones actually warble in pitch (frequency) as they should, but because of the variance in the perceived energy levels of different frequencies to the human ear, they seem to warble in amplitude.

Frequency Components Use the Frequency Components controls to add up to five overtones to the fundamental frequency (Base Frequency).

Enter the multiplier for each overtone below the Frequency Components slider(s). (The actual frequency will be this many times the fundamental.) Then, use the sliders to mix each of the individual components (0 to 100%) in proportion to one another. The overall gain (signal level) may be adjusted with the dB Volume sliders.

If the Lock to these settings only box isn't checked, all of the values can change over the duration of the audio file, so that they morph from the initial to final settings.



You can generate several effects with these five overtones. Experiment and have fun.

dB Volume Use the dB Volume sliders to select the overall gain for each of the right and left channels from -80dB to 0dB. You can control both channels independently when generating stereo tones.

Phasing Choose the appropriate phase:

- Start Phase is the starting location in the cycle that will be produced. If you start at 0 degrees phase, waves will start at the baseline. If you start at 90 degrees, the wave will start at full amplitude (generating a noticeable click as well). If you are working in great detail with tones and need to have the phase “just so”, this option allows you to control that.
- Phase Difference purposefully allows the left channel to be out of phase with the right channel. A value of 0 will be completely in-phase, and 180 will be completely out of phase.
- Change Rate dynamically changes the relative phase between the two channels of a stereo audio file over time at a given rate. For example, if you enter 1Hz, the phase difference will cycle through 360 degrees each second.

DC Offset The DC Offset value entered here adds a constant DC (“Direct Current”) amplitude to the tone, which shifts the entire waveform up or down by the specified percentage.

Flavor Use this drop-down list to choose the type of waveform to use. Each flavor has a particular sound unique unto itself.

- SineFundamental, no harmonics (pure tone)
- TriangleOdd harmonics with amplitude of 1 to itself (squared)
- Square Odd harmonics with amplitude of 1 to itself
- SawtoothAll harmonics with amplitude of 1 to itself

Duration This setting lets you enter the length of the generated tone (in seconds). Feel free to use decimals for partial seconds. For instance, entering .25 in the Duration field will generate tones for exactly one-fourth of a second.

Source Modulation Choose an option for modulation:

- Modulate: With this option checked, instead of generating new tones, the currently highlighted wave data is be “ring modulated”, or multiplied, by the current tone settings. This is great for adding really weird special effects.

Note: If you check the Modulate option, make sure that a section of the active waveform is highlighted.

- Demodulate: Like Modulate, Demodulate requires a highlighted selection. Demodulate can be used on a previously modulated source with interesting effects.
- Overlap (mix): Checking this option will mix the generated tones on top of the selected audio.

Presets This section allows you to recall and manage Generate Tones presets. More information about Presets can be found in the “Looking at the Work Area” chapter.

Preview Press this button to audition the current Generate Tones settings before you apply them to the active waveform.

Chapter 10: Analyze Menu – Edit View

The Analyze menu contains functions that give you information about the file or selection being displayed when in the Edit View mode.

Show Frequency Analysis

This command displays Adobe Audition's Frequency Analysis window. It contains a graph of the frequencies at the cursor position or at the center of a selection. The x-axis (left to right) represents frequency (measured in Hz), while the y-axis (bottom to top) corresponds to the amplitude of the corresponding frequency (measured in dB) on the x-axis.



Frequency Analysis Window

The Frequency Analysis window can float above Adobe Audition's main interface, or dock at any one of several places within it.

Linear View Check the Linear View box to display the graph's plot with a linear horizontal frequency scale; uncheck it to display the plot on a logarithmic scale.

Hold Press one of the four Hold buttons (which are numbered 1, 2, 3, and 4) as a waveform is playing to take a "snapshot" of the waveform's frequency at that moment in time. The frequency outline (which is rendered in the same color as the button pushed) is frozen on the graph and overlaid on the other frequency line(s). Up to four frozen frequency outlines may be shown at once.

To clear a frozen frequency outline, press its corresponding Hold button again.

Status Areas The left status area directly underneath the graph displays the waveform's overall highest frequency, as well as the maximum amplitude for both the left and right channels.

The right status area directly underneath the graph displays the frequency for the left and right channels measured at the insertion point. It also shows, in parentheses, each channel's musical equivalent note number and positive/negative cents.

Options Use this drop-down list to select a graphing display option.

- **Lines:** The waveform's left and right channels appear as simple lines, with each channel displayed in a different color.
- **Area (Left on top):** The Area modes are similar to the Lines option, but the area beneath the lines is rendered as a solid color. The lines that represent the frequencies are smoothed out, and the left channel appears in front of the right channel.
- **Area (Right on top):** Choose this option to have the right channel appear in front of the left channel in Area mode.
- **Bars (Left on top):** While the Area modes smooth out the lines that represent the frequencies, the Bars modes show the actual limitations on analysis resolution by splitting the display into rectangular segments. The higher the FFT size, the greater the analysis resolution, and the narrower the bar. Select this option to have the left channel appear in front of the right channel.
- **Bars (Right on top):** Choose this option to have the right channel appear in front of the left channel in Bars mode.

Scan Click this button to scan the highlighted selection and show all frequencies present in that selection.

Advanced Press the Advanced button to see additional options.

FFT Size This drop-down list allows you to select from several FFT (Fast Fourier Transform) sizes. Higher FFT sizes give you more accurate results in terms of frequency (such as the overall frequency estimate), but are also much slower.

***Note:** When you set the FFT size to 4096 or lower, the Frequency Analysis window updates in real time while you play your file. (Keep in mind that how well it ultimately updates in real time with these settings is based on your computer's speed.)*

You can also generate a step-by-step animation by clicking on the main waveform window and then holding down the Right Arrow key. As the cursor scrolls across the display, CEP displays the spectral information in the Analysis window.

FFT Window Type This drop-down list lets you choose from six types of FFT windows. Each window type displays a slightly different kind of frequency graph.

The Triangular window gives a more precise frequency estimate, but it's also the noisiest. This means that other frequencies will be shown as present, even though they may be much lower in volume.

At the other extreme, the Blackmann-Harris window has a broader frequency band which isn't as precise, but the sidelobes are very low, making it easier to pick out the major frequency components.

Reference The value entered into this field means that a full scale (FS) sine wave will show up as that amplitude. For instance, with the default value of zero, a full scale sine wave will display with its peak at 0dB. A value of 30dBFS would mean that a full scale sine wave displays with its peak at 30dB. In other words, this value essentially just scoots the display up or down.

Copy to Clipboard Press this button to send a text-based frequency report of the current waveform to the Windows Clipboard, ready to paste into your favorite Windows-based text editor or word processor.

Zooming (Frequency Ruler) To zoom in on the Frequency Analysis window's x-axis (horizontal) ruler – the one that measures frequency – place your mouse pointer over the ruler. Then, right-click and drag to select the area you wish to zoom in on. (You'll notice that your pointer turns into a magnifying glass.) Release the right-click button and the zoom-in occurs.

To navigate a zoomed-in frequency display, put your mouse pointer over the horizontal ruler – the pointer then changes into a hand. Left-click and drag the hand horizontally to scroll the current zoom range forward and backwards.

To zoom out, place your mouse cursor over the horizontal ruler and right-click. A menu pops up with two options.

- Select Zoom Out to get back to the previous view displayed before performing the last zoom-in.
- Choose Zoom Out Full to display the entire horizontal waveform or selection in the Frequency Analysis window.

Zooming (Amplitude Ruler) To zoom in on the Frequency Analysis window's y-axis (vertical) ruler – the one that measures the amplitude of a frequency – place your mouse pointer over the ruler. Then, right-click and drag to select the area you wish to zoom in on. (You'll notice that your pointer turns into a magnifying glass.) Release the right-click button and the zoom-in occurs.

To navigate a zoomed-in frequency amplitude display, put your mouse pointer over the vertical ruler; the pointer then changes into a hand. Left-click and drag the hand horizontally to scroll the current zoom range up and down.

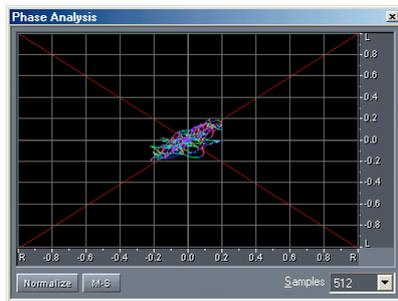
To zoom out, place your mouse cursor over the vertical ruler and right-click. A menu pops up with two options.

- Select Zoom Out to get back to the previous view displayed before performing the last zoom-in.
- Choose Zoom Out Full to display the entire vertical ruler in the Frequency Analysis window.

Show Phase Analysis

This command displays Adobe Audition's Phase Analysis window. Using a Lissajous Plot graph, it lets you easily see any phase differences between the channels of the current waveform. ("Phase" as used here is the amplitude difference between channels.)

- A mono (in phase) waveform will display as a rising diagonal line from left to right.
- An inverse left channel 180-degrees out-of-phase waveform (the right channel is in phase) will display a line descending from left to right.
- A right channel-only waveform will display as a horizontal line.
- A left channel-only waveform will display as a vertical line.
- Most stereo audio is generally fuzzy looking, upward sloping from left to right.
- If audio has very wide stereo separation, the display will be going in almost all directions.



Phase Analysis Window

The Phase Analysis window can float above Adobe Audition’s main interface, or dock at any one of several places within it.

Note: *The Phase Analysis window is designed to be used with stereo waveforms only. If used with a mono audio file, all you’ll see on the graph is a single line.*

Should only one channel of stereo waveform be highlighted, the Phase Analysis window will show a moving straight line as the audio file is played: a horizontal line if it’s the left channel, a vertical line if it’s the right channel.

Normalize Press this button to make the phase analysis visualization as large as possible so the largest of its edges reaches the graphing window’s boundary.

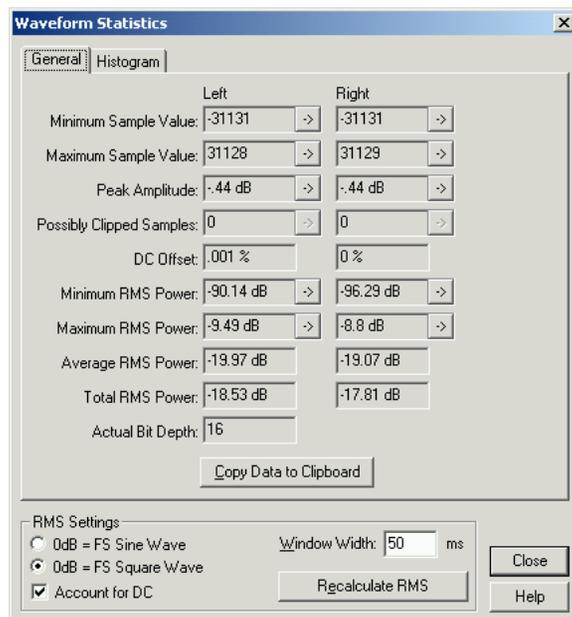
M-S Button When the M-S (Mid-Side) button is pushed, the display is essentially rotated to the left by 45 degrees. The x-axis (horizontal) ruler plots the side channel $[(\text{right} - \text{left})/2]$ while the y-axis (vertical) ruler plots the mid channel $[(\text{right} + \text{left})/2]$.

Samples This drop-down list allows you to select from several sample sizes.

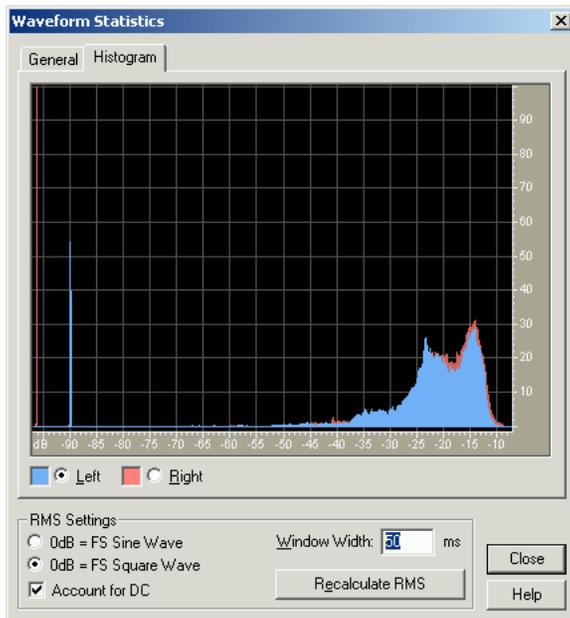
Higher sample sizes give you more accurate results, but require much more processing power to be effective. Choose the sample size that produces the best performance for your system.

Statistics

The Waveform Statistics dialog displays a wealth of information about the active waveform. It consists of two tabbed areas (General and Histogram) plus a section for RMS settings. Details about each channel of a stereo waveform are clearly shown.



Waveform Statistics dialog – General tab



Waveform Statistics dialog – Histogram tab

You can jump to the exact location in the waveform for certain values by clicking on the appropriate arrow button to the right of that value.

General Tab

Minimum Sample Value This field shows the minimum sample values in the waveform or highlighted range. Press the [->] button to place the playbar cursor at that location. The Waveform Statistics dialog is automatically closed.

Maximum Sample Value This field shows the maximum sample values in the waveform or highlighted range. Press the [->] button to place the playbar cursor at that location. The Waveform Statistics dialog is automatically closed.

Peak Amplitude The waveform's peak amplitude is shown here. Peak amplitude is the absolute maximum sample value given in decibel form. Press the [->] button to have the playbar cursor jump to the peak amplitude in the waveform. The Waveform Statistics dialog is automatically closed.

Possibly Clipped Samples This field displays the number of samples which may be clipping (those that reach -32768 or 32767 for 16-bit waveforms, for example). Press the [->] button to have the playbar cursor jump to the first clipped sample. The Waveform Statistics dialog is automatically closed.

DC Offset The measure of the DC ("Direct Current" or center of the waveform) is shown here, measured in percent. Positive values are above the center line (zero volts) and negative values are below.

Minimum RMS Power This value is the minimum Root Mean Squared amplitude (closer to what the ear hears) of the waveform that is scanned using a window of the size given below as the RMS Window Width. Press the [->] button to have the playback cursor jump to the average quietest section of the waveform. The Waveform Statistics dialog is automatically closed.

Maximum RMS Power This value is the maximum Root Mean Squared (closer to what the ear hears) amplitude of the waveform that is scanned using a window of the size given below as the RMS Window Width. Press the [->] button to have the playback cursor jump to the average loudest section of the waveform. The Waveform Statistics dialog is automatically closed.

Average RMS Power This figure represents the average power of the entire selection. This is a good measure of the overall loudness of the waveform selection.

Total RMS Power This figure represents the total power of the entire selection given as RMS amplitude.

Actual Bit Depth The waveform's bit depth (or "float" if it uses the full 32-bit float range) is reported here.

Copy Data to Clipboard Press this button to copy all information displayed in the General tab to the Windows clipboard. You can then paste this info into a text editor or any other Windows application that supports pasting of text from the clipboard.

Histogram Tab

This section displays a histogram of the RMS amplitude values in the highlighted audio selection. (A histogram is a bar chart that shows the distribution of frequencies, with heights of the bars representing observed frequencies.) This can be useful for determining future amplification amounts, dynamic compression amounts, and more.

The graph shows which amplitudes are most common. The x-axis (horizontal) ruler measures the amplitude in dB, while the y-axis (vertical) ruler measures percentage. The more audio there is at a particular amplitude, the more the graph will show for that amplitude.

Left/Right Buttons Use the buttons to display the corresponding channels.

- Select the Left option to see the waveform's left channel visualized as a histogram.
- Pressing the Right option displays the right channel of the waveform in histogram form.

Different colors are used to represent each of the two channels.

RMS Settings Control the display with these settings:

- **0dB = FS Sine Wave:** If this option is checked, then this means that 0 dB displayed in any of the RMS boxes is equivalent to a sine wave at full scale (with peak amplitude at 0dB – using every sample value in the 16-bit range). If you generate a max amplitude sine wave, it'll read 0dB if this button is chosen.

- **0dB = FS Square Wave:** If this option is selected, then a full scale square wave will read out as 0dB for the RMS. Some users expect a full scale square wave to be 0dB, while others expect a full scale sine wave to be 0dB. Since a full scale square wave is about 3.02dB louder than a full scale sine wave, these RMS boxes will adjust by about 3dB when you toggle between them.

Account for DC If this box is checked, when calculating the RMS values, the DC offset found is subtracted out before doing the RMS calculation. This means that if your signal has a DC offset (say, a quiet signal that is well above the center line) then the actual RMS signal amplitude is actually quite small, but could read much higher because of the DC amount. So, for the most accurate signal-relative RMS readings, you would want to account for DC. Otherwise, even a signal that is total silence, but has a high DC offset, will read as having a high RMS, which may not be what you want.

Window Width Adobe Audition measures the RMS over a window (measured in milliseconds) of this size when calculating the RMS minimum and maximum values.

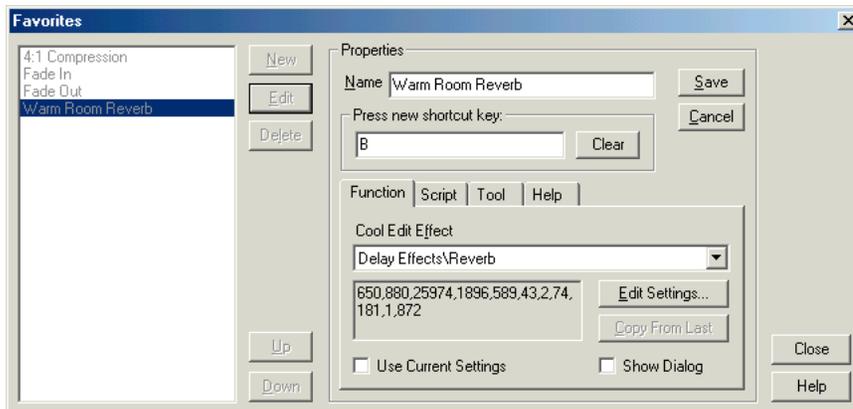
Recalculate RMS After adjusting the Window Width value, click this button to recalculate the RMS values based on the new window size.

Chapter 11: Favorites Menu – Edit View

The Favorites menu allows you to create, customize, and save your favorite Adobe Audition effects, scripts, and even third-party tools (the latter using command line executables). All of the favorites you create appear on this menu. You can also categorize the menu items into hierarchical submenus for easy organization.

Edit Favorites

This command brings up the dialog where you create, delete, edit, and organize items that appear in Adobe Audition's Favorites menu.



Edit Favorites Dialog

New Press this button to enable the fields in the Properties area of the Favorites dialog so that you can create a new Favorite.

Edit Pressing this button allows you to modify the properties of a highlighted entry in the Favorites list.

Delete This button deletes the highlighted Favorite.

Up This button moves the selected item up in the list. The order of this list is the same as in the Favorites menu.

Down This button moves the selected item down in the list. The order of this list is the same as in the Favorites menu.

Name Enter the name of a created Favorite in this field.

You may create hierarchical menus by using a backslash. For example My Delays\Hall Reverb places the item under Hall Reverb in the My Delays submenu.

To place a separator bar into the Favorites list to separate one function type from the others, enter a series of dashes (e.g., “-----”) into the Name box. If you want more than one separator, enter a different number of dashes, or change the text so that it doesn't match something already in the list. For instance, “-----2” is considered a valid separator; the 2 after the dashes will be ignored.

To create a separator bar for a submenu, enter the submenu path first (e.g., “My Effects\-----”). You can also enter any text you would like the Favorites menu to display. The text will be nonfunctional, of course.

Save Press this button to save a new or edited entry to the Favorites List.

Note: Make sure you do this before you press the Favorites dialog's Close button.

Cancel A press of this button cancels the current Favorites edit and disables the fields in the Properties area.

Press new shortcut key You may assign a key or combination of keys to act as a shortcut to any of your Favorites. Adobe Audition gladly accepts most single key shortcuts. (The most notable exceptions are the Print Scrn, Scroll Lock, Num Lock, Caps Lock, Tab, and Enter keys.) However, you can use the Ctrl, Shift, and Alt keys (or any combination of the three) to act as the first part of a multiple key shortcut.

With your text cursor in the Press new shortcut key field, press the desired key(s) to use for the Favorite's shortcut key.

Note: If the shortcut key you choose is already in use elsewhere in Adobe Audition, you'll be alerted with a dialog box and given the opportunity to overwrite the current shortcut key assignment.

Clear Press this button to remove the entry in the Press new shortcut key field.

Function Tab

Audition Effect All of the items on Adobe Audition's Effects and Generate menus appear on this drop-down list. Just select the desired item. The last used settings for the item chosen appears in a window just below the Effect list.

Edit Settings Click on this button to call up the settings window that corresponds to the particular Effects or Generate function. You can then change the settings to be used when this item is chosen from the Favorites menu.

Copy From Last Press this button to copy all the settings from the last Effects or Generate function that was started successfully.

Use Current Settings Check this option to apply to the Favorite the settings last used for that particular Effects or Generate function.

Show Dialog Check this box to bring up the settings dialog for the particular Effects or Generate function when you call the Favorite. The settings dialog will show the settings you chose in Edit Settings.... Uncheck the Show Dialog box to automatically use the chosen settings to run the function.

Script Tab

Script Collection File This displays the current script collection in use. The small button to the left of the field brings up a browse dialog that allows you to choose an Adobe Audition script collection (*.scp) file.

Script This drop-down list contains all scripts available in the selected collection. Choose the particular script you want to run.

Pause at Dialogs If checked, the script stops at each settings dialog to allow you to modify the values. Otherwise, the entire script will run to completion before control returns.

Tool Tab

Command Line Enter the command line for the tool you want to run here, including any command line switches the particular tool may need. The small button to the left of the field brings up a dialog that lets you browse your system for the desired program.

Help Tab

This area provides instructions for adding separators and submenus to the Favorites menu.

To add a Fade Out function to the Favorites List and give it the hotkey of Alt+F7:

- 1 Select Edit Favorites... from the Favorites menu.
- 2 On the Favorites dialog that appears, push the New button.
- 3 In the Name field, type Fade Out (without the quotes).
- 4 Check the Alt box and select F7 from the Hot Key drop-down list.
- 5 From the Audition Effect drop-down list in the Function tab, Select Amplitude\Amplify.
- 6 Press the Edit Settings...button, which will display the Amplify effect dialog.
- 7 Choose Fade Out from the Amplify dialog's list of Presets and click OK.
- 8 Make sure the Show Dialog box is unchecked.
- 9 Press the Save button to add Fade Out to the Favorites list and press Close.

The next time you want to fade out a portion of a waveform, just highlight it and press Alt+F7.

Chapter 12: Options Menu – Edit View

The Options menu gives you access to Adobe Audition's configuration options, system settings, batch-processing functions, and more.

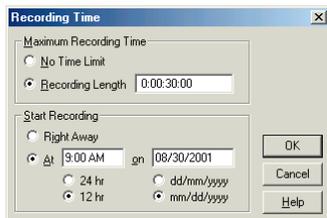
Loop Mode

The Loop Mode option toggles Adobe Audition's Play control between Loop and Play states.

- When Loop Mode is checked, pressing the Play control will continuously loop the current selection (or the entire file, if nothing is selected).
- When there is no checkmark, the Play control plays the selection (or file) once and then stops playback.

Timed Record

This option can be used to set a finite time limit allowed when recording, or to specify a time when Adobe Audition will auto-start and stop recording. With the Timed Record option checked, pressing the Record control on the transport toolbar will bring up a dialog box prompting you to enter the recording time.



Timed Record Dialog

Maximum Recording Time

- No Time Limit: Select No Time Limit and Adobe Audition will record until you tell it to stop (or when it runs out of free hard disk space).
- Recording Length: If the Recording Length option is selected, Adobe Audition will record for the period of time entered in the Recording Length text box. When that time period is up, the recording is automatically stopped.

Upon pressing OK, Adobe Audition will be placed into record mode, either until the Stop control is pressed, or the time length has expired.

Start Recording

- Right Away: Choose Right Away and Adobe Audition will begin recording as soon as the OK button is pushed.

- **Time/Date:** If you want the recording to begin at a future time (for instance, to have Adobe Audition capture a radio broadcast at a certain time), select the time/date option. Fill in the starting time and date in the appropriate fields, and choose the desired time and date options.

Upon pressing OK, the record button will blink, and recording will start when your computer's time reaches the time specified. Recording will then continue for the length given, until it's stopped manually, or when Adobe Audition runs out of available hard disk space.

Monitor Record Level

Checking this option will activate the Level Meters and start monitoring the recording source, which is useful for setting levels before recording. To quit monitoring, press the Stop control or uncheck Monitor Record Level on the Options menu.



Use the F10 key to quickly turn on and off the Level Meters.

Show Levels on Play and Record

This item enables or disables Adobe Audition's Level Meters, which are used to visualize audio levels during recording and playback. It doesn't, however, cause the Level Meters to go away. To do that, go to the View menu and uncheck Show Level Meters.

MIDI Trigger Enable

By checking MIDI Trigger Enable, any of Adobe Audition's shortcuts that have been assigned to MIDI events can be called from a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command. Disable this option if you don't wish to have shortcuts respond to MIDI events.

***Note:** Before attempting to enable MIDI triggering, you'll need to choose a device for MIDI In that's recognized by Windows, such as a soundcard's built-in MIDI interface or a MIDI interface card. To do this, go to Options > Device Properties... and click on the MIDI In tab. Use the drop-down list to select the input device to be used for triggering.*

Synchronize Cursor Across Windows

With this command check-marked, the playback cursor bar or currently highlighted selection will retain its position when switching between open waveforms within Adobe Audition.

For example, if you are looking at 1.wav and highlight from 0:01:00 to 0:02:00 and then switch to 2.wav using the Window menu or the Organizer window, 2.wav would also have 0:01:00 to 0:02:00 highlighted. This can be useful if you're switching between different versions of the same waveform during editing.

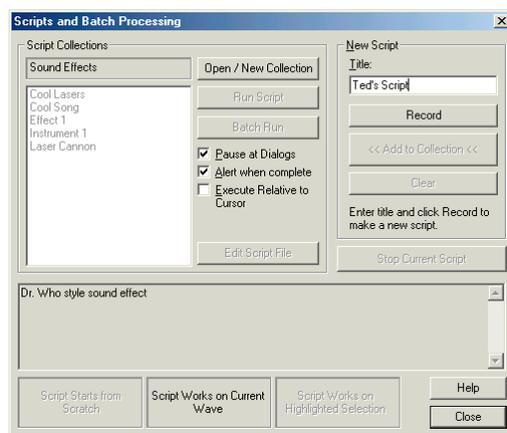
When unchecked, the positions of each of the waveform's previous cursor or range areas will be retained and redisplayed. If the waveform that you're switching to doesn't contain the currently selected time location (as in a .5 sec waveform, in our example), then switching acts this way as well.

Windows Recording Mixer

Select this command to display the Record Control part of the Windows Mixer (a.k.a. Volume Control) so that you adjust sound card recording levels.

Scripts & Batch Processing

Adobe Audition scripts allow you to save a series of actions (such as a cut/paste or applying an effect) to be performed again with the click of a button. Scripts – which are simple text files – are similar to macros; the exact movements of your mouse and any tweaking of parameters are stored, ready to be repeated in the same sequence when recalled.



Scripts and Batch Processing Dialog

As an example, suppose you have a combination of effects with particular settings (an EQ setting, a Hall reverb, etc.) that you like to apply often and in combination to achieve a certain sound. You can record these steps, along with specific settings, to be carried out again by simply calling the script.

There are three types of scripts. The type created depends upon Adobe Audition's state when you initiate the recording:

- Scripts that start from scratch. These scripts start with no waveform opened, and their first command is File > New.
- Scripts that work on the current audio file. This type of script operates on the entire waveform. It requires a file to be opened, but with no selection made. Actions begin at the playback cursor position in the waveform, and will affect any data present at that point.

- Scripts that work on a highlighted selection. This type requires a highlighted selection to be made. All actions in the script pertain only to the portion that is highlighted, leaving the rest of the waveform untouched.

Script Collections This text area displays the title of the currently opened script collection. If the collection in use hasn't been named, New Collection will be displayed.

To name a script collection, press the Edit Script File button. The Windows Notepad applet will be launched with the script file loaded. Look for the Collection: entry (it's on the first line of the script file) and overwrite New Collection with your new title. Then, save the file.

Note: The text in the Script Collections field will not reflect the name change until you reload the script.

Scripts List This area lists the names of all scripts in the currently opened collection. Highlight a script in the list before you edit or run it.

Open/New Collection Pressing this button brings up a standard Open dialog with which you can browse your system's folders for script (*.scp) files. Double-click an existing .scp file to open that script collection.

To create a new script collection, navigate through the dialog to the folder that you would like to save the .scp file, and enter a name for it in the File Name: field. Next, press Open to save the script file. It will then be ready for use in a new script collection.

To name a script collection, press the Edit Script File button. The Windows Notepad applet will be launched with the script file loaded. Look for the Collection: entry (it's on the first line of the script file) and overwrite New Collection with your new title. Then, save the file.

Note: The text in the Script Collections field will not reflect the name change until you reload the script.

Run Script Click on this button to run the currently selected script. If the script type isn't compatible with Adobe Audition's current state (i.e., trying to a Script Starts from Scratch script with a file open), the script won't run until Adobe Audition's condition matches the requirements (in this case, closing all open files).

Batch Run A single script can be run repeatedly over a group of source files. The script must have been recorded in a Works on Current Wave mode – that is, before the script was recorded, there must have been an open waveform (perhaps blank) and no highlighted selection. The Batch Run button is only selectable if a script of this type is selected from the Scripts List.

Note: For more information, see the section on Batch Processing later in this chapter.

Pause at Dialogs Check this option to have the script stop at each dialog, allowing you to modify the values of that function. Pressing Cancel in any dialog will stop the script. Pressing OK will continue it.

Alert when complete If this option is checked, a dialog will signal the completion of the script.

Execute Relative to Cursor When using Works on Current Wave types of scripts, check this option to have all script operations performed relative to the original positioning, as opposed to at the original position.

For example, suppose you have a script which was originally recorded with a cursor position at 0:10:00. Check this box to apply the script at the current cursor location, plus 10 seconds. (If your cursor is at 0:05:00, the results would be at 0:15:00.)

With this option unchecked, the script will be performed at the original location (0:10:00 in our example).



To have a script which you'll likely execute at the current cursor position, record the script at a 0:00:00 position and check this option.

Edit Script File This will open the currently loaded script collection in the Windows Notepad applet for editing.

Title Before creating a new script, you must first supply a title here.

Record After titling your soon-to-be script, click on the Record button to begin capturing the script's steps.

Add to Collection Pressing this button will add the script you've just created to the currently open script collection. Keep in mind that if you wish to supply a description for the script, do so in the description text area below before pushing this button.

Clear If you don't wish to add the script you've just recorded to the collection, push the Clear button. This button is only available immediately after pressing Stop. If you want to remove a script after it's been added, you'll need to edit the script by pressing the Edit Script File button.

Stop Current Script After completing all of the steps to be included in the script, return to Adobe Audition's Scripts and Batch Processing dialog and press Stop Current Script.

Description Immediately after recording a script, you may enter a description in this text area for the script you've just recorded. The description will appear in a non-editable form when you highlight the script to run.

Note: The only time you can enter a description in this box is after recording – not before and not after it's been added to a script collection file. If you want to edit the description thereafter, press the Edit Script File button to bring up the Windows Notepad applet with the script collection file loaded.

Script Type These three fields indicate what type of script is currently highlighted in the Scripts List. There are three possible script types: Script Starts from Scratch, Script Works On Current Wave, and Script Works on Highlighted Section. Each operates respectively on either the entire waveform, a selection of a waveform, or from scratch. The type of script currently selected will darken, leaving the other two "grayed out".

Note: Again, to create a particular type of script, simply match Adobe Audition's current state with that of the target for the script. For example, if you wish to make a script that starts from scratch, close all open waveforms before recording the script.

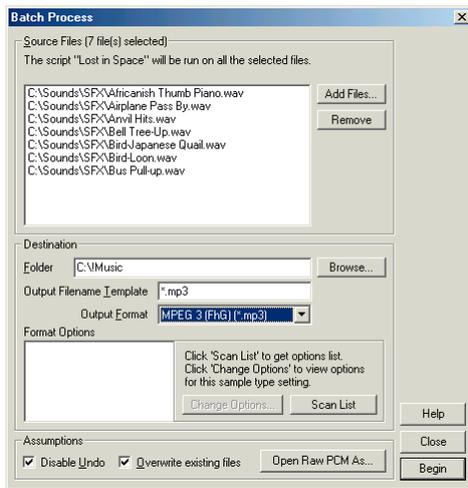
Note: While all scripts will be displayed in the script list, only scripts whose type matches your current status (open waveform with a highlighted selection, for instance) will be allowed to run.



A single script can be run on a batch of files by pressing the Batch Run button. For more information, see Batch Process.

Batch Process

Use this part of Adobe Audition to run a single script repeatedly over a group (batch) of source files. To open the Batch Process dialog, press the Batch Run button on the Scripts and Batch Processing window.



Batch Process Dialog

Note: In order for a single script to be run on a batch of files, the script must have been recorded in the Script Works on Current Wave mode. That is, before the script was recorded, there must have been an open waveform (perhaps blank) and no highlighted selection. The Batch Run button will only be selectable if a script of this type is selected from the scripts list.

Here are the major components of the Batch Process dialog:

Source Files This is the list of files you want to include in the batch for processing. If desired, the waveforms can be different formats.

Add Files This button brings up Adobe Audition's Please choose the source files dialog, allowing you to add files or folders to the batch for processing. To add multiple files or folders to the list, hold down on the SHIFT key for contiguous selections, or the CTRL key for noncontiguous selections.

Remove This button removes any selected files from the Source Files list. To remove multiple files from the list, hold down on the SHIFT key for contiguous selections, or the CTRL key for noncontiguous selections.

Destination Folder This is the folder to which the processed files will be saved after the script has been run on them. Select the destination folder by typing in its path in this field or by pressing the Browse... button.

Output Filename Template The names of files in your batch can be modified before being saved to the Destination folder. When running a batch, the processed file's extension will automatically change to that of the format chosen using the Output Format control (for instance, *.aif for Apple AIFF). You can, however, force another extension or alter the first part of the filename by using the Output Filename Template. There are two characters to use when altering the Output Filename Template:

- A question mark (“?”) will signify that a character doesn't change.
- An asterisk (“*”) will denote the entire original file name or entire original file extension.

Here are some examples of how filenames will be saved given the original file name and the filename template:

zippy.aif	*.wav	zippy.wav
toads.pcm	q*.voc	qtoads.voc
funny.mp3	b???????.*	bunny.mp3
biglong.au	????.au	bigl.au
bart.wav	*.x.wav	bartx.wav

Output Format This selects the file format that all processed waveforms will be saved to. You may choose from any of Adobe Audition's supported file types, as well as enter any appropriate options for the target format by pressing the Change Options... button, if available.

Change Options Press this button to view options (and optionally change them) for each highlighted sample type setting in the Format Options box.

Note: You would almost always use the Scan List button (described below) before clicking on Change Options.

Scan List Pressing the Scan List button causes Adobe Audition to look through all of the files in the Source Files list and examine their sample rate, bit depth, channels and file type. After this is done, the options can be set for the output format, depending on the properties of the input.

Disable Undo Check this box to disable the saving of undo information for the duration of the batch run. Unless the script was written expecting the Undo function to be enabled, this is a completely safe thing to do. Plus, it speeds up processing since undo information doesn't continually need to be saved on your hard drive.

Overwrite Existing Files Check or uncheck this box to control overwriting.

- With this option checked, files will always be saved to the Destination folder after performing the script, regardless of whether a file of the same name already exists at this location.
- If this box is unchecked, and the destination filename already exists, the batch will not even attempt to run the script on the file in question, but skip it instead.

Open Raw PCM As If a source file is unreadable, or in a raw format without any header information, then Adobe Audition’s Batch Processor needs to know what file type to assume for the data. Pressing the Open Raw PCM As... button allows you to define a global file type for interpretation in such cases; you will need to set the sample rate, channels, resolution, and formatting. This ensures that the batch process will run continuously without interruption by dialogs asking for input data formats on headerless files.

Begin Press this button to begin batch processing all files in the Source Files list.

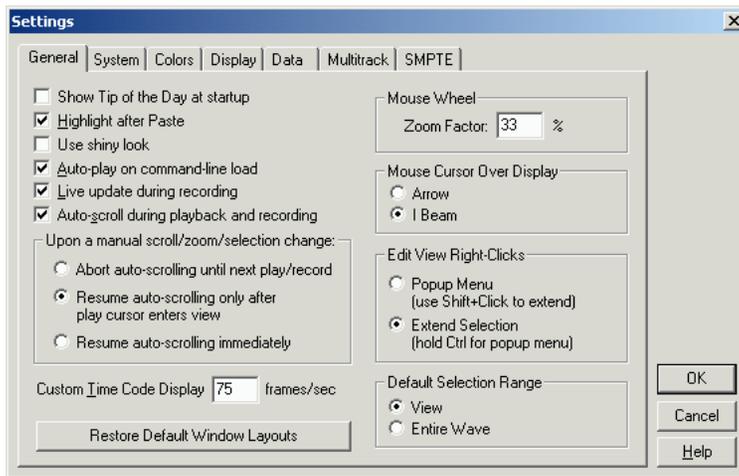
 *If you want to convert several files from one format to another (say, from .wav to .mp3), try using File > Batch File Convert instead.*

Settings

You can customize Adobe Audition’s colors, use of memory and hard disk space, spectral view, behavior when pasting, and miscellaneous other settings in Options > Settings. See below for information on the individual settings available.

 *Pressing the F4 key is a fast way to display the Settings dialog.*

General Tab The General Tab of the Settings dialog provides controls for adjusting Adobe Audition’s mouse behavior, as well as parameters for auto-play, live update, auto-scroll, and more.



The General Tab of the Settings Dialog

Show Tip of the Day When this box is checked, a tip about program operation is displayed whenever Adobe Audition is launched. If you’d rather not see a tip when Adobe Audition loads, uncheck this box.

Use shiny look If you'd prefer that certain elements of Adobe Audition's interface (toolbar buttons mostly) take on a flatter look, uncheck this box. The flat look will take effect the next time you start Adobe Audition. While there aren't any performance benefits to not using the default shiny look, Adobe Audition might look better if your video display is set to 256 colors.

Auto-play on command-line load When launching Adobe Audition, checking this option will automatically play a file that has been specified on the command line. For example, if you go to the Run dialog in the Windows Start menu and enter "c:\cool\coolpro.exe thisfile.wav" as your command line, then Adobe Audition will launch and begin playing thisfile.wav.

Live update during recording This enables live waveform drawing while recording. On faster machines, you can have the waveform displayed in real-time as audio is being recorded. However, if you find the recorded audio becoming choppy, leave this option disabled.



In the Edit View's Spectral View mode, and at lower spectral resolutions (around 256) a nice scrolling spectral plot can be performed while recording with this option on.

Auto-scroll during playback and recording If enabled, the waveform display will scroll in sync with playback. This only affects when you are zoomed in on a portion of a waveform, and play past the viewed portion (by pressing Shift+Play, for example).

Upon a manual scroll/zoom/selection change: When a manual scroll, a zoom, or a selection change occurs in Adobe Audition, three options are available: You can abort auto-scrolling until the next time you play or record; resume auto-scrolling only when the play cursor enters the view; or resume auto-scrolling immediately. Choose the one that best suits your needs.

Custom Time Code Display This field is used to define the number of frames per second (FPS) assigned to the Custom time format in the View > Display Time Format menu.

Restore Default Window Layouts Press this button to reset all window sizes and positions to Adobe Audition's default arrangement.

Ctrl key allows dockable windows to dock By default, holding down the Ctrl key when moving one of Adobe Audition's dockable windows prevents the window from docking. Check this box if you'd prefer to disable this behavior.

Mouse Wheel

- **Zoom Factor:** This is the amount to zoom in when rotating the mouse wheel found on Intellipoint-compatible pointing devices. Values from 10% to 80% work well. The higher the value, the further you'll zoom in when you roll the mouse wheel.

Mouse Cursor Over Display Choose whether you want your mouse pointer to appear as an arrow or as an I-beam when it's over the waveform display.

Edit View Right-Clicks These options let you decide the behavior for a right-click in the Wave Display of the Edit View.

- **Popup Menu:** When right-clicking in the Wave Display, a menu will pop up if this option is selected. You can then use Shift+Left-Click to extend a selection.
- **Extend Selection:** If you check this option, right-clicking in the Edit View's Wave Display will let you extend the edge of a waveform selection instead of displaying the pop-up menu. To see the pop-up menu, hold down the Ctrl key as you right-click.

Default Selection Range This setting refers to the amount of waveform data that automatically gets selected (if nothing is already highlighted) when you go to apply an effect.

- **View:** If this option is selected, the area that's automatically selected is limited to the area you can currently see on screen.
- **Entire Wave:** When you choose this option, the entire waveform is automatically selected, even if you're only viewing a portion of it.

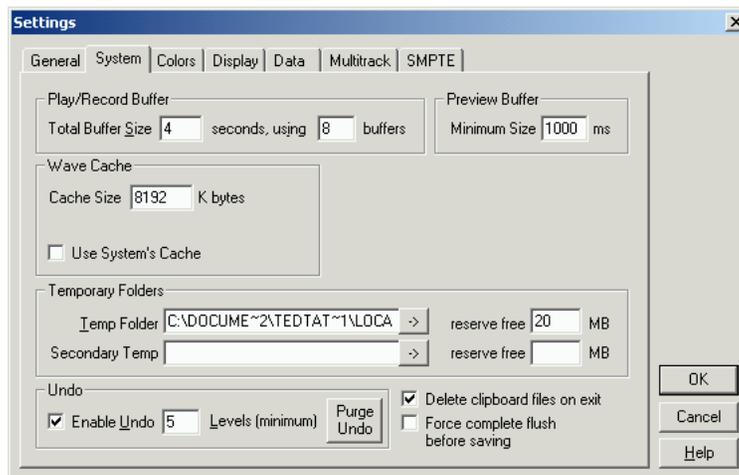
 *Double-clicking always selects the current view. Triple-clicking always selects the entire waveform.*

Highlight after Paste When performing a Paste operation (including Mix Paste), you can elect to have the inserted selection automatically highlighted, or to have the cursor placed at the end of the pasted selection. Check this box to enable highlighting.

 *Leave this option unchecked for easier multiple pastes, one after the other.*

System Tab

Adjust the options in this section of the Settings window to tell how Adobe Audition should interact with your system.



The System Tab of the Settings Dialog

Play/Record Buffer Different sound card drives can require different memory buffer settings. Adobe Audition's default settings should work fine for most sound cards, but if you hear "choppiness" (skips or dropouts) in recording or playback, you may need to adjust the buffer size or number of buffers used.

Use the two fields in the Play/Record Buffer area to reserve more memory for recording and playback by entering a higher buffer size, both in seconds and a number of buffers.

Keep in mind that while a greater buffer size will allow for increased multitasking when audio is being played, it does so at the expense of taking more of your computer's memory.

 *If you experience breakups in your audio, or you can't stop a recording in progress, increase the buffer size or switch to a faster, non-compressed hard drive.*

Wave Cache Adobe Audition maintains its own data buffer, and reserves for it the amount of memory (measured in MB) specified in the Cache Size field. Recommended cache sizes are from 8192KB to 32768KB (8192KB is the default).

- Use System's Cache: Check this box to let Windows handle all disk caching. Keep in mind that Adobe Audition usually handles caching better than Windows can. However, this option reserves the least amount of memory, so it may be desired for systems with low amounts of RAM.

Preview Buffer This is the minimum buffer size to be used when sending data to your sound card for the real-time Preview feature found in many effect dialogs. The default value is 1000 milliseconds.

Different sound card drives can require different memory buffer settings. If you hear "choppiness" (skips or dropouts) when you use the Preview feature, you can try to remedy it by adjusting the buffer size used. (Choppiness can be caused by insufficient processing power as well.) Keep in mind that a larger Minimum Preview Buffer Size will require more of your computer's memory.

Temporary Folders Adobe Audition creates temporary files for use when performing edits on your audio. Use these two fields to specify the paths to folders in which these files will reside. You will need to have enough space available in these folders to accommodate the total size of all the audio files you wish to edit simultaneously.

Use the reserve free fields to specify an amount to leave available for headroom purposes for both the primary and secondary temporary folders.

- Temp Folder: This is where you specify Adobe Audition's main temporary folder. Ideally it should be on your fastest hard drive. Press the "->" button to browse your system for the folder, or type its path in the field provided.
- Secondary Temp: This is where you specify Adobe Audition's secondary temporary folder. For best results, this should be on a different physical hard drive than the primary temp folder. Press the "->" button to browse your system for the folder, or type its path in the field provided.

Note: *Providing you have enough free space on the drive that holds the primary temporary folder, Adobe Audition will work just fine if no Secondary Temp folder is specified.*

Undo Everybody makes mistakes or changes their mind. Adobe Audition's Undo feature lets you revert back to your last edit with the press of a hotkey (Ctrl+Z), menu command, or toolbar button. Think of it as a virtual "safety net".

- **Enable Undo:** If this box is checked, the Undo function is active. Because Undo requires extra desk space for its temporary files, and time to save them before processing, sometimes you may want to turn this feature off by unchecking this box.
- **Levels (minimum):** Specify the least amount of Undo levels in this field.
- **Purge Undo:** Push this button to delete all of Adobe Audition’s Undo files. This will free up disk space, but you won’t be able to revert to any previous edits.

Delete Clipboard files on exit If this option is checked, Adobe Audition automatically purges its clipboard files when it’s closed. It’s generally a good idea to leave this option enabled, since nearly all of the time when you’re finished with an Adobe Audition session, these clipboard files are no longer needed and would just take up valuable hard disk space.

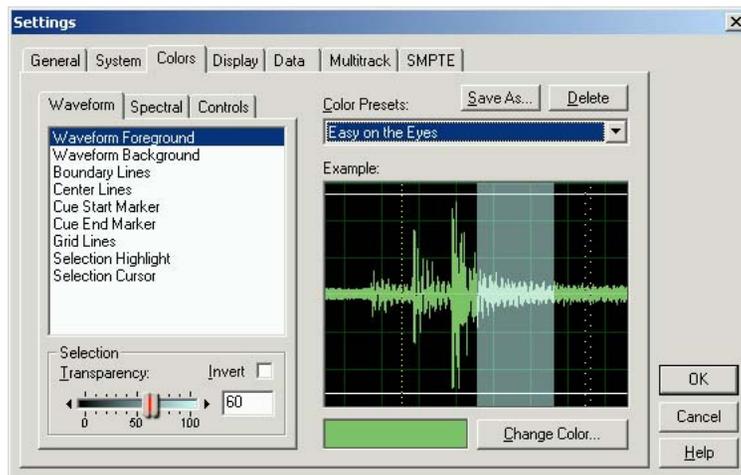
Uncheck this box to retain Adobe Audition’s clipboard files on your hard drive after you exit the program.

Force complete flush before saving When checked, this option disables Adobe Audition’s quick save feature. Normally, Adobe Audition quick saves a file if only a small portion of the file was modified. If you force flushing before saving, Adobe Audition must first make a backup copy of the file internally and then write the entire file back during the save process.

This option is disabled by default. It was added for users who have problems saving back to the same filename. Enabling this option will increase the save time considerably for large files. Don’t enable it unless you have a problem with Adobe Audition’s quick save function.

Colors Tab

Don’t care for Adobe Audition’s default color scheme? This section of the Settings window is where you can customize its interface’s various hues and shades to your heart’s content.



The Colors Tab of the Settings Dialog

Color Scheme Presets Most color assignments in Adobe Audition can be adjusted to your liking and saved as a preset for later recall. This drop-down menu contains all of the presets that come with the program as well as the ones you've created yourself. To choose one, simply select it from the list. The currently selected color scheme will be displayed in the Example window.

Save As Press this button to save the currently selected color scheme as a preset.

Delete Clicking this button deletes the currently highlighted color scheme preset.

Waveform Tab This list contains all of Adobe Audition's waveform elements which can be given custom colors. Choose an item from the list and press the Change Color... button to bring up a dialog for adjusting the element's color. After the new selection has been made, the example to the right will change to reflect the newly chosen color.

Selection The controls in this section are used to adjust the appearance of the selected (highlighted) portions of waveforms and blocks. The example to the right dynamically changes as you make modifications.

- **Transparency:** Use this slider to adjust the transparent value (measured in percentage) of a selection; 0 is no transparency, while 100 is maximum transparency. You may also directly enter a percentage in the box to the right of the slider.
- **Invert:** Check this box to have the selection colors be the inverse of the non-selected colors. The Transparency controls are disabled.

Spectral Tab

This is where you can adjust the display parameters for Adobe Audition's spectral display. Pick an item from the list and press the Change Color... button to bring up a dialog for adjusting the element's color. After you've made your selection, the example to the right will change to reflect the newly chosen color.

Spectrum Choose one of these options:

- **Reverse Direction:** This option inverts the normal colors of the spectrum display, similar to an Invert or Negative command in a photo editor.
- **Gamma:** This setting works just like the Gamma function in many image editors. The value entered adjusts the overall brightness of the Spectral View. Positive numbers make the display brighter, while negative numbers darken the display.

Selection The controls in this section are used to adjust the appearance of the selected (highlighted) portions of a waveform when shown in spectral display. The example to the right dynamically changes as you make modifications.

- **Transparency:** Use this slider to adjust the transparent value (measured in percentage) of a selection; 0 is no transparency, while 100 is maximum transparency. You may also directly enter a percentage in the box to the right of the slider.
- **Invert:** Check this box to have the selection colors be the inverse of the non-selected colors. The Transparency controls are disabled.

Controls Tab

This window lists the Adobe Audition control elements which can have their colors changed. Select an item from the list and press the Change Color... button to bring up a dialog for adjusting the element's color. After the new choice has been made, the example to the right will change to reflect the newly chosen color.

Segmented Progress Bar Select this option to have Adobe Audition's progress bar (which is seen when you run an effect or open/save large waveforms) appear segmented instead of solid.

White Progress Background Check this box to make the background of Adobe Audition's progress bar white.

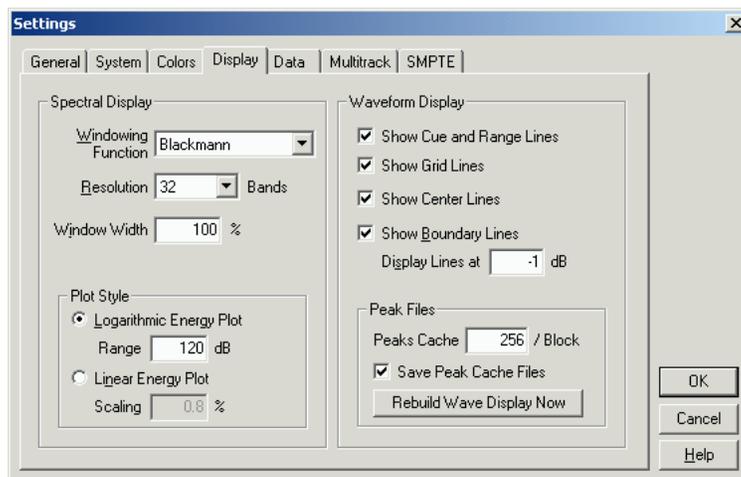
Dockable Windows Select one of these options to decide how Adobe Audition's dockable windows will look.

- Use system 3D color: With this option checked, dockable windows will take on your system's 3D color (the same color that Windows uses to render most of the other windows on your system).
- Used darkened system 3D color: Check this option to have dockable windows use the darkened version of your system's 3D color.
- Use specified 3D color: Enabling this option will cause Adobe Audition's dockable windows to take on the 3D color you specify.

To change the 3D color, make sure Dockable Windows 3D Color is highlighted in the controls list. Then, press the Change Color... button to access the Adobe Audition Color dialog to select a new color.

Display Tab

In this area is where you choose options for Adobe Audition's Spectral View and Waveform View modes.



The Display Tab of the Settings Dialog

Spectral Display Choose an option:

- **Windowing Function:** This drop-down list lets you choose the method Adobe Audition uses to window the spectral data before it displays it. The windows are listed in order from the narrowest frequency band/most noise to the widest frequency band/least noise. Blackmann or Blackmann-Harris are good choices.
- **Resolution:** This setting specifies the number of vertical bands to be used in drawing frequencies. Keep in mind that the larger this number is, the longer it will take for Adobe Audition to render the spectral display. Performance will vary based upon the speed of your computer.
- **Window Width:** This is the width of the window (or frame size) used in plotting the spectral data, where 100% is a frame size of your FFT size. Window Width basically lets you increase time resolution at the expense of some frequency resolution. So the display will become more accurate along the timeline (left and right) and less accurate along the frequency scale (up and down) as the window width decreases. The default setting is 75%, but you should lower the value (50 to 75 works best) if you want to increase the resolution horizontally, such as to find out exactly where a certain frequency starts.

Plot Style Choose an option:

- **Logarithmic Energy Plot:** In this mode, colors change with the decibel value of the energy at any particular time and frequency. More details in the very quiet ranges can be seen in this mode, especially if the Range is quite high (above 150dB). Use the Range value to adjust the sensitivity in plotting frequencies.
- **Linear Energy Plot:** If this option is selected, colors are chosen based on percentage of maximum amplitude instead of decibel amplitude. Linear Energy Plot can be useful for viewing the general overview of a signal without getting bogged down by detail at much quieter levels. The Scaling factor can be adjusted to highlight audio of different intensities, and can be thought of as a sensitivity value.

Waveform Display Choose an option:

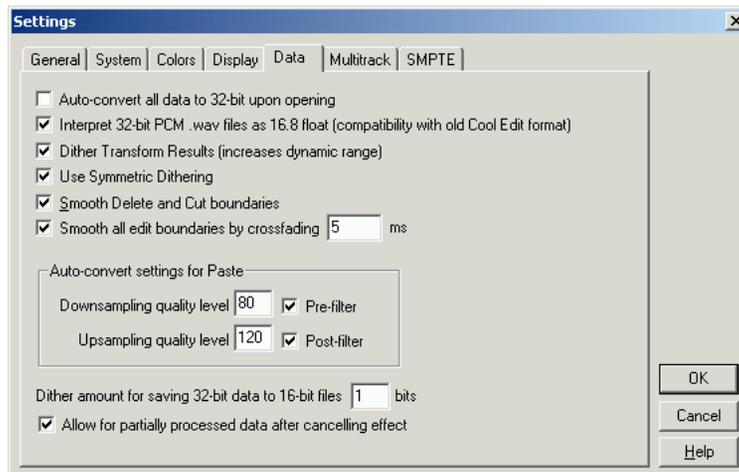
- **Show Cue and Range Lines:** Check this box to display the cue marker and range lines in Waveform View; uncheck the box to hide them. Cue marker and range entries in the Cue List are shown with vertical dotted lines overlaying the audio, connecting the arrows from the top to the bottom of the display.
- **Show Grid Lines:** Check this box to display the grid lines in Waveform View; uncheck the box to hide them. The grid lines mark off time on the horizontal x-axis and amplitude on the vertical y-axis.
- **Show Center Lines:** Check this box to display the center lines in Waveform View; uncheck the box to hide them. The center lines represent zero amplitude of the waveform's right and left channels.
- **Show Boundary Lines:** Check this box to display the boundary lines in Waveform View; uncheck the box to hide them. Boundary lines are the horizontal lines that visually indicate where the waveform's amplitude approaches or exceeds the clipping level. The value in the Display Boundary Lines at box specifies at which amplitude you'd like the boundary lines to appear

Peak Files Adobe Audition's peak (.pk) files store information about how to display .wav files, and can make file loading almost instantaneous by greatly reducing the time it takes to draw the waveform (especially with larger files).

- **Peaks Cache:** This value determines the number of samples per block to be used when storing peak files. Larger values will reduce the RAM requirement for large files at the expense of slightly slower drawing at some zoom levels. If RAM is an issue on your system, and you're working with very large files (several hundred megabytes or more in size), you should consider increasing the Peaks Cache to 1024 or even 1536 or 2048.
- **Save Peak Cache Files:** If this option is enabled, all .wav files will have peak files saved with them (in the same folder) with the extension .pk following the original audio filename. Uncheck this box if you don't want Adobe Audition to save peak files on your hard drive.
- **Rebuild Wave Display Now:** Press this button to have Adobe Audition rescan the current file for sample amplitudes and redraw the waveform.

Data Tab

The controls in this area of the Settings dialog have to do with the way Adobe Audition handles audio data.



The Data Tab of the Settings Dialog

Auto-convert all data to 32-bit upon opening If this item is checked, all 8-bit and 16-bit data is automatically converted to 32-bit when a file is loaded, and all subsequent operations will keep the data in the 32-bit realm.

Interpret 32-bit PCM .wav files as 16.8 float (compatibility with old Cool Edit format) Checking this option causes this version of Adobe Audition to be compatible with previous versions when it comes to handling 32-bit PCM .wav files.

[[how should we handle reference above to Cool Edit?-GS]]

Dither Transform Results (increases dynamic range) Check this option to enable dithering when processing effects such as FFT Filter or Amplify. Most processing done by Adobe Audition uses arithmetic greater than 16-bit, with the results converted back to 16-bit when complete. During this conversion, dithering provides a higher dynamic range and cleaner results, with less distortions and negative artifacts.

If this option is disabled, the results are truncated to 16 bits when converting back, thus losing the more subtle information.

When enabled, the addition of dither retains this subtle information. The drawback is that with each operation a small amount of white noise is added at the quietest volume level. However, the trade-off between using dither (thus adding noise) and truncating the data (thus creating artifacts and correlated quantization noise) generally favor using dither, so it's best to leave this option enabled.

With dithering, you get almost 24-bit sample performance in only 16-bits, as the dynamic range is increased by another 10dB or so, allowing signals as quiet as -105dB.

Use Symmetric Dithering Normally you would leave this option checked. If unchecked, a DC offset of one-half sample will be added each time something is dithered. Symmetric dithering has just as many samples added above zero as below zero. By contrast, non-symmetric dithering just toggles between 0 and 1. Sometimes in a final dither, this may be desired to reduce the bit range of the dither. However, both methods produce identical audible results in every respect.

Smooth Delete and Cut Boundaries Check this box to have Cut and Delete operations smoothed at the splicing point. This will prevent audible clicks at these locations.

Smooth all edit boundaries by crossfading When applying an effect, enable this option to automatically apply a crossfade to the starting and ending boundaries of the selection. This will smooth any abrupt transitions at these endpoints, thus preventing audible clicks when filtering small portion of audio. You can enter a value (in milliseconds) in the crossfade time box to specify the crossfade duration to be applied.

Auto-convert settings for Paste When pasting different sample formats, Adobe Audition uses these settings when auto-converting the clipboard to the current sample format. Valid settings range from 30 to 1000.

- Downsampling quality level: Enter a value (30 to 1000) for downsampling quality.
 - Higher values retain more high frequencies while still preventing the aliasing of higher frequencies to lower ones.
 - A lower quality setting requires less processing time, but will result in certain high frequencies being “rolled-off”, leading to muffled-sounding audio.

Because the filter's cutoff slope is much steeper at higher quality settings, the chance of ringing at high frequencies is greater. Usually values between 80 and 400 do a great job for most conversion needs. The default value is 80.

- Pre-Filter: To prevent any chance of aliasing, the pre-filter on downsampling will remove all frequencies above the Nyquist limit, thus keeping them from generating false frequencies at the low end of the spectrum. In general, this box should be checked for best results.
- Upsampling quality level: Enter a value (30 to 1000) for upsampling quality.
 - Higher values retain more high frequencies while still preventing the aliasing of higher frequencies to lower ones.

- A lower quality setting requires less processing time, but will result in certain high frequencies being “rolled-off”, leading to muffled-sounding audio.

Because the filter’s cutoff slope is much steeper at higher quality settings, the chance of ringing at high frequencies is greater. Usually values between 100 and 400 do a great job for most conversion needs. The default value is 120.



You should use high quality settings whenever you downsample from a high sample rate to a low rate. When upsampling, a low quality setting will sound almost identical to a high quality setting. The difference lies in the larger phase shift that exists at higher frequencies, but since the phase shift is completely linear, it’s very difficult to notice. Downsampling, at even the lowest quality setting, generally won’t introduce any undesired noisy artifacts. Instead, it may just sound slightly muffled because of the increased high end filtering.

- Post-Filter: To prevent any chance of aliasing, the post-filter on upsampling will remove all frequencies above the Nyquist limit, thus keeping them from generating false frequencies at the low end of the spectrum. In general, this box should be enabled for best results.

Dither amount for saving 32-bit data to 16-bit files This option enables/disables dithering when pasting 32-bit audio to 16-bit. The default value of 1 (bit) will enable dithering, while a value of 0 will disable dithering. For semi-dithering, choose a value of 0.5.

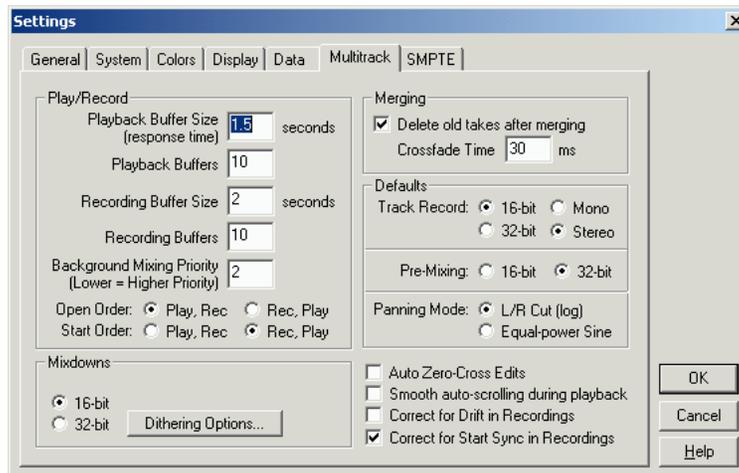
With dithering, you get almost 24-bit sample performance in only 16-bits, as the dynamic range is increased by another 10dB or so. This allows signals as quiet as -105dB.

Allow for partially processed data after canceling effect This option decides what happens after you hit the Cancel button while in the middle of applying an effect to a waveform.

- If the box is checked, Adobe Audition will leave the effect applied to all data processed up until the point you pressed Cancel.
- Uncheck this box to have Adobe Audition automatically remove the effect on already processed data when Cancel is pushed.

Multitrack Tab

Use these settings to adjust system performance and default parameters for Adobe Audition’s Multitrack View.



The Multitrack Tab of the Settings Dialog

Play/Record Set the following:

- **Playback Buffer Size:** This is the buffer size (measured in seconds) to be used when sending data to your sound card when playing back a multitrack session. Different sound card drivers can require different memory buffer size settings. Adobe Audition's default settings should work fine for most sound cards. If you hear "choppiness" (skips or dropouts) in multitrack playback, you can try to remedy it by adjusting the buffer size used. (Choppiness in multitrack playback can also be attributed to the background mixing process not being far enough ahead). A greater buffer size will require more of your computer's memory. The default setting is 1.5 seconds.
- **Playback Buffers:** This field is used to adjust the number of buffers Adobe Audition uses for playback in the multitrack environment. If you experience break-up in your audio, try reducing the number of buffers. Increasing the number of buffers may also help for some configurations. The default setting is 10.
- **Recording Buffer Size:** Use this field to reserve memory for recording in a multitrack session by entering a buffer size (in seconds). Different sound card drivers can require different memory buffer size settings. Adobe Audition's default settings should work fine for most sound cards. If you experience dropouts while recording in multitrack (especially when playback seems fine), try increasing this setting. (First, be sure the background mixing process is sufficiently complete when you go to record as this may cause the same symptom.) A greater buffer size will require more of your computer's memory. The default setting is 2 seconds.
- **Recording Buffers:** This field is used to adjust the number of buffers used for recording in the multitrack environment. If you do experience break-ups in your audio, try reducing the number of buffers. Increasing the number of buffers may also help for some configurations. The default setting is 10.
- **Background Mixing Priority Level:** Here is where you can adjust the priority level of the background mixing process done in a multitrack session. Lower values indicate a higher level of priority above other system events. Fractional numbers (such as 0.8) may be used. The default setting is 2.

- **Open Order:** This setting determines the order that the sound card device driver will be opened for use in the multitrack environment. Because the multitrack environment requires full-duplex capability, it will “open”, or ready, both playback (out) and record (in) on the card. Some cards require this order to specific, but most don’t.

Note: Owners of the Digital Audio Labs CardD should set this option to Rec, Play. Owner of Sound Blaster cards should set this option to Rec, Play.

- **Start Order:** After opening each device driver (see Open Order), this setting determines the order that the sound card device driver will be started in the multitrack environment. Because the multitrack environment requires full-duplex capability, it will start both playback (out) and record (in) on the card. Some cards require this order to specific, but most don’t.

Note: Owners of the Digital Audio Labs CardD should set this option to Play, Rec. Owner of Sound Blaster cards should set this option to Rec, Play.

Mixdowns When performing a mixdown, this is the bit-resolution that will be used. Regardless of the session format (16-bit or 32-bit), mixdowns can be generated at either 16-bit or 32-bit quality with this option. The default is 16-bit.

Press the Dithering Options... button to bring up a dialog that lets you specify how you want the 16-bit mixdown to be dithered.

Merging Set the following:

- **Delete old takes after merging:** Check this box to have Adobe Audition automatically delete any unused takes created during a Punch-In when you select a take to go with. If this setting is off, unused takes remain available to the Session (in the Insert menu) and occupy hard drive space.
- **Crossfade Time:** This is the amount of time (measured in milliseconds) over which crossfading occurs when a take created using Punch-In is merged back into the surrounding waveform.

Defaults Set the following:

- **Track Record:** Check the desired radio buttons to specify how waveforms are created when recording directly into the Multitrack View: as mono or stereo, and as 16-bit or 32-bit.
- **Pre-Mixing:** This is the bit size used for the background mixing process. Best quality is achieved by leaving this at the default 32-bit setting. However, if you’re using multiple sound cards, it may be advantageous to choose 16-bit for pre-mixing, as less data will be transferred across the hard drive(s), which will speed things up. For single output device situations, or faster hard drives, 32-bit is better as it provides optimization at mixdown.

Panning Mode This section sets the method used for panning waveforms in a multitrack session.

- **L/R Cut (log):** If this option is chosen, left-panning is achieved by reducing the volume of the right channel, and right-panning is performed by reducing the left channel volume. The channel being panned to doesn’t increase in volume as panning gets closer to 100%.
- **Equal-power Sine:** Choose this option to pan left and right channels with equal power, so a hard pan left will contain the same loudness as both channels together. This results in an increase of 3dB RMS on the channel being panned to when at 100%.

Note: Because panning can actually make one channel louder than the original waveform, audible clipping can occur in 16-bit sessions. To avoid this, work in the 32-bit realm if you're using the Equal-power Sine panning method.

Auto Zero-Cross Edits Check the Auto Zero-Cross Edits box to have Adobe Audition automatically adjust the beginning and end points of all Cut, Copy, and Paste-type edits to the nearest place where the waveform crosses the center line (zero amplitude point).

If the amplitudes aren't lined up on both sides of the selection, the endpoints are at different amplitudes. This often results in an audible pop or click at that point.

Smooth auto-scrolling during playback By default Adobe Audition uses a "paging" method of scrolling in Multitrack View instead of the smooth scrolling technique used in Edit View. This saves on system resources. If you'd rather use smooth scrolling when playing back audio in Multitrack View, check this box.

Correct for Drift in Recordings If this box is checked, Adobe Audition will look at the sync between the "master" audio playback device (generally, the first Out device listed in the session – the one on Track 1) and the record device of the waveform being recorded. If the true sample rates on the cards differ enough that the recording would have drifted out of sync with the original if both were played back at exactly the same sample rate, then the recording is corrected by resampling to make it the proper length. This option only works with new record tracks, not with recording on top of existing waveforms, or punch-ins.

Note: On sound cards that support sample accurate devices (i.e., synchronized device starting, and all devices keyed off of the same clock) you don't need to check this box. This option allows for some measure of near sample-accurate synchronization across different sound cards, or when using with a single sound card that doesn't use the same clock for playback and recording (such as the Sound Blaster Live! card).

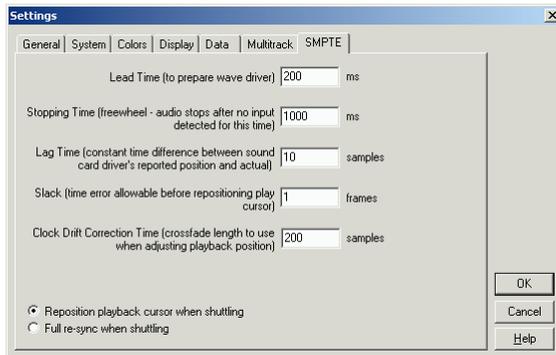
Correct for Start Sync in Recordings If this box is checked, Adobe Audition looks at the exact true time that the record device started and compares that with the time the master playback device started. If different, the recorded block's position is adjusted so the recording starts in perfect sync with the playback. This option only works with new record tracks, not with recording on top of existing waveforms, or punch-ins.

Note: If you have this option enabled, and you do a loopback test (by connecting the audio Out to the audio In and recording some "ticks") and each recording is still a fixed amount out of sync, then you can adjust for this by entering this amount (in milliseconds) in the Latency field of Options > Device Properties for the recording device being used. To compute milliseconds, look at the difference in samples, multiply by 1000 and then divide by the sample rate. For instance, if the recording consistently appears 27 samples ahead of the playback, the latency would be $27 * 1000 / 44100$ or 0.61 ms. (The reason for the milliseconds format and not samples is because at various sample rates this latency will be different in terms of samples, but will be the same in terms of milliseconds.)

Note: On sound cards that support sample accurate devices (i.e., synchronized device starting, and all devices keyed off of the same clock) you don't need to check this box. This option allows for some measure of near sample-accurate synchronization across different sound cards, or when using with a single sound card that doesn't use the same clock for playback and recording (such as the Sound Blaster Live! card).

SMPTE Tab

This is where you adjust Adobe Audition's settings for working with time code.



The SMPTE tab of the Settings dialog

Lead Time This sets is the amount of time (measured in milliseconds) that Adobe Audition has to establish sync with the incoming time code. Lower settings (200ms) will result in faster response (when pressing Play), but at the risk of true sync not being established. 500ms to 1000ms should provide plenty of Lead Time to establish lock-up. The default value is 200ms.

Stopping Time Stopping Time is the amount of time (measured in milliseconds) Adobe Audition will continue playing if a dropout in the time code is encountered. The default value is 1000ms.

Lag Time This setting, measured in samples, fine tunes the relationship between time code coming in and audio data coming out. It can be used to make up for any discrepancies introduced by your sound card buffers. The default value is 10 samples.

Slack This sets the number of frames CEP can periodically be “out of sync” with the time code before either repositioning the playback cursor to match the code, or perform a full re-sync. Generally a setting of about 2.5 frames will do, as there are occasions when the time code frame location numbers may be incorrect, usually corrected on the next frame sent. This will not cause a sync problem in the audio sense. The default value is 1 frame.

Clock Drift Correction Time This specifies how many samples to crossfade when making time corrections to audio played back, so the playback chases the time code. The default value is 200 samples.

Reposition playback cursor when shuttling If selected, the playback position will readjust if the sync is off by the Slack amount.

Full re-sync when shuttling If selected, a full re-sync will be performed if the sync is off by the Slack amount.

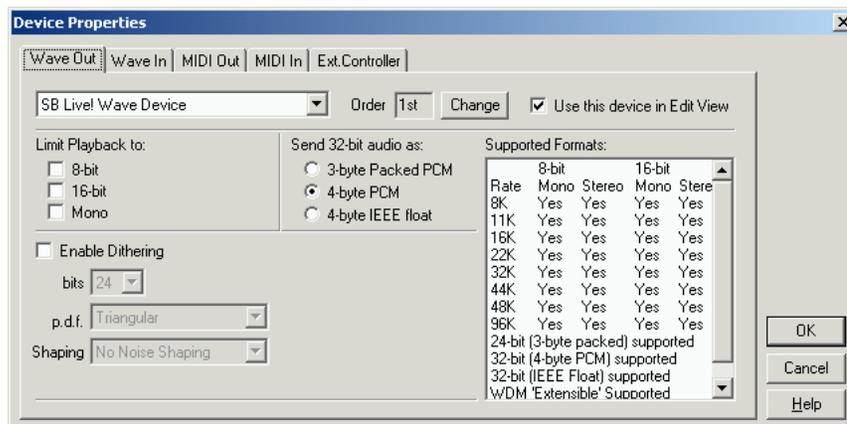
Device Properties

The Device Properties dialog is where you tell Adobe Audition which devices to use for playback and recording of waveforms, as well as input and output of MIDI signals.

Wave Out Tab

This area is where you specify Adobe Audition's parameters for playback of waveforms. If you have multiple sound cards, or a single card that has multiple outputs, they'll show up as choices in the drop-down list at the top left corner of this window.

The capabilities of the selected output device are shown in the Supported Formats table. A Yes or No is shown for different combinations of sample rate and bit resolution. This table also shows what (if any) 32-bit formats the output device can handle, and whether it can accept the WDM driver extensible wave format.



The Wave Out Tab of the Device Properties Dialog

Order The Order field displays the device preference order for the highlighted output device. To change this order, press the Change button to bring up Adobe Audition's Device Ordering Preference dialog.

Use this device in Edit View Check this box to have Adobe Audition use the highlighted output device for playing waveforms in Edit View.

Limit Playback to: Use this option to compensate for limitations imposed by your hardware. For instance, if your sound card doesn't handle 32-bit audio correctly, you can have Adobe Audition limit the playback of 32-bit files to either 16-bit or 8-bit and/or mono.

Send 32-bit audio as: This option is enabled if you don't have any of the boxes checked in Limit Playback to. Depending on if your output device supports it, you can send 32-bit audio as 3-byte Packed PCM, 4-byte PCM, or 4-byte IEEE float.

Enable Dithering: Check this box to activate dithering during waveform playback. If dithering isn’t enabled, Adobe Audition truncates instead. This means that bits that aren’t used are simply chopped off and discarded. This gives a “crackly” effect that fades in and out on very quiet audio passages.

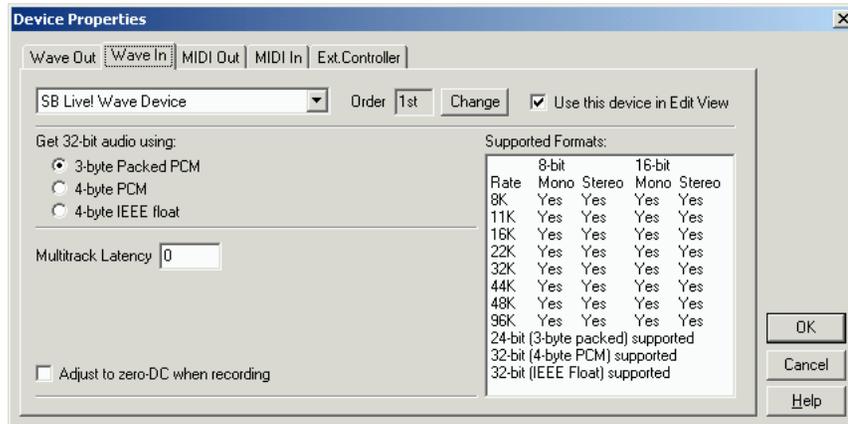
Note: Dithering only kicks in if you’re limiting high bit depth audio to a lower bit depth playback.

- **bits:** This drop-down list lets you choose the number of bits of Dither to be applied, from 8 to 32. Generally, values of (what) to (what) give the best results without adding too much noise. Note, however, that as this value is lowered, other unwanted harmonic distortion noise will appear.
- **p.d.f.:** The “probability distribution function” controls how the dithered noise is distributed away from the original audio sample value. Five p.d.f. types are available from the drop-down list: Rectangular, Triangular, Gaussian, Shaped Triangular, and Shaped Gaussian. Usually one of the Triangular p.d.f. functions is a wise choice, because it gives the best tradeoff between SNR, distortion, and noise modulation.
- **Shaping:** Select one of the several noise shaping presets (known as “noise shaping curves”) to have Adobe Audition determine the placement when moving noise to different frequencies. The same amount of noise overall is present, but you can choose less noise to be placed at one frequency at the expense of placing more noise at another. You can also specify that no noise shaping is used.

Wave In Tab

This area is where you specify Adobe Audition’s parameters for recording waveforms. If you have multiple sound cards, or a single card that has multiple inputs, they’ll show up as choices in the drop-down list at the top left corner of this window.

The capabilities of the selected recording device are shown in the Supported Formats table. A Yes or No is shown for different combinations of sample rate and bit resolution.



The Wave In Tab of the Device Properties Dialog

Order The Order field displays the device preference order for the highlighted recording device. To change this order, press the Change button to bring up Adobe Audition's Device Ordering Preference dialog.

Use this device in Edit View Check this box to have Adobe Audition use the highlighted output device for recording waveforms in Edit View.

Get 32-bit Audio using: Depending on if your recording device supports it, you can send 32-bit audio as 3-byte Packed PCM, 4-byte PCM, or 4-byte IEEE float.

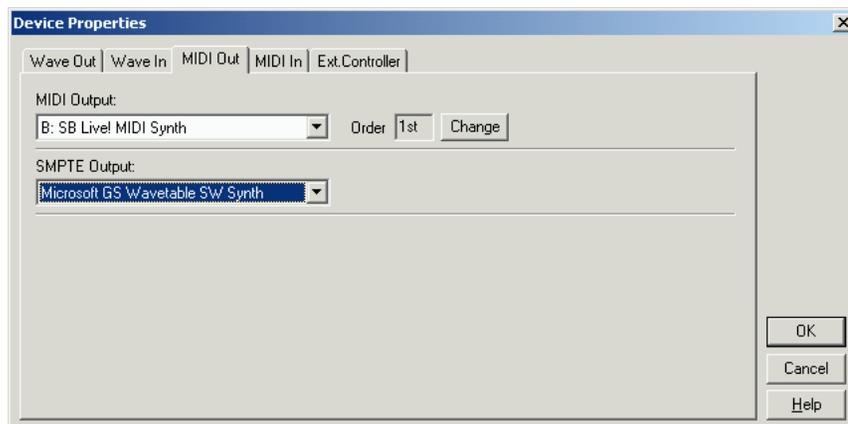
Multitrack Latency Some sound cards introduce a latency of several milliseconds when they record. For example, if you record on Track 1, and then record Track 2 while listening to Track 1 and find that the two tracks are out of sync by 7 milliseconds, it's probably because of this latency problem.

This is a constant number, though, so through trial and error you can figure out how out of sync a particular device gets, and then enter the number of milliseconds here (in our example, it would be 7) to force sync. (The default value is 0.)

Adjust to zero-DC when recording Check this box to have Adobe Audition automatically remove any detected DC bias when recording.

MIDI Out Tab

If your system is equipped with one or more MIDI devices, this area is where you specify Adobe Audition's parameters for MIDI and SMPTE output. If you have multiple MIDI devices, or a single device that has multiple outputs, they'll show up as choices in the two drop-down lists at the top left corner of this window.

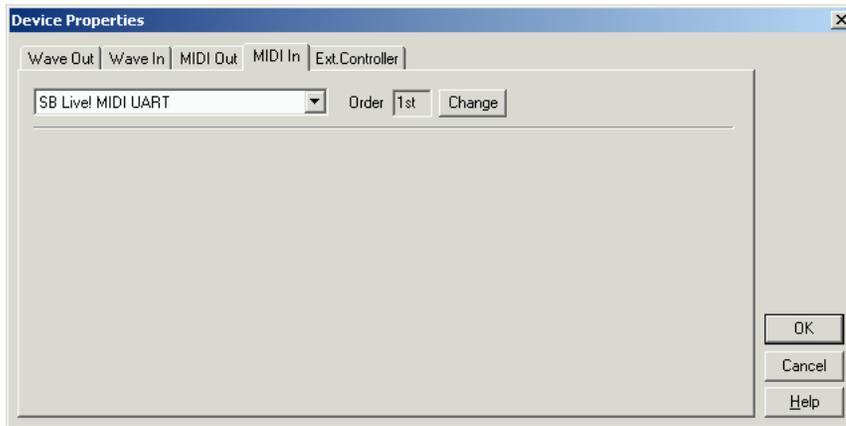


The MIDI Out Tab of the Device Properties Dialog

Order The Order field displays the device preference order for the highlighted MIDI Output device. To change this order, press the Change button to bring up Adobe Audition's Device Ordering Preference dialog.

MIDI In Tab

If your system is equipped with one or more MIDI devices, this area is where you specify Adobe Audition's parameters for MIDI input. If you have multiple MIDI devices, or a single device that has multiple inputs, they'll show up as choices in the drop-down list at the top left corner of this window.



The Wave In Tab of the Device Properties Dialog

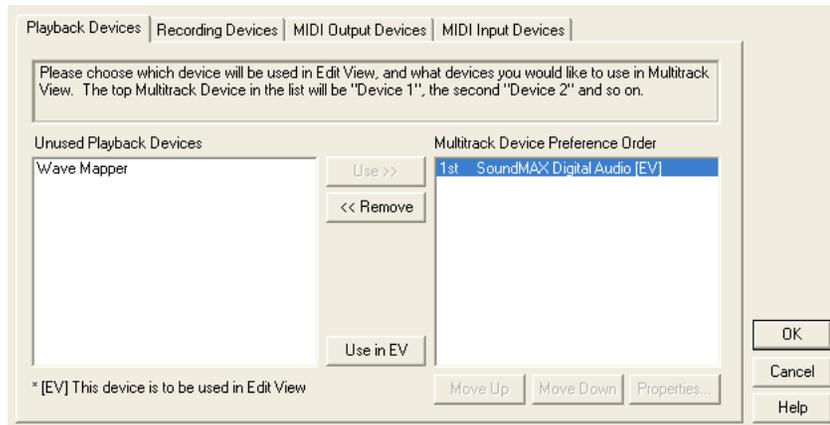
Order The Order field displays the device preference order for the highlighted MIDI Input device. To change this order, press the Change button to bring up Adobe Audition's Device Ordering Preference dialog.

Ext. Controller Tab If you have any external devices that Adobe Audition supports, you can select them here and adjust their Volume Increment value (specified in db).

Device Order

The Device Ordering Preference window allows you to set the order in which Adobe Audition uses your system's playback, recording, MIDI output, and MIDI input devices. The order of preference chosen determines which tracks get assigned to which devices when loading multitrack sessions on systems with different sound card configurations. Move between the tabs to adjust the parameters for each type of device.

Playback Devices Tab



The Playback Devices tab of the Device Ordering Preference Dialog

Unused Playback Devices The items in this list are the playback devices on your system not currently utilized by Adobe Audition.

Use>> Press this button to move a selected item from the Unused Playback Devices list to the Device Preference Order list.

<<Remove Click on this button to remove a highlighted item from the Device Preference Order list. It will then appear in the Unused Playback Devices list.

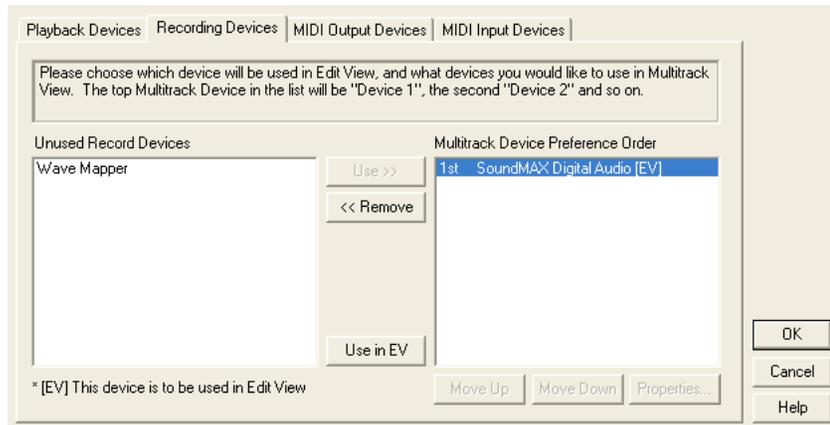
Move Up Pushing this button moves the selected item up in the Device Preference Order list.

Move Down Pressing this button moves the highlighted item down in the Device Preference Order list.

Use in EV Use this button to designate that the selected item in the Device Preference Order List should be used for playback in Adobe Audition's Edit View. [EV] appears next to the device's name. Just one playback device can be used in Edit View.

Properties A press of this button brings up the highlighted item's properties in Adobe Audition's Device Properties dialog.

Recording Devices Tab



The Recording Devices tab of the Device Ordering Preference Dialog

Unused Record Devices The items in this list are the record devices on your system not currently utilized by Adobe Audition.

Use>> Press this button to move a selected item from the Unused Record Devices list to the Device Preference Order list.

<<Remove Click on this button to remove a highlighted item from the Device Preference Order list. It will then appear in the Unused Record Devices list.

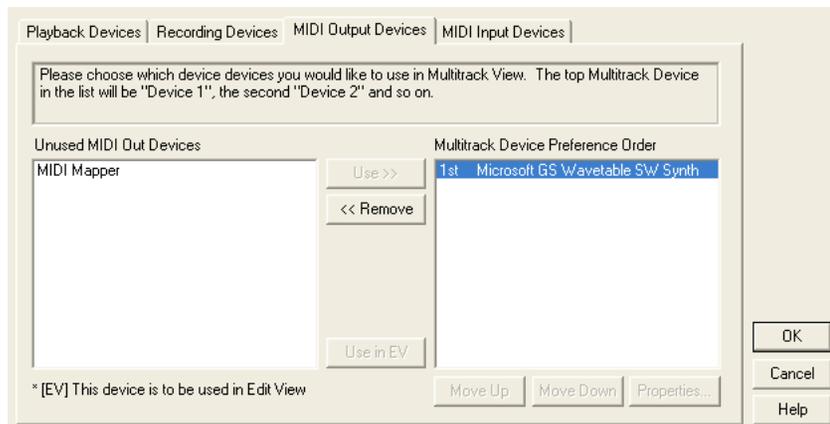
Move Up Pushing this button moves the selected item up in the Device Preference Order list.

Move Down Pressing this button moves the highlighted item down in the Device Preference Order list.

Use in EV Use this button to designate that the selected item in the Device Preference Order List should be used for recording in Adobe Audition's Edit View. [EV] appears next to the device's name. Just one record device can be used in Edit View.

Properties A press of this button brings up the highlighted item's properties in Adobe Audition's Device Properties dialog.

MIDI Output Devices Tab



The MIDI Output Devices tab of the Device Ordering Preference Dialog

Unused MIDI Out Devices The items in this list are the MIDI Out devices on your system not currently utilized by Adobe Audition.

Use>> Press this button to move a selected item from the Unused MIDI Out Devices list to the Device Preference Order list.

<<Remove Click on this button to remove a highlighted item from the Device Preference Order list. It will then appear in the Unused MIDI Out Devices list.

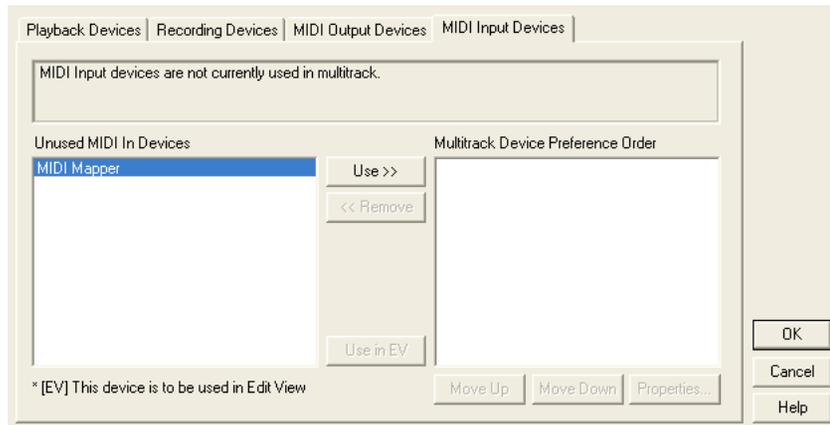
Move Up Pushing this button moves the selected item up in the Device Preference Order list.

Move Down Pressing this button moves the highlighted item down in the Device Preference Order list.

Use in EV Use this button to designate that the selected item in the Device Preference Order List should be used for MIDI playback in Adobe Audition's Edit View. [EV] appears next to the device's name. Just one MIDI Out device can be used in Edit View.

Properties A press of this button brings up the highlighted item's properties in Adobe Audition's Device Properties dialog.

MIDI Input Devices Tab



The MIDI Input Devices tab of the Device Ordering Preference Dialog

Unused Playback Devices The items in this list are the MIDI In devices on your system not currently utilized by Adobe Audition.

Use>> Press this button to move a selected item from the Unused MIDI In Devices list to the Device Preference Order list.

<<Remove Click on this button to remove a highlighted item from the Device Preference Order list. It will then appear in the Unused MIDI In Devices list.

Move Up Pushing this button moves the selected item up in the Device Preference Order list.

Move Down Pressing this button moves the highlighted item down in the Device Preference Order list.

Use in EV Use this button to designate that the selected item in the Device Preference Order List should be used for MIDI input in Adobe Audition's Edit View. [EV] appears next to the device's name. Just one MIDI In Device can be used in Edit View.

Properties A press of this button brings up the highlighted item's properties in Adobe Audition's Device Properties dialog.

Shortcuts (Keyboard & MIDI Triggers)

Shortcuts are used to speed up the editing process by allowing you to execute almost any command in Adobe Audition from your computer and/or MIDI keyboard. For instance, instead of using your mouse to go to the Edit menu and selecting Cut to cut the highlighted portion of a waveform, you can simply press Ctrl+X.

Likewise, you can execute commands from a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command. This type of shortcut is referred to as a MIDI Trigger. An example of this would be assigning the Play command in Adobe Audition to the C4 note on your MIDI keyboard.

Note: *The Shortcuts (Keyboard & MIDI Triggers)... dialog is fully documented in the “Keyboard Shortcuts” chapter.*

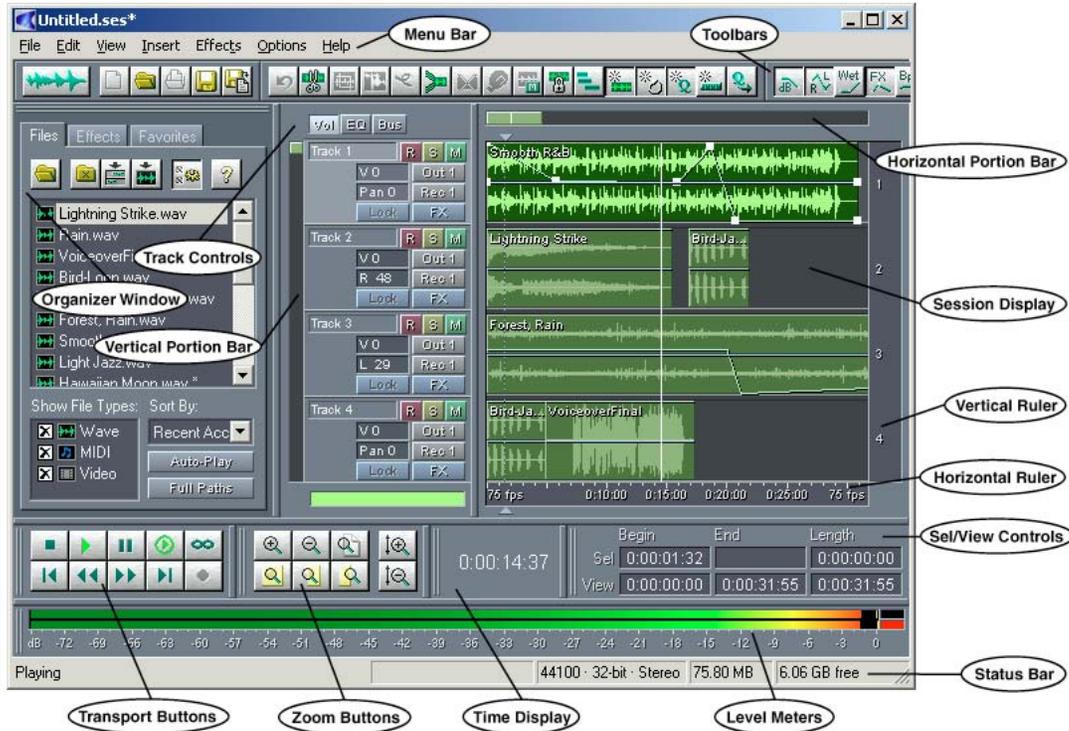
Chapter 13: Navigating the Multitrack View

Multitrack View is one of Adobe Audition's two major workspaces. (Edit View is the other.) Use Multitrack View to record, assemble, mix, and play your audio creations using up to 128 tracks. Add real-time effects such as equalization, reverb, and dynamics processing. Precisely adjust the desired volume, pan, and effects levels for each track.

In this chapter we'll introduce you to Adobe Audition's concepts of images, blocks, and sessions. We'll tell you about the basic mouse movements that are crucial to navigation. Plus, we'll document the interface elements that are specific to the Multitrack View: the Track Controls, the Session Display, the Track Context menu, and the Block Context menu.

***Note:** Interface components that are common to both Edit View and Multitrack View (such as the Level Meters and the Transport Controls) are documented in the "Looking at the Work Area" chapter, while all of the items referenced on the Multitrack View's drop-down menus are explained in the Edit View menu chapters.*

The following is a typical layout for Adobe Audition's Multitrack View Screen:



Multitrack View

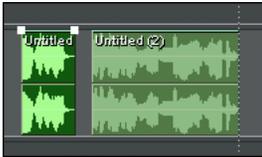
About Images

When you're working in Adobe Audition's Multitrack View and you move a waveform, MIDI file, or video soundtrack from one track and place to another, or create a series of duplicates for looping, you aren't actually moving the original audio file data in any way. In fact, you're manipulating a virtual "image" of that waveform or waveform range in a non-destructive fashion that doesn't affect the original audio file in any way.

Essentially, an image is a set of playback instructions that are tagged to the original audio file(s). For example, creating a series of 16 loops within the Multitrack View window doesn't actually waste disk space by making sixteen copies of the original waveform. Instead, the session file will instruct the program to playback the same portion of the original audio file (the image) 16 times in a repeating fashion. This concept lies at the heart of modern-day hard disk editing.

About Blocks

Blocks are actually images of waveforms, MIDI files, or video soundtracks that are graphically displayed within the Multitrack View that allow you to mix, move, copy, loop, alter volume and pan levels, and add effects in a non-destructive fashion. (Non-destructive in this case means that you can alter image parameters without affecting the original audio data in any way.)



Two waveform blocks on the same track



You can right-click on a block to call up a menu that contains shortcut commands relating to multitrack editing.

About Session Files

Adobe Audition uses session (.ses) files to save all of the information that relates to a current multitrack project (a.k.a. session), such as audio file placement and info about mixing and panning levels, mute and solo status, and application of real-time effects.

Navigating the Session Display

The following list describes the mouse movements that are crucial to navigating around the Multitrack View's Session Display window.

Defining a Range Left-click on one side of the range portion that you want to select and drag the mouse to the left or right. You also select any block that is within the dotted-line rectangle you draw with the mouse.

Editing a Wave Block Double-click on a wave block to send it to Adobe Audition's Edit View for modification.

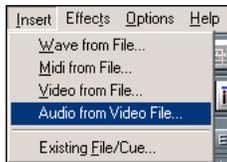
Moving a Block Right-click a block and drag it horizontally to adjust its position within a track. To move the block between tracks, right-click it and drag it vertically.

Displaying Context Menus Two types of context menus can be shown in the Session Display:

- Right-click on a block to view the Block Context menu.
- To see the Track Context menu, right-click on a track anywhere there isn't a block.

Inserting Existing Audio Into Your Session

Unless you're recording a multitrack session from scratch, the first thing that you'll probably want to do when starting a session is to begin loading existing audio files into the Multitrack View's session window. Adobe Audition provides several ways to do this (including dragging files from the Organizer window), but for many people, the most straight-ahead method is through the Insert menu.

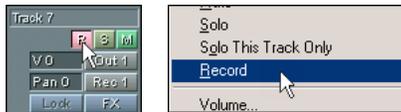


The Insert Menu

The Insert menu provides commands for inserting Wave, MIDI, Video, and Video Soundtracks from any drive attached to your system, as well as the ability to insert files and cues that are already open in Adobe Audition's Edit View. You may also right-click on a track and select from a similar Insert item.

Recording New Audio Into Your Session

Before you record into an Adobe Audition Multitrack session, you'll need to enable the track(s) to which you want to record. To enable a track for recording, push the track's "R" button in the Track Controls. Or, you may right-click in the track anywhere there isn't a block and select the Record item from the pop-up menu.



If you have more than one device in your system that's capable of recording (such as another sound card), you can then select the device from which you'll be recording. Press the track's Recording Device button, which by default will say "Rec 1". (The Record Devices dialog may also be displayed by right-clicking in the track anywhere there isn't a block and selecting the Recording Device item from the pop-up menu.)

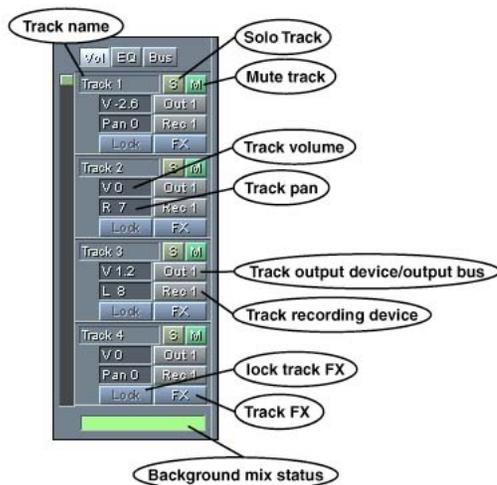
Not only does the Record Devices dialog allow you to select what device will feed the input for the track, it lets you decide if you want the track to record in stereo, left channel only, or right channel only, plus specify if the track should be recorded in 16-bit or 32-bit modes.

Note: You'll also want to make sure that the levels in your sound card's mixer are set correctly. This includes enabling the correct input channel and making sure that the input volume isn't too soft (which can add excessive noise) or too loud (which creates distorted recordings).

After you've enabled the desired track(s) and optionally specified its/their recording properties, press the Record button in Adobe Audition's Transport Controls or use the Ctrl+Spacebar keyboard shortcut to begin recording from the cursor. When recording is complete, press the Stop button in Adobe Audition's Transport Controls or hit the Spacebar to terminate the recording process.

Track Controls

Adobe Audition puts 128 tracks of recording and mixing pleasure at your disposal. Luckily, the controls for each track are identical – so if you learn one, you've learned them all. The Track Controls window appears to the left of the session window, and can be resized to be as wide or as narrow as you wish.



Track Controls set to the Vol Tab

Along with the Vertical Portion Bar and the Mix Gauge, Adobe Audition's Track Control window consists of three main areas for adjusting volume, equalization, and bus properties. Pushing the Vol, EQ, or Bus buttons at the top of the Track Controls window switches between these three areas.



Right-click on a Track Controls module to display Adobe Audition's Track Properties dialog.

Vertical Portion Bar By default, the Vertical Portion Bar runs along the left boundary of the Track Controls window. It's used to scroll through all of Adobe Audition's 128 available tracks, as well as to control the number of tracks Adobe Audition shows you at any one time.

- Left-click and drag the little hand over the Vertical Portion Bar's handle to scroll through Adobe Audition's visible track ranges from tracks 1 through 128. Drag up to see higher track numbers, drag down to see lower track numbers. The number of tracks that will be visible at one time will remain the same, but the track ranges being viewed will be shifted either up or down.

- Left-click and drag the magnifying glass (which appears when your mouse passes over either edge of the Vertical Portion Bar's handle) to adjust the number of viewable on-screen tracks. Either edge of the handle can be stretched, so the longer the handle is, the more tracks are on screen at once. If you make the handle smaller overall, you're zooming in, and if you make the handle larger, you're zooming out. The colored portion of the bar (i.e., the handle) always shows you what portion (hence the name) of the total file you can see.
- Right-click anywhere on the Vertical Portion bar to call up a menu with three zooming options: Zoom In, Zoom Out, and Zoom to Selected Tracks. The menu also has options for moving the Vertical Portion Bar to the left or right of the Session Display, as well as returning it to its default position to the left edge of the Track Controls window.

Track Name Field At the top of each Track Control module, the track's name is displayed in a "sunken" field. By default the track will be named something like "Track 4", but it can be easily renamed by left-clicking on the name of the track and then typing over it.

Record Enable Button Pressing the Record Enable button (it's the button with the "R" on it) will arm the track for recording. When the Record button in the Transport Controls is pushed, this track will begin recording.

Note: The Record Enable button is available in the Vol, EQ, and Bus areas.

Solo Button This is the button with the "S" on it. Pressing it puts the track into Solo mode, meaning that the output of all other tracks will be muted so that you can hear the track by itself. Multiple tracks may be soloed by pushing down their "S" buttons while holding down the Ctrl key.

Note: The Solo button is available in the Vol, EQ, and Bus areas.

Mute Button Clicking on the button with the "M" on it mutes the track, which means that the track will no longer be heard in the mix. Multiple tracks may be muted by pushing their "M" buttons.

Note: The Mute button is available in the Vol, EQ, and Bus areas.

Track Volume Field This field – which has a "V" followed by a number – displays the volume of the track in dB. By default, it's set to 0dB, but you can adjust the track volume from -120 dB to 32 dB by left-clicking the field and sliding your mouse vertically or horizontally. The dB number changes as you slide.

Note: The Track Volume Field only appears in the Vol area.

Output Button Pressing the Output Button (which is labeled "Out 1" by default) brings up Adobe Audition's Playback Devices dialog, which allows you to specify the track's output device or bus. The text on the button changes to reflect the output device or bus chosen.

Note: The Output Button is found in both the Vol and Bus areas.

Track Pan Field This field – which by default has "Pan 0" in it – displays the pan position of the track in dB. You can pan the track to the left or right by up to 100%. Just left-click the field and slide your mouse vertically or horizontally. An "L" in the field designates a left pan, an "R" represents a pan to the right, and the number after the letter designates the percentage of the pan.

Note: The Track Pan Field only appears in the Vol area.

Input Button Pressing the Input Button (which is labeled “Rec 1” by default) brings up Adobe Audition’s Record Devices dialog, which allows you to specify the device from which you are recording. The text on the button changes to reflect the input device chosen.

Note: The Input Button is found in both the Vol and Bus areas.

Lock Button When real-time effects are applied to the track using the FX button, the Lock button is enabled. Pressing it tells Adobe Audition to lock in the effect(s) and disables the FX button. Release the button to unlock the effect(s) and re-enable the FX button.

Note: The Lock Button is found in both the Vol and Bus areas.

Note: When a track is locked, what actually happens is that the effect is mixed into the background mix, thus removing the real-time effect from the CPU load. Keep in mind that when a track is locked, you can’t do anything to the audio in it, such as moving a block or changing effects parameters.

Note: While it takes time to process a track when you lock it, unlocking it is instant. Try locking down tracks that you are not working with at the time to save some CPU power for your other work.

FX Button Click on the FX button to bring up the track’s Effects Rack. The Effects Rack is used to apply any combination of real-time, off-line, and multitrack effects to the track

After effects have been applied to the track, the button is renamed to whatever you’ve typed as a “friendly name” for the effects rack. If you haven’t given a friendly name to the effects rack, a number is added on to the button’s label. For instance, “FX” will become “FX 2” when effects have been applied to Track 2. Press the Lock Button to the left to lock in the effect(s), which disables the FX button.



Right click on the FX button for quick access to the track’s effects rack, settings, and mixer, or to bypass the effects on that track. When track effects are bypassed, they take no CPU time.

Note: The FX Button is found in the Vol, EQ, and Bus areas.

Track Equalization Fields The Low, Mid, and Hi Track EQ fields show the current low frequency, mid frequency, and high frequency equalization amplitude for the track. By default, they’re all set at 0. However, you can easily change a field’s value from -32 to 32 by left-clicking on it and sliding your mouse horizontally or vertically. Or, right-click on an EQ field to bring up Adobe Audition’s Track Equalizers dialog, which gives you more precise control over the track’s EQ.



Track Controls set to the EQ Tab

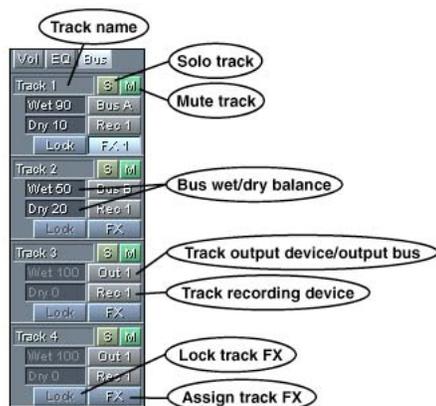
Note: The Track Equalization Fields normally show up only in the EQ area, but will appear under the Vol tab if your Track Controls area is sized large enough.

EQ / A and EQ / B Buttons These buttons allow you to quickly switch between two different banks of equalization parameters. A single-click on the EQ / A button changes it to an EQ / B button, thus bank-switching the EQ settings and allowing you to A/B two sets of equalization parameters. By default, the setting for the EQ / B button is “flat”, so you can A/B against “no equalization” at any time by just clicking that button. However, if you make EQ modifications while the EQ / B button is active, those settings are preserved. This allows you to A/B any two settings you want. To force both banks to be equal to the current settings, double-click on the EQ / A or EQ / B button.



Right-click on either button to bring up the EQ Presets dialog, which is fully documented later in this chapter.

Wet/Dry Fields If a track’s output is set to a Bus, the Wet/Dry Fields are enabled and display the bus’ wet and dry percentages. Change the field’s percentages by left-clicking on it and sliding your mouse horizontally or vertically.



Track Controls set to the Bus Tab

Note: The Wet/Dry Fields only appear in the Bus area.

Mix Gauge The Mix Gauge (which is located underneath the Track Controls of the bottommost viewable track) indicates how much background processing Adobe Audition has completed. It can be thought of as a fuel tank gauge, except that instead of fuel, it measures how much processing needs to be carried out by Adobe Audition “behind the scenes”. The further that the Mix Gauge’s green bar progresses to the right, the more processing has been done to complete the current mixing tasks. Once the processing has been completed, the bar lights up to a bright shade of green, which means that the entire session has been fully mixed.



The Mix Gauge

Some notes about the background mixing process: Mixing involves combining all of the placed waveforms into two (or more) channels for output. Because you can edit, add, and subtract waveforms in the multitrack environment, Adobe Audition must constantly watch for changes to the multitrack session, such as a moved or deleted waveform, a volume change, or new material recorded into a track. When something is altered, Adobe Audition has to immediately work that change into the mixed output, and it does so through background mixing.

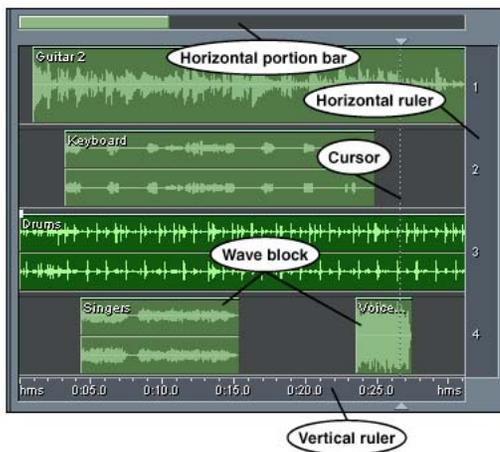
Background mixing occurs behind the scenes much of the time that you’re in the multitrack environment, and is generally very fast. The faster your system – especially your CPU and hard drive(s) – the faster Adobe Audition can mix in the background. The mix progress meter constantly shows how far along Adobe Audition in the process of mixing your session.

However, you need not wait for the meter to reach completion entirely before beginning playback. As stated above, Adobe Audition continuously mixes, and will continue to do so while playing, so you can safely begin playback when the mix progress meter is about halfway full. In general, if the background mix isn't sufficiently completed, you'll hear a break-up or skipping when playing back the mix. If that happens, just wait a few seconds; Adobe Audition will usually "catch up" very quickly, and you can commence playback again.

Of course, Adobe Audition gives you some control over the background mixing process. You can temporarily disable background mixing by selecting Pause Background Mixing from the Options menu. If you're finding that background mixing isn't fast enough, you can increase its priority by going to Options > Settings, clicking on the Multitrack tab, and adjusting the Background Mixing Priority Level.

Session Display

The Session Display is the part of Adobe Audition's Multitrack View that lies to the right of the Track Controls. It contains all track windows, the Horizontal Portion Bar, the Horizontal Ruler, and the Track Display window.



Session Display

Horizontal Portion Bar Use the Horizontal Portion Bar – which, by default, is at the top of the Session Display – to scroll forwards and backwards in time throughout a session. It can also be used to zoom in and out horizontally in the Session Display.

- Left-click and drag the little hand over the Horizontal Portion Bar's "handle" to scroll time-wise through the session.
- Left-click and drag the magnifying glass (which appears when your mouse passes over the edge of the Horizontal Portion Bar's "handle") to change the viewable range of the Session Display. Moving the magnifying glass to the left zooms in, while moving it to the right zooms out.

- Right-click anywhere on the Horizontal Portion Bar to call up a menu with three zooming options: Zoom In, Zoom Out, and Zoom Full. The menu also has options for moving the Horizontal Portion Bar to the bottom of the Session Display, as well as returning it to its default position at the top of the Session Display.

Horizontal Ruler The Horizontal Ruler runs along the bottom of the Session Display and acts as a timeline. It measures time in the same format specified in View > Display Time Format menu (which is hours, minutes, and seconds by default).

- Left-click on the Horizontal Ruler and drag the little hand over the ruler to scroll time-wise through the session.
- Right-click on the Horizontal Ruler and drag to select an area to zoom in on. The zoom in occurs when the right mouse button is released.
- Right-click anywhere on the Horizontal Ruler to call up a menu with several options.
 - The Display Time Format submenu allows you to choose the time format the Horizontal Ruler uses.
 - Snapping allows the edges of any block to “snap” to certain time locations when being dragged. Check the snapping behavior(s) desired.
 - Six Zoom commands are provided: Zoom In, Zoom Out, Zoom Full, Zoom to Selection, Zoom In to Left Edge of Selection, and Zoom In to Right Edge of Selection.

 Several of the zoom commands already have keyboard shortcuts assigned to them, which are shown on the Zooming menu. Regularly using these shortcuts will result in faster editing sessions.

Track Display Ruler The vertical Track Display Ruler – which appears at the right edge of the session window – displays the numbers of all visible tracks. By default Adobe Audition shows you only four tracks, but the range of tracks that are displayed at any one time can easily be changed by using the following mouse commands:

- Left-click and drag the little hand vertically over the Track Display Ruler to scroll through Adobe Audition’s visible track ranges from tracks 1 through 128. Drag up to see higher track numbers, drag down to see lower track numbers. The number of tracks that will be visible at one time will remain the same, but the track ranges being viewed will be shifted either up or down.
- Right-click and drag over the Track Display Ruler to automatically zoom the waveform window to show only the specified tracks.
- Right-click on the Track Display Ruler to call up a menu with three zooming options: Zoom In, Zoom Out, and Zoom to Selected Tracks.

Track Context Menu

Right-clicking on a track in the Multitrack View session window will invoke the Track Context menu.



Track Context Menu

Insert Use the Insert Menu to import waveforms, MIDI files, video files, and audio from video files into Adobe Audition’s Multitrack View. Both active files (those already in Edit View) and inactive files (not already opened) may be easily inserted.

- **Wave from File:** This command activates Adobe Audition’s “Open a Waveform” dialog. It allows you to choose one or more supported waveforms to be placed at the insertion point in the session window. You can open several waveforms at once by selecting them with the Shift (for contiguous selection) or Ctrl (for non-contiguous selection) keys pressed. Multiple waveforms are placed at the insertion point, each on its own track.
- **MIDI from File:** Use this command to place a MIDI file (*.mid or *.rmi) at the session window’s insertion point. Only one MIDI file may be inserted at a time, but you’re welcome to re-run this command as many times as desired to insert multiple MIDI files in a session.
- **Video from File:** This command allows you to insert one video file into Adobe Audition’s session window. A filmstrip graphic that represents the video appears on one track, while the audio from the video is placed in the track below. If you want to actually see the video’s visual data, make sure Adobe Audition’s Video Window is open (View > Show Video Window). Just one video file may be open in Adobe Audition at a time. If you try to insert another one, you’ll be prompted to close the currently open video file.
- **Audio from Video File:** Insert the soundtrack from one or more supported video files to the session window using this command. You’re limited to inserting audio from just one video file at a time, but feel free to re-run this command as often as need to insert addition video soundtracks into your session.
- **Open Waveform List:** Up to 10 open waveforms will be listed here for your convenience. If you have more than 10 waveforms active, filenames of just the last 10 opened are shown.
- **File/Cue List:** This command activates Adobe Audition’s “Insert into Multitrack” window. This window lists all of the files that are currently open in Edit View. If a file has cues in it, a plus sign will appear next to its name to let you expand that file out and see all the cue ranges in it. This allows you to insert either the whole file, or expand it and grab one of the cue ranges directly. To place an item from the list at the session window’s insertion point, just click on it. If you want to insert an item anywhere in the session window, just drag it off the list and drop it on the desired track. No matter which insertion method you choose, the “Insert into Multitrack” window automatically closes.

Mix Down to Track (Bounce) This function performs a mixdown as the Edit > Mix Down command does, and places the result – a mono or stereo file that exactly replicates the sound of your complete session – in the selected track. (This process is commonly known as “bouncing”.) Session elements such as looping, images, effects and envelopes are all reflected in the mixed waveform. You’ll find the Mix Down to Track (Bounce) command handy for creating small sub-mixes in the Multitrack View.

- **All Waves:** Selecting this option will mix all unmuted waveforms in the session and place the resulting stereo waveform in the current track.
- **Selected Waves:** Choose this option to mix all highlighted waveforms in the session and put the resulting stereo waveform in the active track.
- **All Waves (Mono):** Selecting this option will mix all unmuted waveforms in the session to a mono waveform, which will be placed in the current track.
- **Selected Waves (Mono):** Choose this option to mix all highlighted waveforms in the session to a mono waveform, which is put in the active track

***Note:** Adobe Audition treats your mixed track like any other open waveform. You’ll see its name in the Files tab of the Preferences menu as well as in the Insert > Existing File/Cue... pop-up window. It can also be sent to Adobe Audition’s Edit View for modification.*

Mute Track Checkmark this command to mute the track. All blocks within the track are ignored during playback and mixdowns, and are displayed as “grayed-out”. This command does the same thing as pressing the “M” button on the track console.

Solo Track When this command is checkmarked, the track is soloed. A soloed track has the effect of muting out all other tracks, and is used to quickly isolate a track or tracks from the rest of the mix. Unlike the Solo Current Track command, you can run Solo on multiple tracks to place the track into solo mode. This command does the same thing as clicking the “S” button on the track console. (You’ll need to hold down the Ctrl key while pressing the “S” button to solo multiple tracks.)

Solo Current Track Run this command to put just the current track into solo mode. If any other track in the session window is soloed, its solo status will be turned off.

Arm Track for Record Checking this command gets a track ready for recording. This means that the track will be ready to record onto when the Record button in the Transport Controls is pushed. When recording into a track, the input will be fed from the currently selected recording device. This command does the same thing as pressing the “R” button on the track console.

Track Volume This command displays a small floating window which includes a slider for adjusting the track’s overall volume.



Track Volume Window

Dragging the slider's scroll control upward raises the track's volume, while moving the scroll control downward reduces the volume. You may also adjust the track's volume by entering a decibel value in the box above the slider.

Note: Each track has a global volume setting, while each block within the track has an individual volume setting.

Track Pan Use this command to display a small floating window which includes a slider for adjusting the track's overall pan.



Track Pan Dialog

Dragging the slider's scroll control to the right pans the track to the right, while moving the scroll control to the left pans the track to left. You may also adjust the track's pan by entering a value from -100 (hard left) to 100 (hard right) in the box to the right of the slider.

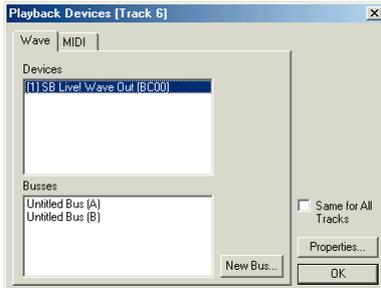
Note: Each track has a global pan setting, while each block within the track has an individual pan setting.

Effects Settings This command does the same thing as pressing the FX button in the Track Console. If real-time effects have been previously added, you'll see a tabbed window with panes for effect settings and mixing. If you haven't applied any real-time effects to the track yet, the Effects Rack dialog is displayed.

Rack Settings Click on this command to display the Effects Rack for the current track.

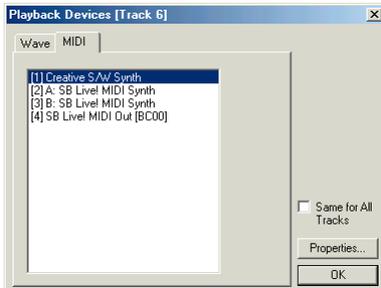
Playback Device This command brings up Adobe Audition's Playback Devices window, which lets you assign the output properties for the selected track.

- Wave Tab:



The Wave Tab of the Playback Devices Dialog

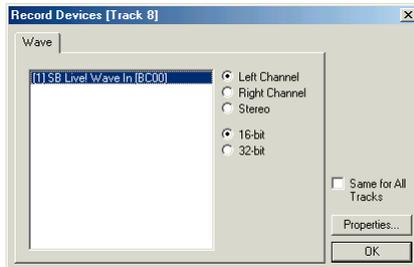
- Devices: All of your system's devices that are capable of playing waveforms are listed here.
- Busses: A list of all Adobe Audition busses you've created is shown here.
- New Bus: Click on this button to add a new bus to the Busses list. A properties dialog for the new bus appears. By default, busses are automatically named with a letter of the alphabet, but you can give a bus a "friendly name" by filling out the appropriate field in the Bus Properties dialog.
- MIDI Tab: A list of all your system's MIDI output devices is shown here.



The MIDI Tab of the Playback Devices Dialog

- Same for All Tracks: Check this box to use the selected Wave device, bus, or MIDI device for all tracks in your session.
- Properties: The button brings up the property dialog for the selected Wave device, bus, or MIDI device.

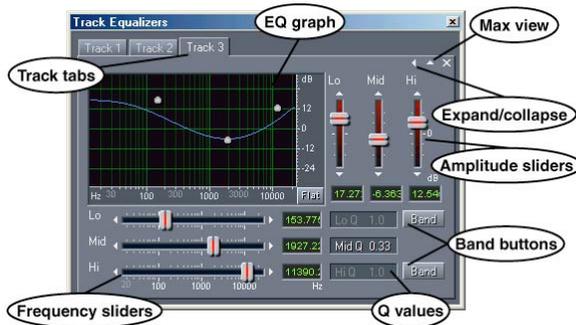
Recording Devices Select this command to display Adobe Audition's Recording Devices window, which lists all available waveform recording devices in your system. Choose the device you wish to use for this track.



Record Devices Dialog

- **Channel Selection:** You can specify if the track will record the left channel, the right channel, or in stereo from the recording device.
- **Bit-depth Selection:** Decide here if you want to record the track in the 16-bit or 32-bit realm.
- **Same for All Tracks:** Check this box if you wish these settings to be the default for all 128 of Adobe Audition's tracks.
- **Properties:** Pressing this button brings up Adobe Audition's Device Ordering Preference screen.

Track Equalizer Activate this item to display Adobe Audition's Track Equalizers window, which gives you more control over a track's equalization than the EQ tab in the Track Controls.



The expanded Track Equalizers window with Track Tabs displayed

The Track EQ window is dockable, which means you can reposition it at one of several places in the Adobe Audition interface. Or, you may detach it and have it float above Adobe Audition's main window.

Here are the major components of the Track EQ window:

- **Track Tabs:** The first time you click on any track in Adobe Audition's session window, a tab is added for that track to the Track EQ window. Click on a tab (which contains the name of the track) to adjust its associated track's settings. The "X" control to the immediate right of the Max View control closes the active tab.

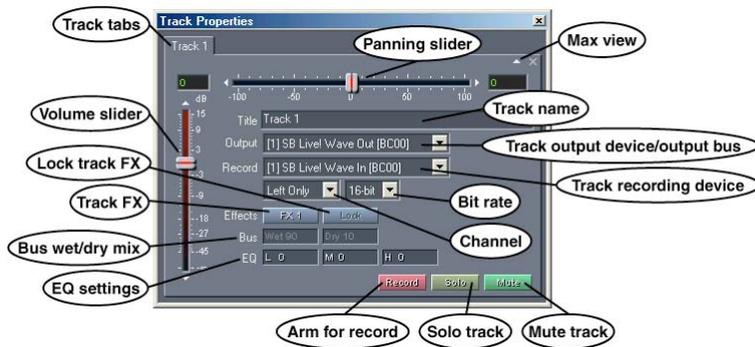
Note: You won't be able to see the Track Tabs if the Max View window display option is selected.

- **EQ Presets:** Press the “P” in the upper right-hand row of icons to access the EQ Presets dialog, which allows you to save favorite equalization settings, as well as delete EQ presets you no longer want. To add the current EQ settings as a preset, click on the “P”, and then click Add New to display the New Preset Name dialog. Enter the name for the new preset, and press OK. Choose an EQ preset by clicking the “P” and clicking on the preset’s name. To delete a preset, press the “P” icon and then click the Delete button. Keep in mind that since clicking on any item chooses it, to delete you must first press Delete, which initiates the delete mode (notice that the button lights up when delete mode is active). Then, you can click on whichever entry you wish to zap.

Note: EQ Presets are also accessible from the EQ section of the Track Controls area by right-clicking on the EQ / A or EQ / B button. The EQ Presets dialog is a “go away” dialog that disappears if you click outside it.

- **Max View:** When you press this control – which looks like a small arrow pointing upward – you switch to Max View. All of the tabs disappear and the controls for the selected track expand to fill the Track Equalizers window. To return to the Track Tabs view, press the Restore control – the small arrow pointing downward.
- **Expand/Collapse:** Use the Expand and Collapse arrows (which point to either the right or the left) to show or hide the equalization frequency and amplitude faders and their associated text entry boxes, as well as the Band buttons.
- **Graph:** The graph shows the track’s current equalization curve. The graph’s x-axis (horizontal) ruler measures frequency, while the y-axis (vertical) ruler represents amplitude. You can drag the three control points (which look like balls) around the graph to adjust the EQ curve. If the Track EQ window is in expanded view, you’ll notice that the two sets of sliders move accordingly when the graph’s control points are manipulated. The left-most control point initially represents the EQ curve’s low frequency, while the middle and right-most control points correspond to the EQ curves’ middle and high frequencies respectively.
- **Q Values:** The low, middle and high frequency Q values of the equalization curve are displayed in these boxes. (Q is the sharpness of the equalization circuit’s peak response.) If a Q box isn’t “grayed out”, you can adjust the Q number by left-clicking on the box and dragging. Dragging to the right increases the Q number, while dragging to the left reduces it.
- **Band Buttons:** By default, only the Mid Q in the Track EQ window is active, but by pushing one or both Band buttons, you can activate the Low Q and/or Hi Q settings. This in effect gives you the control of a parametric equalizer.
- **Frequency Sliders:** Use the vertical Lo, Mid, and Hi Frequency Sliders to adjust the EQ curve’s low, middle, and high frequencies respectively. Drag a slider to the right (or click on the arrow on the right of the slider) to increase the slider’s frequency, while moving it to the left (or clicking on the arrow to the left of the slider) reduces the frequency. You may also specify the slider’s frequency by directly entering the desired Hz value in the text box to the right of the frequency slider.
- **Amplitude Sliders:** The horizontal Lo, Mid, and Hi Amplitude Sliders to are used to increase or attenuate the EQ curve’s low, middle, and high frequencies respectively. Move a slider upwards (or click on the arrow at the top of the slider) to raise the frequency’s amplitude, while bringing it down (or clicking on the arrow to the bottom of the slider) decreases the amplification of the frequency. You may also specify the slider’s amplitude by directly entering the desired dB value in the text box at the bottom of the amplitude slider.

Track Properties Checkmarking this item displays Adobe Audition’s Track Properties dialog, where you can adjust parameters for a selected track’s volume and panning, output and record devices, and more. The Track Properties window is dockable, which means you can reposition it at one of several places in the Adobe Audition interface. Or, you may detach it and have it float above Adobe Audition’s main window.



The Track Properties Dialog in default Tabs view

Here are the major components of the Track Properties window:

- **Track Tabs:** The first time you click on any track in Adobe Audition’s session window, a tab is added for that track to the Track Properties window. Click on a tab (which contains the name of the track) to adjust its associated track’s settings. The “X” control to the immediate right of the Max View control closes the active tab.

Note: You won’t be able to see the Track Tabs if the Max View window display option is selected.

- **Max View:** When you press this control – which looks like a small arrow pointing upward – you switch to Max View. All of the tabs disappear and the controls for the selected track expand to fill the Track Properties window. To return to the Track Tabs view, press the Restore control – the small arrow pointing downward.
- **Volume Slider:** This vertical slider adjusts the track’s volume, measured in dB. Dragging the slider up (or clicking on the arrow at the top of the slider) increases the track’s volume, while moving it downward (or clicking on the arrow underneath the slider) reduces the volume accordingly. You may also specify the track’s volume by directly entering the desired dB value in the text box above the Volume Slider.
- **Panning Slider:** This horizontal slider adjusts the track’s pan, measured in percentage. Moving the slider to the left (or clicking on the arrow at the left of the slider) reduces the volume of the right channel, while dragging it to the right (or clicking on the arrow to the right of the slider) decreases the left channel’s volume. You can also set the track’s panning parameter by directly entering the desired percentage in the text box to the right of the Panning Slider. Negative numbers (such as -100) denote a left pan, while positive numbers (like 100) mean a right pan.
- **Title:** The default title of the selected track (such as “Track 3”) appears in this editable field. You may rename the track by highlighting the text in this field and typing over it. Note that the name of the Track Tab changes accordingly.

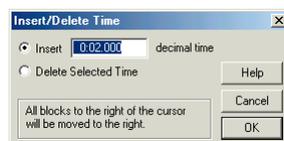
- **Output:** The item shows where the track will be output to, such as a sound card or bus. If you have more than one output device in your system, you can select it from the dropdown list.
- **Record:** The device (such as a sound card) that this track will record through to is shown here. If your system has more than one recording device, you can select it from the dropdown list.
- **Channel:** Use this dropdown list to decide if you want this track to be Left Only, Right Only, or Stereo.
- **Bit Depth:** Here's where you select whether you want the track to be 16 bit or 32 bit. An Adobe Audition session can contain a mix of 16-bit and 32-bit tracks.
- **Effects:** Press the FX button to bring up the Effects Rack for that track.
- **Lock:** If the track has one or more real-time effects applied to it, this button locks the track.
- **Bus:** If you've added this track to a bus, the Wet and Dry percentages are shown here. You can left-click and drag on these boxes to adjust these percentages.
- **EQ:** The Low ("L"), Mid ("M"), and High ("H") equalization values of the track are displayed in these three fields. You can adjust the EQ level of one of these fields by left-clicking on it and dragging. Dragging to the right increases the number, while dragging to the left reduces it. For more equalization control, go to View > Track EQ.
- **Record:** This button arms the track for recording. Press the master record button in the transport controls to start recording.
- **Solo:** This button solos the track so that only it will be heard on playback. All other tracks are muted.

Note: Despite its singular name, more than one track may be soloed. To do this, hold down the Ctrl key while clicking on the solo buttons of all tracks you wish to put into solo mode.

- **Mute:** The Mute button silences the output of this track. More than one track in an Adobe Audition session can be muted at one time.

Select All Blocks in Track As you'd expect, this command selects all blocks in the current track. Each block in the track is highlighted.

Insert/Delete Time Use Insert/Delete Time to introduce an amount of silence into the session (Insert) at the cursor location, or to cut the selected area out of the session (Delete). Insert will shift all material (wave blocks or parts of wave blocks) right of the cursor position to a location after the time amount. Wave blocks will be split if necessary, and the silence inserted. If Delete is chosen, the highlighted area will be removed, with all wave blocks to the right of the selected region shifted to the left.



Insert/Delete Time Dialog

Block Context Menu

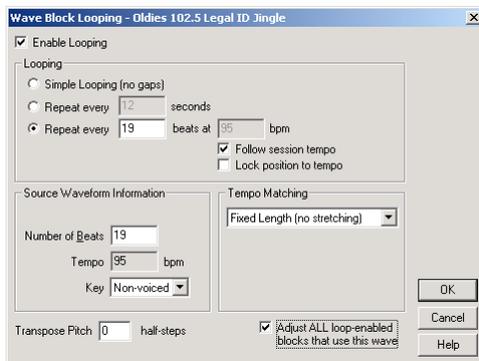
Right-clicking on a block in the Multitrack View session window will invoke the Block Context menu. The items that appear on the Block Context menu will vary depending upon the type of block (waveform, video, or MIDI) selected.



The Block Context Menu for a Wave Block

Edit Waveform This command immediately displays the selected wave block in Adobe Audition's Edit View so that you can edit it.

Loop Properties Select this command (which is only available on the Block Context menu when you right-click on a wave block) to see Adobe Audition's Wave Block Looping dialog.



Wave Block Looping Dialog

- **Enable Looping:** When this box is checked, the wave block can be looped by dragging the right edge of the wave block.
- **Looping:** Use these options to determine the looping method.
 - **Simple Looping (no gaps):** With this option chosen, the wave block will loop continuously, with no spaces between looping instances.
 - **Repeat every X seconds:** If the wave block already has loop info entered, the proper values in the Repeat every X seconds and Repeat every X beats fields will be entered automatically so that the wave block will loop continuously at the proper tempo. If you change the number in the Repeat every X seconds field, Adobe Audition ignores the tempo of the file and stretches the file to force it to finish its loop every X seconds. Normally you'd want to have this option selected and enter the number of beats in the Source Waveform Information area below.
 - **Repeat every X beats:** If the wave block already has loop info entered, the proper values in the Repeat every X seconds and Repeat every X beats fields will be entered automatically so that the wave block will loop continuously at the proper tempo. If you change the number in the Every X beats field, Adobe Audition will stretch the file to force a loop to complete in the specified number beats. However, you'd normally want to have Repeat every X beats selected and enter the number of beats in the Source Waveform Information area below.
 - **Follow session tempo:** When this is checked, the file plays at the session's tempo no matter what its native tempo was. For instance, if you play a 100 bpm loop in a 120 bpm session, the loop will be stretched to match. This disables the "bpm" field above. If you don't have this checked, then the file will play at the bpm specified in the blank above regardless of the session's tempo.
 - **Lock position to tempo:** Checking this box locks the left edge of the block to the bar/beat. So if you change tempo, the block is moved so that it starts at the same beat. Normally if you're stretching to tempo, you'll always want to lock position to tempo. But you may have a one-shot type wave that's not a loop (like a thunder clap or a gong) and want it to start in time with other music that's aligned to the session's tempo.

Source Waveform Information Choose among these options:

- **Number of Beats:** Enter the waveform's number of beats in this field. When you do so, the BPM (Beats Per Minute) is automatically calculated and entered into the Tempo box below.
- **Tempo** is the number of beats per minute in the waveform. When you fill in the Number of Beats field above, the Tempo value is automatically calculated.
- **Key:** Use this dropdown menu to enter the waveform's key. If you specify a key for a waveform, all instances of that waveform will be pitch shifted to match the multitrack session's key as specified in View > Session Information.
- **Tempo Matching** determines how (if at all) the loop will stretch to match the session's tempo. Four options are offered.
 - **Fixed Length:** If this option is chosen, the file won't be stretched to match tempo.
 - **Time-Scale Stretch:** Select this option to stretch the file (just like Adobe Audition's Stretch effect) to match the tempo of the session. Options are provided for Quality (High, Medium, or Low), Frame Size (the number of splices per beat), and the percentage of Frame Overlapping.
 - **Resample** will cause Adobe Audition to resample the loop to match the session's tempo, and as such, the pitch will be affected. High, Medium, and Low Quality options are available.

- The Beat Splice option actually tries to cut up the file at the beats to loop the file that way. This only works on loops that have very sharp (short duration) sounds, like drums. If the waveform already has beat markers, you may elect to use them. Otherwise, the Auto-Find beats option will be selected. If the default values of 10dB and 9 milliseconds aren't satisfactory, you can change them in the fields provided.
- Transpose Pitch: If you want to transpose pitch of the looped block, enter the number of half-steps in this field. Negative and positive numbers are both valid, where positive numbers raise the pitch by half steps and negative numbers lower the pitch.
- Adjust ALL loop-enabled blocks that use this wave: Check this box to globally change the settings for all blocks that reference the same waveform. That is, if you have drums.wav inserted into the Multitrack View four times, and you modify the loop properties on one of its blocks, the other three instances of drum.wav in the session will be adjusted as well.

Allow Multiple Takes With this option enabled, a Take History is preserved for the selected waveform in the event that it's recorded over. Material is not overwritten when recording into an existing waveform block. Instead, a new instance, or "take", of the block is created and recorded into, preserving the previous waveform. With Allow Multiple Takes disabled, recording is done into the same spot each time.

Note: The Punch In function will automatically enable Allow Multiple Takes for the selected Punch area.

Take History The Take History function is available for any track that has Allow Multiple Takes enabled. With multiple takes, material isn't overwritten when recording into an existing waveform block. Instead, a new instance, or "take", of the block is created and recorded into, preserving the previous waveform. Take History is simply a list of each of these instances, allowing you to move between them, delete them, and in the case of a Punch In, merge them back into the surrounding material they once belonged to.

- Take List: This is a list of takes available as alternates for the selected waveform block.
- Merge Current Take: Choose this command to merge a take completed with Punch In back into the surrounding material.
- Delete Current Take: This removes the selected take, deleting it from your hard drive.

Note: Each take exists as an actual waveform on your hard disk, and will take up drive space until deleted. Punch In takes are automatically deleted when Merge Current Take is selected.

Adjust Wave Volume This command displays a small floating window which includes a slider for adjusting the block's overall volume. Dragging the slider's scroll control upward raises the block's volume, while moving the scroll control downward reduces the volume. You may also adjust the block's volume by entering a decibel value in the box above the slider.



Volume Control

Note: Each track has a global volume setting, while each block within the track has an individual volume setting.

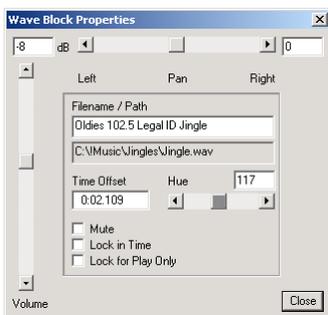
Adjust Wave Pan Use this command to display a small floating window which includes a slider for adjusting the block’s overall pan. Dragging the slider’s scroll control to the right pans the block to the right, while moving the scroll control to the left pans the block to left. You may also adjust the block’s pan by entering a value from –100 (hard left) to 100 (hard right) in the box to the right of the slider.



Pan Control

Note: Each track has a global pan setting, while each block within the track has an individual pan setting.

Wave Block Properties This command displays Adobe Audition’s Wave Block Properties screen, which can be used to adjust several of the wave block’s parameters. This command only appears on the Block Context menu when you’ve right-clicked on a waveform.



Wave Block Properties Dialog

- **Volume:** Use this vertical slider to adjust the wave block's amplitude. You can also directly specify a dB value in the text box above the slider.
- **Pan:** This horizontal slider is used to pan the wave block to the left or right channels of your stereo mix. You may also directly designate the block's pan value by entering it into the box to the right of the slider. A value of -100 is a hard left pan, while 100 means the block is panned all the way to the right channel.
- **Filename/Path:** The name of the wave block (which is initially its filename minus the extension) and the path of the wave block are shown in these two fields. The top field is editable, so you can change how the wave block's name is shown in the session. However, it will actually change the filename when that block is saved back out.
- **Time Offset:** This field shows how far the beginning of the wave block is from the starting point of your session. You can edit this value directly by highlighting the numbers in the field and typing over them, thereby moving the wave block forward or backward through the track.
- **Hue:** Adjust this slider to change the color of the wave block. You can also enter a hue value from 0 to 255 directly in the field over the Hue slider.
- **Mute:** When this box is checked, the block's sound is turned off. Non-muted blocks in that track will play as normal.
- **Lock in Time:** Check this box to prevent the wave block from being moved forward or backwards through the track
- **Lock for Play Only:** This option prevents the wave block from being recorded over.

Transpose (MIDI Block Only) Use this command and its submenu options to transpose the key of a MIDI block up or down by up to 7 steps.

Set Tempo (MIDI Block Only) The command displays the Adjust MIDI Tempo box, where you can make changes to the tempo of the MIDI block using the Tempo field provided. The value used is percentage, with 100 being the default (no change in tempo). Values over 100 speed up MIDI playback, while values of 99 and under slow down the MIDI block's tempo. Check the Use session tempo box to use the tempo value already specified in View > Session information, which disables the Tempo field.

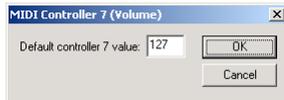


Adjust MIDI Tempo Dialog

Zoom in/out (MIDI Block Only) There are 127 note values that a MIDI note can be assigned to. By default, Adobe Audition figures out which is the highest note in a MIDI block and which is the lowest and zooms in the note display in the block so that the highest note is at the top and the lowest is at the bottom. If you select Zoom Out, all the values are displayed on scale with the 127 note values, which gives you lots of extra space in the block.

Active Track (MIDI Block Only) This command's submenu lists all of the tracks in the MIDI block. Checkmark a track and Adobe Audition will playback only that track when the MIDI block is played. If you want to hear all of the MIDI block's tracks, checkmark the All item.

Set Controller 7 (MIDI Block Only) The Controller 7 value for a MIDI file is the master volume setting. This value gets sent out each time a MIDI file is played and tells the MIDI playback device what volume to play this file at. This option just lets you change the controller value directly in case you have a MIDI file whose Controller 7 value is too low or high.



The MIDI Controller 7 dialog

Save Mixdown to Video As (Video Block Only) When this command is selected, the audio session will be mixed down with the loaded video file to create a new soundtrack for the video. That is, a new video file will be created where all of the audio which is concurrent with the video block is mixed down and provides the soundtrack, and then the video stream from the original video block provides the video stream. The original video stream is actually untouched, so this is the method by which you could save back out a video block with a new soundtrack.

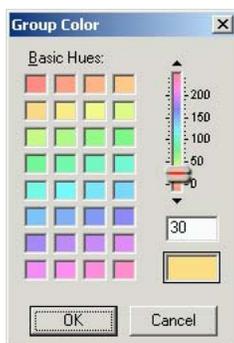
Mix Down to File Running this command (which is only visible when two or more blocks are selected) will mix the selected blocks into a new stereo waveform. This new waveform is automatically placed in the Edit View's Wave Display.

Mix Down to File (Mono) Run this command (which is only show when multiple blocks are selected) to combine the selected blocks into a new mono waveform, which is automatically placed in the Edit View's Wave Display.

Group Blocks This option will group any selected blocks together. Grouped blocks are designated by a different color than ungrouped blocks, as well as a special circular symbol overlaid each block. This feature is fully documented in the "Edit Menu – Multitrack View" chapter.

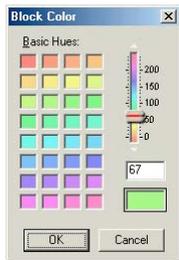
Group Color This command invokes Adobe Audition's Group Color dialog, which allows you to change the color of the current group of blocks. Select from one of 32 preset hues by clicking on it. You may also use the slider to find a color, or you can directly enter a hue value from 0 to 255 in the field provided.

Note: The Block Context menu's Group Color command is only displayed when you have blocks that have been grouped.



Group Color Dialog

Block Color This command invokes Adobe Audition’s Block Color dialog, which allows you to change the color of the current block. Select from one of 32 preset hues by clicking on it. You may also use the slider to find a color, or you can directly enter a hue value from 0 to 255 in the field provided.



Block Color Dialog

Punch In When a particularly difficult passage is being recorded, it’s not uncommon for a note or words to be missed or totally screwed up. No big deal! You can either stop and pick up from before the mishap. Or, you can continue on (as though nothing had happened) and record over the mishap at a later time – this process is known as a “punch in”. Punch In is a recording technique used to replace a section (perhaps a mistake of some kind) in an existing track segment. Replacing this segment usually entails starting and stopping record at exact moments in order to retain the surrounding audio, while material in-between these points is replaced with (hopefully) a better take.

Adobe Audition allows for multiple takes when Punching In; you can continually record over the original material and afterward choose between the best performance, or return to the original data. Punch In will work on one or more waveforms at a time.

To use Punch In:

- Highlight the area of the waveform block (or blocks) you would like to Punch into. For an exact selection, you can switch back to Adobe Audition’s Edit View and use the Find Beats, Zero Crossings, and Snapping functions to make your selection, and then switch back to Multitrack View.
- Choose Punch In from the Edit menu, or from the menu that appears when you right-click on the waveform block. This will enable the selection for record and put the track into record mode.
- You can position the playback cursor sometime before the Punch In location and press the Record button. The session will begin playback, and upon hitting the Punch point, will record into the selected area.
- After recording, you can take another pass at the Punch. When finished, you can monitor the different takes by right-clicking on the waveform block and choosing them from the Take History menu option. Use the desired Merge Take command on the right-click menu when you’ve decided among them.

Note: Whenever one or more Punch Ins has been performed within a wave block, a Take History option will be activated. (Punch Ins and Take History are both available by right-clicking on the wave block.) The Take History option lets you revert the waveform block back to any previously-available take level. This essentially acts as a selective undo function, allowing you to choose any change level between its original un-punched state and the current waveform level.

Note: You can't use Punch In on video or MIDI blocks.

 If possible, choose sections of the waveform where there are gaps in the audio for your Punch points. This will make it easier to achieve a natural sound, as if it were all one take.

Crossfade Crossfades are used to create smooth transitions from the end of one audio segment to the beginning of another. They do so by creating a fade out and a fade in over the transition region. Waveforms to crossfade between should be positioned in different tracks (or can be in the same track, in a back-to-back manner), adjacent if possible.

To crossfade, make a selection with your mouse and include in it the beginning and ending sections of any waveforms you wish to include in the crossfade. The highlight should extend a bit beyond the end of the waveform(s) to fade from, and should start a bit before the beginning of the waveform(s) to fade to. Or, if Edit > Snapping > Snap to Blocks is enabled, the highlight will easily adjust itself to the beginnings and endings of the waveforms. (You can also adjust the highlight by holding the Shift key down when doing a left-click near the edge of the selection.)

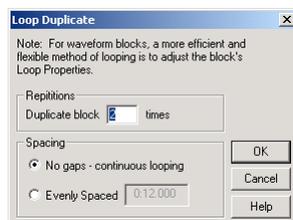
After establishing a highlighted area, you need to select the waveforms to include in the crossfade by holding down the Ctrl key and left-clicking on the desired wave blocks. Next, choose Edit > Crossfade (or Crossfade from the right-click menu) and the type of crossfade you want to do to crossfade between the selected waveforms.

Three main types of crossfades are available. Choose Linear for an even grade, and Sinusoidal for a curved, sine-like slope to the fade. Logarithmic will fade the amplitude of a signal at a constant rate, producing a steeper slope at one end of the fade, depending on whether you are fading in or out.

Note: The Crossfade toolbar button uses Linear fades by default. Choose Crossfade from the Edit or right-click menu to specify a different crossfade type.

 Modify the fade curves created with Crossfade by enabling the Show Volume Envelopes and Enable Envelope Editing options in the View menu.

Loop Duplicate Loop Duplicate will create a specified number of consecutive Images of the selected waveform block creating, in effect, a loop of the waveform.



Loop Duplicate Dialog

- Repetitions: In this field, enter the number of times you'd like to create an Image of the waveform block.
- Spacing: Two options are provided here. Decide if you want no gaps between each duplicated waveform block, or if you wish them to be evenly spaced by a certain amount, which can be specified in the field provided.

To use the Loop Duplicate function to create interesting time delay effects:

- 1 Start by placing the source waveform into the Multitrack View.
- 2 Once entered, hold down the Shift key and right-click on the waveform and drag it to make an Image. The Image should be placed onto a new track (such as below the existing track).
- 3 Next, shift the Image in time, to the right of the original; small amounts to create phase shifts (small time offset), larger amounts to create delays (15-50 ms offsets) or echoes (50 ms or greater offsets).
- 4 Repeat this any number of times, shifting each Image slightly more to the right. You can even create in tempo delays by changing the Display Time Format to Bars and Beats and use the Snapping function to Snap To Ruler locations, placing the delayed Images at every beat, for example.

Try reducing the volume levels slightly for each Image, to soften the echoes (also, the combined waveforms will increase the track's overall level in the mix and could cause clipping). You can also create both Pan and Volume envelopes for each Image for truly dynamic delay effects that change over time.

Convert to Unique Copy This command creates a copy of the selected waveform block. Copying a waveform block in the multitrack environment (as opposed to creating an Image) will create a separate audio file on disk, identical to the original in size and disk space consumption. Creating a copy of a waveform block has the advantage of allowing for separate editing of each occurrence of the waveform (you can add reverb to the one placed in your intro, while the same sound placed in your verse can be dry).

Images have the advantage of not consuming any disk space, and depending on the situation, it may be an advantage to be able to affect all occurrences of a waveform block by editing the original once. (With Images, you can add a Flanger effect to one instance, and all 30 instances, for example would be flanged.)



To quickly add an effect to a waveform block within the Multitrack View, double-click on it (which will open the waveform in the Edit View window). From there, you can use File > Save As... to create a copy of the file that can be transformed using any Effect function. The newly processed file can then be placed into the multitrack environment.

To add multiple effects to an existing waveform non-destructively in Multitrack View:

- 1 Create a copy of the waveform using Edit > Convert to Unique Copy, or by saving the file under a new name using Save As in the Edit View.
- 2 Once a copy of the original file has been made, add any processing or effect you like to it. For effects that have wet/dry balancing, set them to 100%, or close to 100% wet, so the result is primarily effect. (The original waveform that you copied will provide the dry element.)
- 3 Insert the processed copy into the multitrack environment, placing it into an adjacent to the original, in a different track. This way it will simultaneously playback both original file and the effected one.
- 4 You can then vary the effected waveform's (and the original's) volume and pan envelopes to bring the wet/dry in and out, or pan right and left.

Repeat this operation for other effects to have separate control over a number of effects (one for Flange, one for Reverb, etc.).

Mute Block This command mutes or unmutes the selected blocks. This command is different than Track Muting (which occurs when you press the Mute button in the Track Console area), which disables all waveforms within a track. Muted blocks are ignored during playback and mixdowns, and are displayed as “grayed-out”.

Lock in Time Lock In Time “locks” selected waveforms to their current time location in the multitrack session. Locking a waveform is useful when you’ve decided upon its position in time, and don’t want to accidentally move the waveform when right-clicking on it. Locked waveforms are identified by an overlaid graphic of a circled padlock. While a waveform which has been locked can’t be moved in time (left or right), it may be vertically repositioned, allowing you to move it between tracks. To unlock a waveform, simply select it and choose Lock in Time to free it for horizontal movement.

Lock for Play Only This command disables the ability of a waveform block to be recorded into. Even though the track that the playback-locked block resides in may be record-enabled (it has the red “R” button on the Track Console depressed), the marked waveform will not be recorded over. This is signified by the waveform block not turning red when the track is record-enabled.

Split The Split command functions similarly to a traditional tape splice; it “cuts” the waveform, segmenting it into parts. The cut is performed at the current cursor point, or selection (in this case the selection will be spliced out). Once a block has been split, each segment can be moved, deleted, or slid in time independently of the other segments. Splitting a waveform is non-destructive, it does not alter the waveform on disk.

Merge/Rejoin Split Merge/Rejoin Split recombines a split segment of a waveform block with the original surrounding material, provided that the split segment sits at the same relative location as where it was split from, and in the same track as the original segment(s). The boundaries of the segments are joined together using a 30 millisecond crossfade.



The Merge/Rejoin Split command can also be used to rejoin a take created using Punch In to the rest of the waveform.

Note: *This merge operation is destructive in that it does alter your sound file (when rejoining a Punch In segment, the file is rewritten with the new material, for example).*

Align Left The Align Left command (which is only accessible if two or more blocks are highlighted) to make the left-hand edges of all highlighted blocks the same as the left-hand edge of the last highlighted block.

Align Right Click the Align Right command (which is only visible when two or more blocks are highlighted) to make the right-hand edges of all highlighted blocks the same as the right-hand edge of the last block you highlighted.

Adjust Boundaries This function will adjust the boundaries of the selected waveform block either inwardly or outwardly. After highlighting to define an area to adjust to, any material in the waveform block that lies within this highlight will be retained or exposed, removing that which is outside of the highlight (or including material inside the selection that is not currently revealed).

For example, with a 5 second waveform block, highlighting the middle second and choosing Adjust Boundaries will trim off the surrounding material (the first two seconds and the last two seconds), leaving just the middle second. If you decide you’ve trimmed off too much, you can highlight the middle second plus a little to the right, for instance, and again choose Adjust Boundaries to bring back that little bit to the right, thereby lengthening the waveform block.

Note: To fine tune your adjustments in the preceding example, you could also have double-clicked on the trimmed block to bring up the Edit View, double-clicked to highlight the displayed area, then zoomed out to adjust the highlight boundaries to your satisfaction. When finished, switch back to Multitrack View (the highlight will be retained) and choose Adjust Boundaries to have the waveform block readjust to match your highlight.

Note: Unlike the Trim command in Adobe Audition's Edit View, using Adjust Boundaries to trim doesn't remove any data from the actual waveform. Trimming in Multitrack View simply tells Adobe Audition to "disregard this material" when playing back. You may return any trimmed waveform to its original state at any time by selecting it and choosing Edit > Full.

Trim This Trim command works in the same fashion as Trim in the Edit View. All material will be removed from the selected waveform block except the portion within the highlight. Use Trim to remove unwanted material around a selection.

You may also trim an area using the Adjust Boundaries command. Trim differs from Adjust Boundaries in that Trim won't reveal any material within a highlighted region. Trim will only remove that which is outside of the highlight for the selected waveform block.

Note: Unlike the Trim command in Adobe Audition's Edit View, this version does not remove any data from the actual waveform. Trimming in Multitrack View simply tells Adobe Audition to "disregard this material" when playing back. You may return any trimmed waveform to its original state at any time by selecting it and choosing Edit > Full.

Cut Cut will remove the highlighted portion of a selected waveform. For example, if track 3 contains a waveform that is 5 seconds in length, selecting it, highlighting the middle 1 second, and choosing Cut will remove the middle 1 second, leaving the 2 second portions on both sides. If you want to remove unwanted material around a highlighted region, use Edit > Trim instead.

Note: Unlike Cut in Adobe Audition's Edit View, this Cut command does not remove any data from the actual waveform, and places nothing on the clipboard. Using Cut in Multitrack View simply tells Adobe Audition to "disregard this material" when playing back. You may return any Cut waveform to its original state at any time by selecting it and choosing Edit > Full.

Full This option returns any selected waveform block that has been modified with the Adjust Boundaries, Trim, or Cut commands to its original, full state. Any material removed using Adjust Boundaries, Trim, or Cut (while in Multitrack View) will be replaced.

Note: Selecting Full for a waveform block when its segments have been moved will return any missing material at that location, so this may result in having more than one instance of the same waveform block portion throughout a session.

Remove Block This command removes the selected (highlighted) block from a multitrack session.

Removed waveforms will still be available in Adobe Audition's Edit View. They're also available to the session via Insert > File/Cue List until the audio files are closed from the File menu.

Destroy Block (remove & close) This will remove the selected waveform from the multitrack session, and close the waveform.



Use the Shift+Backspace hotkey to quickly destroy a block. To destroy the waveform without being prompted to save, use Ctrl+Shift+Backspace.

Envelopes Use this command's submenu options to clear the Volume, Pan, or FX Mix envelopes from the highlighted portion of a selected block, as well as uses spline (curved) envelopes instead of Adobe Audition's default envelope style (which has straight lines between the envelope's control points).

About Busses

Adobe Audition gives you the ability to organize multiple tracks into busses, which is especially useful for adding one or more real-time effects to and/or adjusting the gain for a common group of tracks. The dictionary states that, electronically speaking, a bus is "an electrical conductor that makes a common connection between several circuits". In the case of Adobe Audition's Multitrack View, a bus lets you make a common connection between several tracks. Sometimes it's more convenient to group two or more tracks together into a bus in order to apply real-time effects or give the tracks the same volume, pan, and equalization properties.

Let's say you're recording a song that has four tracks of background vocals. You want to apply a bit of compression to those tracks to give them a more punchy sound, plus add some reverb. Instead of processing each track separately (which might have a significant drain on your computer's processing speed), you can add the four tracks to their own bus and then apply the reverb to the bus. Up to 26 busses may be created.

How to assign tracks to a bus

Adobe Audition provides several ways of assigning tracks to busses, but here's the easiest:

- 1 Press the Output button in the Track Controls. (The button says "Out 1" by default.) This will display the Playback Devices dialog for the track.
- 2 In the bottom part of the Wave tab of the Playback Devices dialog you'll see a list of existing Busses. If it's the first time you've used busses in Adobe Audition, just one bus – "Untitled Bus (A)" – will be listed. (You can press New Bus... to add another bus to the list – up to 26 if you want.)
- 3 Click on the name of the bus you wish to add the track to.
- 4 Press the Properties... button to display the Bus Properties dialog.
- 5 Use the Bus Properties dialog to give the bus a friendly name and change its output device if desired, as well as assign any Real-Time Effects to the bus. Press OK when you're done.
- 6 Press OK to close the Playback Devices dialog. (Notice how the track's Output button now shows the letter of the bus.)

About Real-Time Effects

Adobe Audition offers three types of effects: Real-Time, Off-Line, and Multitrack. Real-Time Effects have parameters that may be adjusted while listening to the results as part of the mix in the Multitrack View. Since they're non-destructive, they don't actually change the physical sound files in the tracks they're applied to. In addition, Real-Time Effects may be used like Off-Line Effects in Adobe Audition's Edit View.

Off-Line Effects can be used on waveforms that are part of a Multitrack Session, but not in real-time. You'll need to switch to Edit View before you can apply them. Off-Line Effects are destructive, which means they always physically change the waveform they're applied to. (Don't worry – if Undo is active, you can easily reverse the “destruction”.)

Multitrack Effects may only be used in the Multitrack View.

Real-Time Effects include:

- Amplitude: Channel Mixer, Dynamics Processing, Hard Limiting, Pan/Expand, and Stereo Field Rotate
- Delay Effects: Chorus, Delay, Dynamic Delay, Echo, Echo Chamber, Flanger, Full Reverb, Multitap Delay, Quick Verb, Reverb, and Sweeper Phaser
- Filters: Dynamic EQ, Graphic Equalizer, Graphic Phase Shifter, Notch Filter, and Parametric Equalizer
- Special: Convolution and Distortion
- Time/Pitch: Stretch

Off-Line Effects include:

- Amplitude: Amplify, Envelope, and Normalize
- Generate: DTMF Signals, Insert Audio From CD, Noise, and Tones
- Filters: FFT Filter, Quick Filter, and Scientific Filters
- Noise Reduction: Clip/Pop Eliminator, Clip Restoration, Hiss Reduction, and Noise Reduction
- Special: Binaural Auto-Panner
- Analyze: Statistics
- Time/Pitch: Doppler Shifter and Pitch Bender

Multitrack effects include:

- Envelope Follower
- Frequency Band Splitter
- Vocoder

Assigning a Real-Time Effect to a track

There are several ways to add a Real-Time Effect to a track, but here are the two most common.

The Organizer Window Way:

- 1 If you don't have the Organizer Window open, press the F9 key to display it.
- 2 Click on the Effects tab of the Organizer Window.
- 3 Choose the desired Real-Time Effect from the list. (Notice that the names of the non-Real-Time Effects are disabled or “grayed out”.)

- 4 Hold down your left-mouse button and drag the effect over to the track.
- 5 When you release your mouse button, the Track Effects dialog will appear, allowing you to adjust the properties for the effect you've just added, as well as the amount of Wet/Dry mixing of the track's effect.

The Track Controls Way:

- 1 Press the FX button for the track you wish to apply a Real-Time Effect. This will cause the Effects Rack dialog to appear.
- 2 Choose the desired effect from a list of installed Real-Time Effects in the left column.
- 3 Press the Add - > button to add the effect into the Current Effects Rack (the left column) for the track.
- 4 Highlight the effect's name in the left column and press the Properties... button to set the effect's parameters.
- 5 Press Apply if you want to add another effect to the track.
- 6 Press OK when you're finished.

About dialog boxes for Real-Time Effects

When you work with Real-Time Effects in Multitrack View, you'll notice that the look of their dialogs is a little different from the way they appear in Edit View. They take on the visual characteristics of the main Adobe Audition interface, as opposed to the more traditional Windows look.

Functionally, the Real-Time effects are the same as their Edit View versions. Almost always they match up control for control with their non-real-time counterparts in Edit view. In fact, most of the time the control layout is even the same. However, there are two notable exceptions:

- The Stretch effect only does pitch shifting in real time as a track effect, not tempo stretching.
- The Dynamic EQ, Dynamic Delay, and Pan/Expand effects have their envelopes directly on the wave blocks instead of having separate controls to draw the envelopes on.

Chapter 14: File Menu – Multitrack View

The File menu in the Multitrack View offers functions relating to opening, saving, and otherwise manipulating session files.

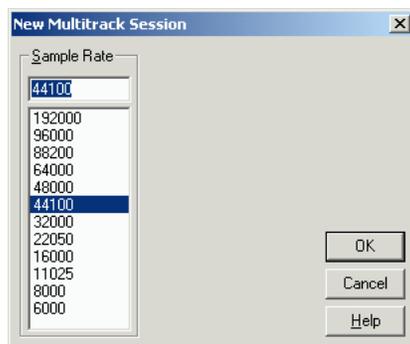
About Sessions

An Adobe Audition “session” uses session (*.ses) files and their specific structure to save all of the information that relates to a current multitrack project. This information includes audio file placement, mix/pan parameters, mute/solo settings, and more. All of your multitrack masterpieces are saved to and recalled from disk as a session file. Only one session can be active at a time.

Note: Keep in mind that your session files save just the information that relates to a current multitrack project. Session files don't contain their associated waveforms. It's important to keep your waveforms intact and within their original folder.

New Session

This command creates a new, empty session, where you record or insert audio into multiple tracks. When creating a new session, you'll be prompted to specify its sample rate. Any audio files to be used in a session must share these properties, or be converted upon inserting them.

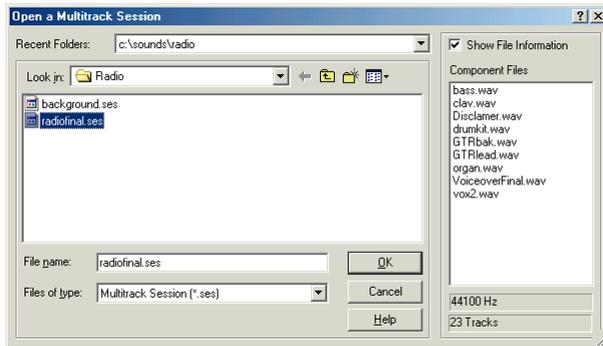


New Multitrack Session dialog

Sample Rate This value determines how many frequencies can be encoded in the audio signal. Higher sampling rates mean a wider bandwidth. Popular sample rates are already listed; just click on one to select it. Or, enter a custom number in the Sample Rate box to specify a different rate from those provided. However, keep in mind that most sound cards only support certain sample rates.

Open Session

This function launches the Open a Multitrack Session dialog, allowing you to load an existing session (.ses) file, along with all audio files used in the session.



Open Session dialog

Here are the components of the Open a Multitrack Session dialog:

Recent Folders The most recently used folders are displayed here in a drop-down menu. This allows quick access to frequently used folders.

Look in Use this drop-down list to navigate through all available drives and folders.

File name Here's where you may specify the name of the file you desire to open, if it resides in the current folder. Entering *.extension (such as *.ses) displays all files with that extension in the open folder.

Files of type Use this control to filter the display of files in the current folder to a specific type of supported session file, such as *.ses. As you'd expect, selecting the All files (*.*) listing displays all files in the current folder.

Show File Information If this item is checked, you'll see all the waveforms referenced by a session in the Component Files list, along with the sampling rate and number of tracks used.

Append to Session

This command opens an existing session and all of its associated audio files, and begins track placement at the end of the currently open session. For example, if the currently open session has material in tracks 1-4, then using Append to Session... to open another four track session would place the new material into tracks 5-8. The selected session must have the same sample rate and bit resolution as the session to be appended to.

The Append a Multitrack Session dialog is virtually identical to the Open a Multitrack Session dialog.

Note: You should note that the appended tracks are loaded into the session at the beginning of the project's timeline (i.e., all of the tracks will load in at the project from its beginning point.) Once loaded, the appended waveforms can easily be moved into place by actively selecting the desired waveforms and by right-clicking on any one of them and then dragging them into position.

 If you'd like to add a previously created session to your current session, this can easily be done using the File > Append to Session command. For example, let's say that we'd like to insert an existing multitrack composition that's been saved as a session. Appending the desired session into our current workspace (which will place them at the beginning of the timeline, onto the next available set of open tracks) we can now select and move the appropriate session waveforms into place (often using the Ctrl-Right Click mouse command to select multiple waveforms) – and Voila – you've integrated two sessions into a whole new composition.

Close Session

Use this function to close the current multitrack session. You'll be asked if you would like to save the session if you've made any modifications since the last save.

Close Session and Waveforms

This command closes the current multitrack session, and also closes all waveforms used in that session. You'll be asked if you would like to save the session if modification have been made since the last save, plus you'll be prompted so save any modified (but unsaved) waveforms.

Close Only Non-Session Waveforms

Select this function to close all waveforms that aren't being used in the current session. You'll be prompted so save any modified (but unsaved) waveforms.

Open Waveform

Use this command to load an existing waveform or CD track into an Adobe Audition editing window. The Open a Waveform dialog appears, allowing you to select one or more audio files for listening or modification. It looks and works similar to a standard Windows Explorer Open window (including allowing you to use the Ctrl and Shift keys to select multiple non-contiguous and contiguous files respectively), but adds useful Adobe Audition-specific features.

Note: More information on the Open a Waveform dialog can be found in the "File Menu – Edit View" chapter.

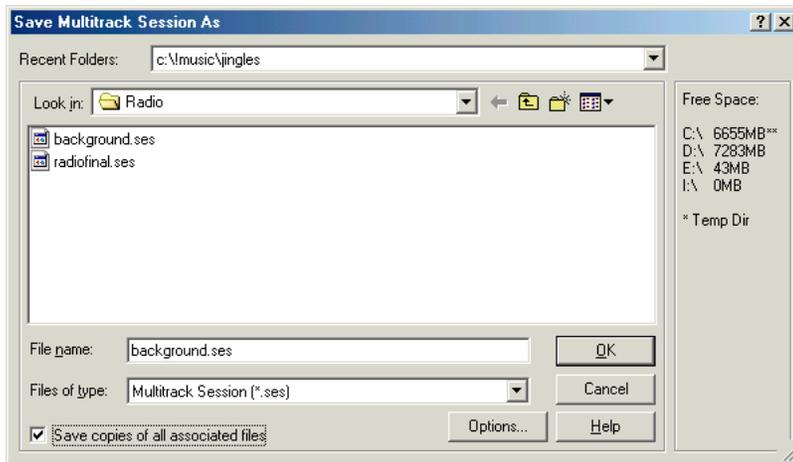
Save Session

Access this command to save the current session back to disk, overwriting the original without confirmation. If the session doesn't yet exist as a file (not previously saved), you'll be presented with the Save Multitrack Session As window first.

Save Session As

The command brings up the Save Multitrack Session As dialog, allowing you to save the current session to a new name or location, along with all waveforms used in the session. Free space on all available drives is displayed in the right section of the window.

You'll be prompted to save any modified (but unsaved) waveforms.



Save Multitrack Session As dialog

Here are components of the Save Multitrack Session As dialog:

Recent Folders The most recently used folders are displayed here in a drop-down menu. This allows quick access to frequently used folders.

Look in Use this drop-down list to navigate through all available drives and folders.

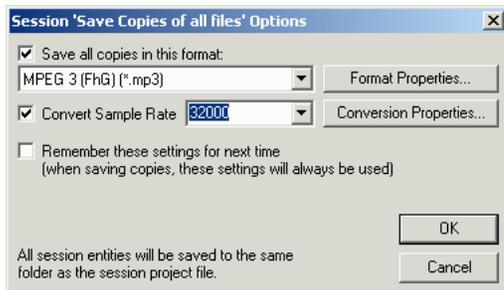
File name Here's where you specify the name of the session. Adobe Audition will automatically add the ".ses" to the filename when you save it.

Files of type Use this control to filter the display of files in the current folder. The default is *.ses files. As you'd expect, selecting the All files (*.*) listing displays all files in the current folder.

Save copies of all associated files If this item is checked, copies of all files used in the session (including waveforms) will be saved in the same folder as the .ses file.

Options When the Save copies of all associated files box is checked, the Options button is enabled. Pressing it brings up the Session Save Copies of all files Options dialog.

Session “Save Copies of all files” Options



Session Save Options dialog

Save all copies in this format Check this box to activate the drop-down menu below, which lets you choose the format to save the files to. If the chosen format has options, the Format Properties... button will be enabled, allowing you to adjust format parameters.

Convert Sample Rate Checking this box enables the drop-down menu to the right, which lists many commonly-used sample rates. Choose the sample rate you wish to convert your session files to.

It also enables the Conversion Properties... button. Press it to display Adobe Audition’s Waveform Conversion Options dialog, which is essentially identical to the Edit View’s Convert Sample Type control.

Remember these settings for next time Check this box if you wish Adobe Audition to always use the Save all copies in this format and Convert Sample Rate settings each time.

Save Mixdown As

This command brings up the Save 16-bit Mixdown To or Save 32-bit Mixdown To window (depending on if you’re working in the 16-bit or 32-bit realm), which allows you to save the current mixdown to a new name, location, or format. You’ll also use the File > Save Mixdown As... function to save newly-constructed mixdowns to disk. Free space on all available drives is displayed in the right section of the window.



Save Mixdown As dialog

Here are the major components of the Save Mixdown As dialog.

Recent Folders This shows the most recently used folders. Select from the drop-down box to quickly navigate to a listing.

Save in Use this control to navigate through all available drives and folders.

File name Here's where you specify the name of the file.

Save as type Use this control to choose the format you wish to save from the drop-down list of supported file types, such as Windows PCM (*.wav) or MPEG 3 (*.mp3).

Some formats can't be written to using waveforms of certain sample types. In this case, Adobe Audition asks you if you want to convert before saving.

Adobe Audition also warns you if you're trying to save a waveform to a compressed (lossy) format, and suggests that you keep a backup of your original file if you wish to preserve full audio fidelity. If you don't want to see this dialog box again, check the Don't display this message in the future box.

Save extra non-audio information Uncheck this box if you don't want to save header fields such as copyright and author information.

Note: If you plan to burn the waveform to compact disc using another program, unchecking the box is highly recommended. Some CD recording interprets non-audio information incorrectly and will place an unpleasant burst of noise as the beginning of each CD track.

Options Most audio file formats support various options. (See the "Waveform File Formats" chapter for more information.) Press the Options button to bring up a window to select the desired settings.

If the format doesn't have any options, the Options button is disabled ("grayed out").

Save Mixdown To Video As

If you have an .avi video file inserted into the session, you can mix down the session and make that your audio track for the video.

This command brings up the Save 16-bit Mixdown To Video or Save 32-bit Mixdown To Video window (depending on if you're working in the 16-bit or 32-bit realm), which allows you to save the current mixdown and video to a new name, location, or format.

The Save Mixdown To Video window is nearly identical to the Save Mixdown As window previously documented. However, the Options... button brings up the Installed Audio CODEC window, which allows you to select the audio format for the .avi file.

Note: While Adobe Audition can open other types of movie files to get at their soundtracks, the ability to save back the audio track only works with .avi files.

Save All

Run the Save All command to save all open, modified waveforms to disk, as well as your open, modified session.

Default Session

If you'd like to make the current session the template for future sessions, select Set Current Session as Default from the Default Session submenu. (If the current session contains blocks which may not exit in the future, Adobe Audition will ask you if you still want to use the session for the default layout.) The template session will be loaded every time a new session is created. You might want to do this because of specific device assignments or session master volume settings.

Select Clear Default Session from the Default Session submenu to make Adobe Audition "forget" the previously saved session template settings.

Free Up Space in Temp Files

Use this command to bring up the Free Up Space in Temp Files window for managing Adobe Audition's use of the space on your hard drive(s), including the instances of Undo for each waveform.

The window also automatically appears when available hard drive space nears zero kilobytes. If your system has lots of available free hard drive space, Adobe Audition's undo levels are cleared regularly, so you may not see this window unless you manually call it from the File menu.

Note: More information about the Free Up Space in Temp Files dialog can be found in the "File Menu – Edit View" chapter.

MRU List

The Most Recently Used list displays the last nine session files opened in Adobe Audition. You can quickly open any of these files (providing they haven't been deleted or moved) by selecting it from the list with your mouse. Or, type the number that's associated with the desired Session.

Exit

As you'd expect, this command closes Adobe Audition. If the session has been modified, or are any edited audio files that are open that haven't been saved to disk, you'll be asked if you want to save them. All temporary files that Adobe Audition has created are deleted as well.

Chapter 15: View Menu – Multitrack View

The View Menu displays options that relate to change the display or viewable area when in the Multitrack View mode.

Edit Waveform View

This command switches to Adobe Audition's Edit View mode, where individual waveforms may be modified.

 Pressing the leftmost button in the Multitrack View File toolbar takes you to Edit View mode. The face of the View button represents the target View that the system will “jump to” when pressed. For example, when you're working in Multitrack View, this button's appearance changes to let you know that if you push the button, you'll go to the Edit View.

Show Pan Envelopes

Checkmark this option to show the Pan Envelopes for each block in the session display. Pan envelopes appear as a vertical blue line through the center of the block and represent the sound file's left/right stereo placement – the top of the block is hard left, and the bottom is hard right.



A wave block with its Pan Envelope set to pan hard left, then hard right, and then back to center

Note: To change a pan envelope, there must be a checkmark next to Enable Envelope Editing (further down the View menu).

Note: Pan Envelopes are nondestructive, meaning they don't affect the original audio file in any way. Therefore, if you return to the Edit View or go to another application and play a sound file that you've applied an envelope to, you won't hear the envelope panning. The only time this envelope is applied is within Adobe Audition's Multitrack View.

Show Volume Envelopes

Select this option to show the Volume Envelopes for each block in the Session Display. Volume envelopes appear as a green line that initially goes straight across the top of a block. The top of the block represents 100% (normal) application, while the bottom of the block represents full attenuation (silence).



A wave block with its Volume Envelope set to do a slow fade out starting about halfway into the block

Note: To change a volume envelope, there must be a checkmark next to *Enable Envelope Editing* (further down the *View* menu).

Note: *Volume Envelopes* are nondestructive, meaning they don't affect the original audio file in any way. Therefore, if you return to the *Edit View* or go to another application and play a sound file that you've applied an envelope to, you won't hear the envelope volume changes. The only time this envelope is applied is within Adobe Audition's *Multitrack View*.

Note: *Volume envelopes* can also be applied to *MIDI blocks* and edited like they are on audio blocks. However, they are used to change the velocity at which the notes are played back. To help you understand, think of a piano: it plays loudly when you bang on the keys and softly when you lightly touch them. Many *MIDI synthesizers* are programmed to respond to velocity so that they sound more "expressive" when played, and the most common way is by changing the volume of a sound. However, some synthesizers are programmed so that velocity changes the pitch or the harmonic content of the sound instead; if you find that your *MIDI files* sound a little stranger than normal when using volume envelopes on your *MIDI blocks*, this may be why.

Show Wet/Dry Mix Envelopes

This command enables the display of *Wet/Dry Mix Envelopes*, which will appear on any block that is in a track that has real-time effects. These envelopes control the wet/dry mix of the effects over time. When the envelope is at the top, the signal coming out of that track is 100% wet (all effects, in other words). If the envelope is at the bottom, the track is 100% dry (which means the original track is unaffected by the effects).



A wave block with its *Wet/Dry Mix Envelope* set to slowly transition from *Dry* to *Wet* over the course of the block

Note: To change *Wet/Dry Mix envelopes*, there must be a checkmark next to *Enable Envelope Editing* (further down the *View* menu).

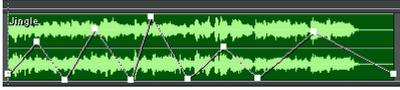


If *View > Status Bar > Data Under Cursor* is checkmarked, you'll be able to see the mix percentage as you drag an envelope's control point around.

Note: *Wet/Dry Mix Envelopes* overlap with the *Volume Envelopes*, so you might have to turn off the *Volume Envelopes* (uncheck *View > Show Volume Envelopes*) to see them.

Show FX Parameter Envelopes

With this command checked, *Effects Envelopes* appear when you apply certain Adobe Audition effects to a track, such as *Pan/Expand*, *Dynamic EQ*, and *Dynamic Delay*. When you add one of these effects, there will be an *Automated* checkbox in its settings dialog. If it's checked, an envelope for that setting will appear. (For instance, *Dynamic Delay* lets you vary *Delay* and *Feedback* over time with these envelopes.)



A wave block with FX Parameter Envelopes (in this case, Dynamic Delay) displayed

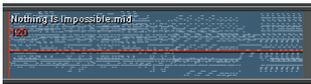
Note: To change FX Parameter envelopes, there must be a checkmark next to *Enable Envelope Editing* (further down the *View* menu).



If *View > Status Bar > Data Under Cursor* is checkmarked, you'll be able to get visual feedback as you manipulate an envelope's control point.

Show Tempo Envelopes

Checkmark this command to enable the display of Tempo Envelopes on MIDI files. Tempo Envelopes – which are view-only – show the MIDI file's tempo map on the MIDI block.



A MIDI block with *Show Tempo Envelopes* turned on

Enable Envelope Editing

With this command checked, Pan, Volume, Wet/Dry, and Effects Envelopes are placed in edit mode, where you may click directly on the envelope line to add control points. Once added, you can simply drag control points up and down to draw the shape of the envelope, or drag a control point off the block to remove it.

Enable Block Edge Dragging

When Block Edge Dragging is enabled, you're able to show or hide more or less of a selected wave block by dragging the block's left or right edge. This non-destructive action has the effect of chopping off (and thereby muting) the first or last part of the block. Note that your mouse cursor changes when it's over a block's edge.



Block Edge Dragging can be used for looping a file. If you turn on looping for a block (right-click and select *Loop Properties*, then check the *Enable Looping* box in the *Wave Block Looping* dialog) and drag the edge, the file will loop as you drag. Even if you then turn Block Edge Dragging off, you can still drag loops with the gripper handle that appears in the bottom-right corner of the block.

Session Properties

Click on this command to show Adobe Audition's Session Properties window, a dockable window which displays useful information about your multitrack session. This window is especially handy to have visible and docked when you're working on loop-based sessions, because loops with defined tempos will match the specified parameters (which is most useful with matching tempo).



Multitrack Session Properties window

Here are the components of the Session Properties window:

Tempo The tempo of the session, measured in beats-per-minute, is shown in this field. Click on the field to make it editable so that you can enter a new figure, if desired.

Beats/bar This field shows the number of beats-per-bar. You can change this figure by clicking in the field and entering a new number.

Key The session's key is shown here. Select a new key by using the drop-down list, which will automatically update the beats/bar field.

Time You'll see the session's time signature in this field. A different time signature may be chosen using the drop-down list.

Advanced Clicking on this button takes you to the Advanced Session Properties dialog, which is documented later in this chapter.

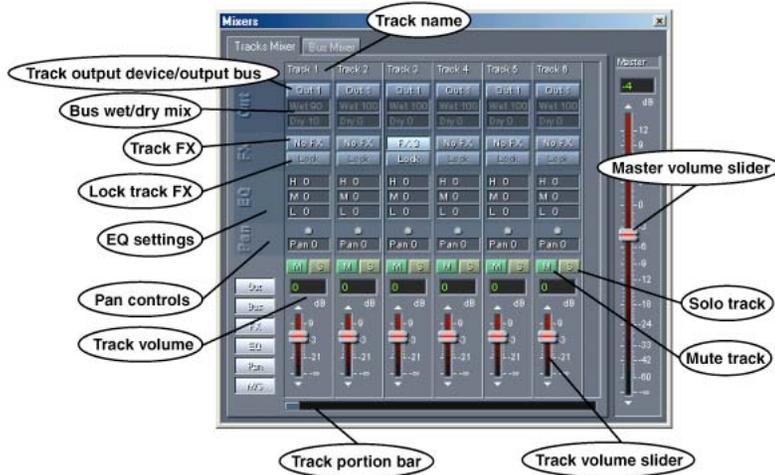
Metronome Press this button to toggle the on/off status of Adobe Audition's built-in metronome.

Show Mixers Window

Checking this item displays Adobe Audition's Mixers window, which consists of two tabbed panes – the Track Mixer and the Bus Mixer – as well as a slider that controls the master volume of your session.

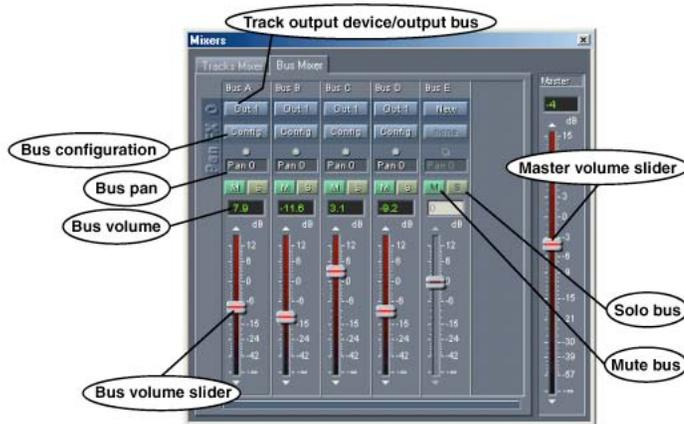
The Mixers window is dockable, which means you can reposition it at one of several places in the Adobe Audition interface. Or, you may detach it and have it float above Adobe Audition's main window.

The Track Mixer's interface mimics a real-world mixing console. It gives you an alternative to Adobe Audition's session window by providing a broader overview of your mixing project, especially if you're working with more than a handful of tracks at once.



Tracks Mixer window

The Bus Mixer pane provides the controls for adjusting the output of up to 26 of the available buses you can have in Adobe Audition, as well as many other bus parameters.



Bus Mixer window

 Even if you don't use the Mixers window, you might want to dock it and resize it so that only the master volume slider is visible. You'll be surprised how often this slider will come in handy.

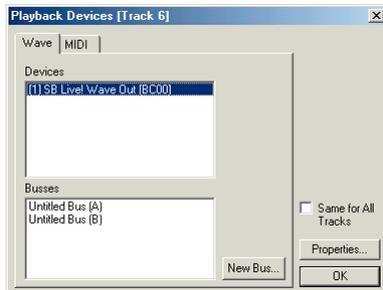
Tracks Mixer Tab

Here are the major components of the Tracks Mixer:

View Display Buttons These six buttons – Out, Bus, FX, EQ, Pan, and M/S – allow you to customize the appearance of the Tracks Mixer. Each button corresponds to a different element of the Tracks Mixer interface.

- Out: Shows/hides the Output device selection buttons
- Bus: Shows/hides the West/Dry Bus fields
- FX: Shows/hides the FX and Lock buttons
- EQ: Shows/hides the three equalization fields (H, M, L)
- Pan: Shows/hides the Pan field and control
- M/S: Shows/hides the M (Mute) and S (Solo) buttons

Out 1 Button Pressing this button brings up Adobe Audition’s Playback Devices window, which lets you assign the output properties for the selected track. The button’s text changes to reflect the output device. For instance, if it were set to device #2, it would say “Device 2”. Or, if it were set to Bus C, it would say “Bus C”.

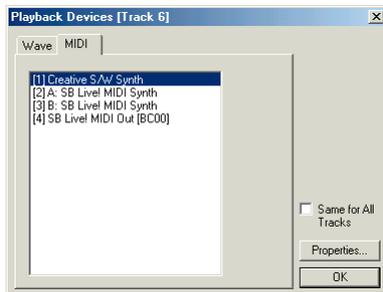


Playback Devices window – Wave tab

Wave Tab

- Devices: All of your system’s devices that are capable of playing waveforms are listed here.
- Busses: A list of all Adobe Audition busses you’ve created is shown here.
- New: Click on this button to add a new bus to the Busses list. A properties dialog for the new bus appears. By default, busses are automatically named with a letter of the alphabet, but you can give a bus a “friendly name” by filling out the appropriate field in the Bus Properties dialog.

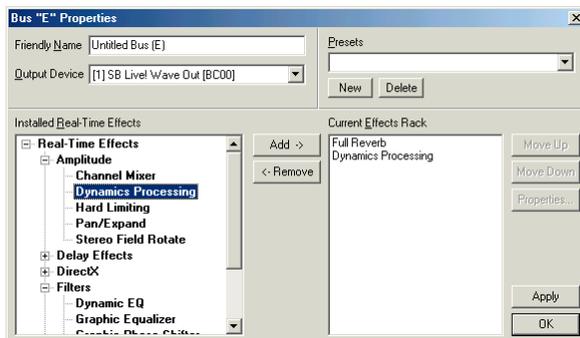
MIDI Tab



Playback Devices windows – MIDI tab

A list of all your system’s MIDI output devices is shown here.

- Same for All Tracks: Check this box to use the selected Wave device, bus, or MIDI device as the default for tracks in your session.
- Properties: The button brings up the property dialog for the selected Wave device, bus, or MIDI device.
 - Bus Properties: Adobe Audition gives you the ability to organize multiple tracks into busses, which is especially useful for adding one or more real-time effects to and/or adjusting the gain for a common group of tracks. For instance, you might have four tracks of background vocals that you want to apply the same reverb effect to. Instead of adding that effect to each track (which might have a significant drain on your computer’s processing speed), you can add the four tracks to their own bus and then apply the reverb to the bus. Up to 26 busses may be created. Use the Bus Properties dialog (which looks very similar to Adobe Audition’s Effects Rack window) to specify and change a bus’s settings.



Bus Properties Dialog

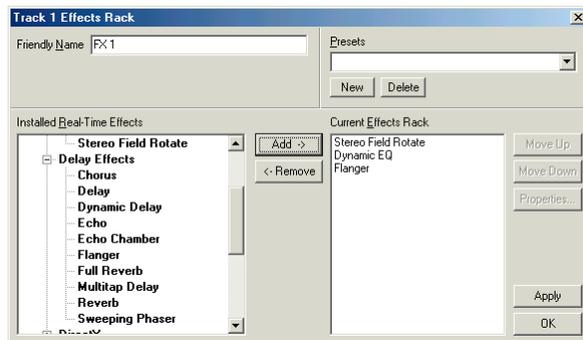
- Friendly Name: By default, Adobe Audition names its busses after a letter of the alphabet – such as Bus (A) – but you can easily rename a bus by highlighting the text in the Friendly Name field and typing over it.

- Output Device: This drop-down list is used to select the desired output device for the bus.
- Installed Real-Time Effects: All installed Adobe Audition and DirectX effects are listed in hierarchical format. To add an effect to a bus, select it and press the Add -> button. The effect will then appear in the Current Effects List pane.
- Current Effects Rack: All effects assigned to the bus are listed here. To remove an effect from the Current Effects Rack, select it and press the <-Remove button. Move an effect up in the Current Effects Rack list (as long as it's not already in the top position) by selecting the effect's name and pressing the Move Up button; the Move Down button moves the select effect down in the list if it's not already at the bottom.
- Presets: The Bus Properties window supports the saving of favorite settings as presets. Press the New button to bring up a dialog for naming a new preset. When OK is clicked, a new item is added to the drop-down Presets menu. To remove the currently visible preset listing, press the Delete button.
- Properties: Pressing this button brings up a tabbed dialog used to adjust the properties of each effect in the Current Effects Rack.

Wet/Dry Fields These fields show the percentage of bus effects that are active on this track. A Wet value of 100 means that 100% of the effect has been added and none of the original audio remains. A Dry Value of 100 means that just the original audio (with no effects) can be heard.

To change the Wet or Dry values, select the desired field, hold down your left-mouse button, and drag. Dragging to the right increases the value, while dragging to the left reduces it.

FX/No FX Buttons If a track doesn't yet have an effect assigned to it (regardless of what effects may be assigned if the track is part of a bus), pressing the No FX button brings up the Effects Rack for that track.



A typical Effects Rack window

Should an effect already be assigned to a track, clicking on the FX button shows the effects properties dialog.

Lock Button If the Lock button is in a disabled (“grayed-out”) state, it’s because there are no effects on that track. If the Lock button is up but not lit, the track isn’t locked. If the Lock button is down and lit, the track is locked.

Equalization Fields The H, M, and L fields show the amount (amplitude) of high, mid, and low equalization frequencies applied to the track. To change the H, M, and L values, select the desired field, hold down your left-mouse button, and drag. Dragging to the right increases the value, while dragging to the left reduces it.

Pan Controls Two types of pan controls for adjusting the balance of each track are included. The control that looks like a small ball is used to quickly pan the track hard left or hard right, or return the track to a centered state. Use your mouse to switch the ball to three possible positions: hard left, zero pan, and hard right. The other pan control shows the current pan value and provides a more precise way of adjusting pan. To change the pan value for the track, select the pan field, hold down your left-mouse button, and drag. Dragging to the right pans the track to the right channel, while dragging to the left pans the track to the left channel.

Mute/Solo Buttons The M (Mute) button turns off the output of the track. You can mute as many tracks as you want by clicking on their M buttons. Press the S (Solo) button to solo the track. To solo multiple tracks, hold down the Ctrl key as you click on their S buttons.

Track Faders The fader is used to adjust the track's relative volume in the mix. Move the fader up (or click on the arrow above the fader) to increase the volume; move it down (or click on the arrow below the fader) for a volume reduction. You may also directly specify the track's volume by entering a dB value in the text box above the fader.

Portion Bar Unless you have a really, really large monitor, you won't be able to see all of Adobe Audition's available 128 tracks on screen at once. Use the portion bar to scroll from tracks 1 to 128 and any point in between.

Bus Mixer Tab

Here are the major components of the Bus Mixer:

Out Pressing this button shows the Bus Properties dialog.

Config Click on this button to display the configuration window for the selected bus. Along with giving you access to the parameters for each effect you've added to the bus, the configuration window's Mixer pane provides sliders for adjusting the volume of all effects in the bus, as well as the bus' desired Dry Out level.

Push the Serial button if you want the effects on the bus to run in order (one effect and then the next).

Click on the Parallel button to feed the track into all the effects separately, and then mix the effects back together, so the output of one effect doesn't change the input of another effect.

Clicking on the Rack Setup button brings up Adobe Audition's Bus Properties dialog.

Pan Controls Two types of pan controls for adjusting the balance of the bus are included.

The control that looks like a small ball is used to quickly pan the bus hard left or hard right, or return the bus to a centered state. Use your mouse to switch the ball to three possible positions: hard left, zero pan, and hard right.

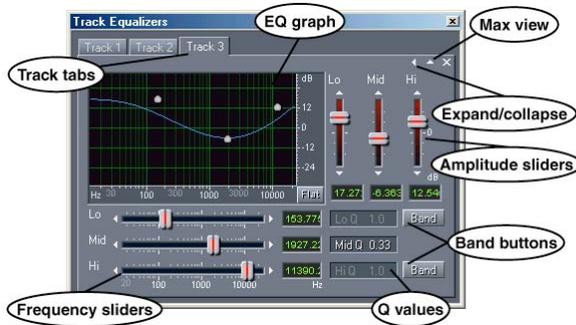
The other pan control shows the current pan value and provides a more precise way of adjusting pan. To change the pan value for the bus, select the pan field, hold down your left-mouse button, and drag. Dragging to the right pans the bus to the right channel, while dragging to the left pans the bus to the left channel.

Mute/Solo Buttons The M (Mute) button turns off the output of the bus. You can mute as many busses as you want by clicking on their M buttons. Press the S (Solo) button to solo the bus. To solo multiple busses, hold down the Ctrl key as you click on their S buttons.

Bus Faders The fader is used to adjust the bus' relative volume in the mix. Move the fader up (or click on the arrow above the fader) to increase the volume; move it down (or click on the arrow below the fader) for a volume reduction. You may also directly specify the bus' volume by entering a dB value in the text box above the fader.

Show Track EQ Window

Activate this item to display Adobe Audition's Track Equalizers window, which gives you more control over a track's equalization than the EQ tab in the Track Controls.



Track Equalizers window

The Track EQ window is dockable, which means you can reposition it at one of several places in the Adobe Audition interface. Or, you may detach it and have it float above Adobe Audition's main window.

Here are the major components of the Track EQ window:

Track Tabs The first time you click on any track in Adobe Audition's session window, a tab is added for that track to the Track EQ window. Click on a tab (which contains the name of the track) to adjust its associated track's settings. The "X" control to the immediate right of the Max View control closes the active tab.

Note: You won't be able to see the Track tabs if the Max View window display option is selected.

Max View When you press this control – which looks like a small arrow pointing upward – you switch to Max View. All of the tabs disappear and the controls for the selected track expand to fill the Track Equalizers window. To return to the Track Tabs view, press the Restore control – the small arrow pointing downward.

Expand/Collapse Use the Expand and Collapse arrows (which point to either the right or the left) to show or hide the equalization frequency and amplitude faders and their associated text entry boxes, as well as the Band buttons.

Graph The graph shows the track's current equalization curve. The graph's x-axis (horizontal) ruler measures frequency, while the y-axis (vertical) ruler represents amplitude.

You can drag the three control points (which look like balls) around the graph to adjust the EQ curve. If the Track EQ window is in expanded view, you'll notice that the two sets of sliders move accordingly when the graph's control points are manipulated.

The left-most control point initially represents the EQ curve's low frequency, while the middle and right-most control points correspond to the EQ curves' middle and high frequencies respectively.

Q Values The low, middle and high frequency Q values of the equalization curve are displayed in these boxes. (Q is the sharpness of the equalization circuit's peak response.) If a Q box isn't "grayed out", you can adjust the Q number by left-clicking on the box and dragging. Dragging to the right increases the Q number, while dragging to the left reduces it.

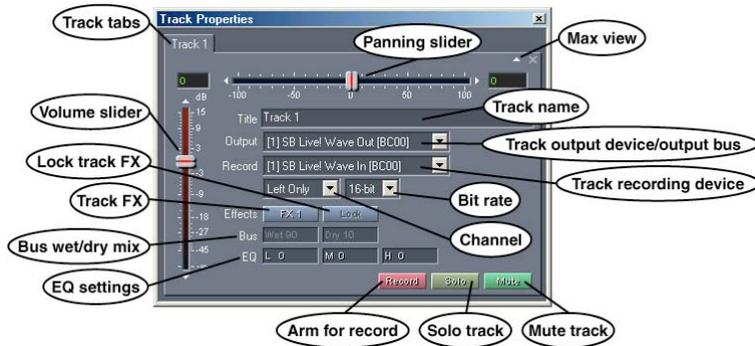
Band Buttons By default, only the Mid Q in the Track EQ window is active, but by pushing one or both Band buttons, you can activate the Low Q and/or Hi Q settings. This in effect gives you the control of a parametric equalizer.

Frequency Sliders Use the vertical Lo, Mid, and Hi Frequency Sliders to adjust the EQ curve's low, middle, and high frequencies respectively. Drag a slider to the right (or click on the arrow on the right of the slider) to increase the slider's frequency, while moving it to the left (or clicking on the arrow to the left of the slider) reduces the frequency. You may also specify the slider's frequency by directly entering the desired Hz value in the text box to the right of the frequency slider.

Amplitude Sliders The horizontal Lo, Mid, and Hi Amplitude Sliders to are used to increase or attenuate the EQ curve's low, middle, and high frequencies respectively. Move a slider upwards (or click on the arrow at the top of the slider) to raise the frequency's amplitude, while bringing it down (or clicking on the arrow to the bottom of the slider) decreases the amplification of the frequency. You may also specify the slider's amplitude by directly entering the desired dB value in the text box at the bottom of the amplitude slider.

Show Track Properties

Checkmarking this item displays Adobe Audition's Track Properties dialog, where you can adjust parameters for a selected track's volume and panning, output and record devices, and more.



Track Properties window

The Track Properties window is dockable, which means you can reposition it at one of several places in the Adobe Audition interface. Or, you may detach it and have it float above Adobe Audition’s main window.

Here are the major components of the Track Properties window:

Track Tabs The first time you click on any track in Adobe Audition’s session window, a tab is added for that track to the Track Properties window. Click on a tab (which contains the name of the track) to adjust its associated track’s settings. The “X” control to the immediate right of the Max View control closes the active tab.

Note: You won’t be able to see the Track tabs if the Max View window display option is selected.

Max View When you press this control – which looks like a small arrow pointing upward – you switch to Max View. All of the tabs disappear and the controls for the selected track expand to fill the Track Properties window. To return to the Track Tabs view, press the Restore control – the small arrow pointing downward.

Volume Slider This vertical slider adjusts the track’s volume, measured in dB. Dragging the slider up (or clicking on the arrow at the top of the slider) increases the track’s volume, while moving it downward (or clicking on the arrow underneath the slider) reduces the volume accordingly. You may also specify the track’s volume by directly entering the desired dB value in the text box above the Volume Slider.

Panning Slider This horizontal slider adjusts the track’s pan, measured in percentage. Moving the slider to the left (or clicking on the arrow at the left of the slider) reduces the volume of the right channel, while dragging it to the right (or clicking on the arrow to the right of the slider) decreases the left channel’s volume.

You can also set the track’s panning parameter by directly entering the desired percentage in the text box to the right of the Panning Slider. Negative numbers (such as -100) denote a left pan, while positive numbers (like 100) mean a right pan.

Title The default title of the selected track (such as “Track 3”) appears in this editable field. You may rename the track by highlighting the text in this field and typing over it. Note that the name of the Track Tab changes accordingly.

Output The item shows where the track will be output to, such as a sound card or bus. If you have more than one output device in your system, you can select it from the drop-down list.

Record The device (such as a sound card) that this track will record through to is shown here. If your system has more than one recording device, you can select it from the drop-down list.

Channel Use this drop-down list to decide if you want this track to be Left Only, Right Only, or Stereo.

Bit Rate Here's where you select whether you want the track to be 16-bit or 32-bit. An Adobe Audition session can contain a mix of 16-bit and 32-bit tracks.

Effects Press the FX button to bring up the Effects Rack for that track.

Lock If the track has one or more real-time effects applied to it, this button locks the track

Bus If you've added this track to a bus, the Wet and Dry percentages are shown here. You can left-click and drag on these boxes to adjust these percentages.

EQ The Low ("L"), Mid ("M"), and High ("H") equalization values of the track are displayed in these three fields. You can adjust the EQ level of one of these fields by left-clicking on it and dragging. Dragging to the right increases the number, while dragging to the left reduces it. For more equalization control, go to View > Track EQ.

Record This button arms the track for recording. Press this master record button in the Transport Buttons to actually start recording.

Solo This button solos the track so that only it will be heard on playback. All other tracks are muted.

Note: Despite its singular name, more than one track may be soloed. To do this, hold down the Ctrl key while clicking on the solo buttons of all tracks you wish to put into solo mode.

Mute The Mute button silences the output of this track. More than one track in an Adobe Audition session can be muted at one time.

Show Organizer Window

This command lets you see Adobe Audition's Organizer window. This handy, tabbed window allows you to easily open and close files, choose effects (including dragging real-time effects directly to tracks), and more.

Note: The Organizer window is fully documented in the "Looking at the Work Area" chapter.

Show Cue List

The command opens the Cue List window, which is a dockable window. A cue list is a list of time offsets, or locations defined within an audio file. A cue can be either a point that specifies a cursor position, or a range that specifies a selection. You may define and save an unlimited number of cues in the Cue List for later recall, or for assembling in Adobe Audition's Play List.

Note: The Cue List is fully documented in the "View Menu – Edit View" chapter.

Show Transport Buttons

Check this item to display Adobe Audition's Transport Buttons, which are used to shuttle the active waveform or session.

Note: More information on the Transport Buttons can be found in the "Looking at the Work Area" chapter.

Show Zoom Buttons

Select this item to display Adobe Audition's Zoom Buttons. The Zoom Buttons window is a dockable window, allowing you to reposition it at one of several places in the Adobe Audition interface.

Note: More information on the Zoom Buttons can be found in the "Looking at the Work Area" chapter.

Show Time Window

Check this item to show Adobe Audition's Time window. This window displays the current position of the playback cursor. The default display value is mm:ss:ddd (minutes:seconds:thousandths of a second), but this can easily be changed by right-clicking on the Time window and choosing another option, such as Compact Disc, Samples, Bars and Beats, or one of several SMPTE values.

Note: More information on the Time window can be found in the "Looking at the Work Area" chapter.

Show Sel/View Controls

When checked, the Sel/View Controls window is displayed. ("Sel" is an abbreviation for Selection.) It shows the beginning and ending points, as well as the total length of both the selection and the section of the session that's currently viewed. Both the selection and display range is shown using the current time display format.

Note: More information on the Sel/View Controls can be found in the "Looking at the Work Area" chapter.

Show Level Meters

Click on this option to display Adobe Audition's Level Meters, which appear near the bottom of both Adobe Audition's Edit View and Multitrack View windows. These meters display both record and playback levels.

Note: More information on the Level Meters can be found in the "Looking at the Work Area" chapter.

Show Load Meter

This option displays Adobe Audition's Load Meter, which runs in a dockable window. The Load Meter shows you how much CPU time is available, which is especially handy if you regularly use real-time effects. When the meter fills up, Adobe Audition has no more available computing power and your system will act erratically.

Show Video Window

If this option is checked, a window used to view video data that's inserted in Adobe Audition's Multitrack View is displayed. It initially appears in the upper left-hand corner of your desktop, and can be resized to fit your frame size. Plus, since the Video window is dockable, you can reposition it at one of several places in the Adobe Audition interface.

Show a Placekeeper

Select Show a Placekeeper to create a blank, dockable window called a "placekeeper". Use it to more accurately define where you want a dockable window to be, especially for windows that are only useful when viewed in a certain aspect ratio.

Note: More information on Placekeepers can be found in the "Looking at the Work Area" chapter.

Display Time Format

The Display Time Format menu lets you choose between several time display readouts, including Decimal, Compact Disc, different versions of SMPTE, Samples, Bars and Beats, and Custom time code frame rates. Adobe Audition will then use the format chosen here when referencing time: the Time window, the x-axis (time) ruler, time status boxes, Status Bar, etc.

Note: More information on the Display Time Format menu can be found in the "View Menu – Edit View" chapter.

Toolbars

Use this menu to specify which, if any, groups of icons you'd like to display on Adobe Audition's Multitrack View toolbar. Checkmark the desired icon group(s) to display. If no items are checked, the toolbar area will disappear until you check at least one icon group.

Options are also provided to limit how many rows of toolbar icons Adobe Audition displays at any one time: one row, two rows, or three rows.

Status Bar

Access the items on the Status Bar submenu to turn on Adobe Audition's status bar – which runs along the bottom of the Adobe Audition window – as well as decide which information fields you wish displayed in the status bar. You can also find this menu by right-clicking on the status bar.

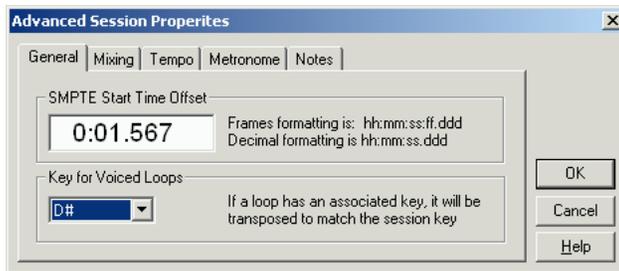
To hide the Status Bar, right-click on it and select the Hide menu option. Or, uncheck Show from the Status Bar submenu item on the View menu.

Note: More information on the Status Bar can be found in the "Looking at the Work Area" chapter.

Advanced Session Properties

This command brings up the Advanced Session Properties window, a tabbed dialog that lets you adjust several parameters of your multitrack session, as well as store additional session-based information within your session file.

General Tab

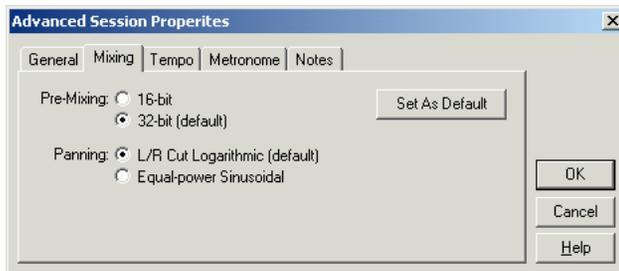


The General tab of the Advanced Session Properties window

Session Start Time Offset Here you can enter a time offset, which is the time location that Adobe Audition will wait for to begin playback. Locations must be entered in the format of hours:minutes:seconds:frames.divisions, or hours:minutes:seconds.division if not being used with SMPTE.

Key for Voiced Loops This specifies the session's key. Any loops that have a key specified will be pitch-shifted to match the session's key.

Mixing Tab



The Mixing tab of the Advanced Session Properties window

Pre-Mixing This is the bit depth used for the background mixing process. Best quality is achieved by choosing the 32-bit setting, especially for single output device situations or if you have a fast hard drive.

However, if you're using multiple sound cards, it may be advantageous to pick 16-bit for pre-mixing, as less data will be transferred across the hard drive(s), which will speed things up.

Note: Even if you do 16-bit pre-mixing, no data is lost, and a final mixdown can still be 32-bit.

Panning This section sets the method used for panning waveforms in a multitrack session.

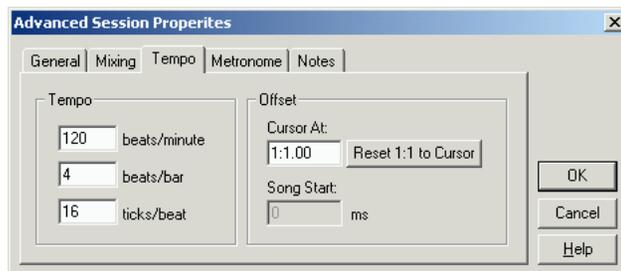
- **L/R Cut Logarithmic:** If this option is chosen, left-panning is achieved by reducing the volume of the right channel, and right-panning is performed by reducing the left channel volume. The channel being panned to doesn't increase in volume as panning gets closer to 100%.
- **Equal-power Sinusoidal:** Choose this option to pan left and right channels with equal power, so a hard pan left will contain the same loudness as both channels together. This results in an increase of 3dB RMS on the channel being panned to when at 100%.

Note: Because panning can actually make one channel louder than the original waveform, audible clipping can occur in 16-bit sessions. To avoid this, work in the 32-bit realm if you're using the Equal-power Sinusoidal panning method.

Set As Default If you've changed the Pre-mixing and/or Panning settings, press this button to have Adobe Audition remember them as the default settings.

Note: Adobe Audition's default settings for Multitrack View are under the Multitrack tab of Options > Settings.

Tempo Tab



The Tempo tab of the Advanced Session Properties window

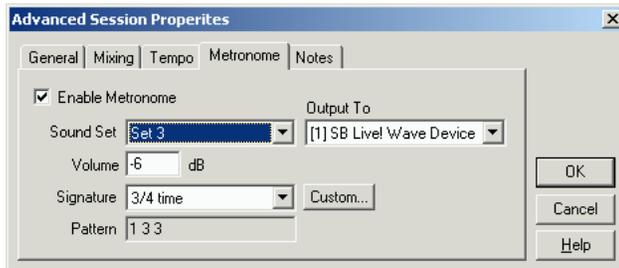
Tempo The tempo of the session is specified here. Any loops that have a tempo specified will be time-stretched to match the session's tempo. The time-stretch method used is determined by the individual loop's settings. If you need more detailed tempo settings, right-click on the horizontal ruler in Adobe Audition's Session Display.

- **Beats/minute:** Enter the number of beats-per-minute here.
- **Beats/bar:** Input the desired number of beats-per-bar in this field.
- **Ticks/beat:** This field is where you can specify the amount of ticks-per-beat.

Offset Choose from the following:

- **Cursor At:** This shows the current position of the cursor in bars-and-beats format. You can also enter a new cursor position here and Adobe Audition will recalibrate the session, including negative time. This is so you can line up your metronome with a song that’s already been created and you’re adding more to. Any point can be at any time, so that you know the metronome beat will be in the right place.
- **Reset 1:1 to Cursor:** Press this button to set the cursor to the first beat of the first measure.
- **Song Start:** The exact position (stated in milliseconds) of the beginning of the song is shown here.

Metronome Tab



The Metronome tab of the Advanced Session Properties window

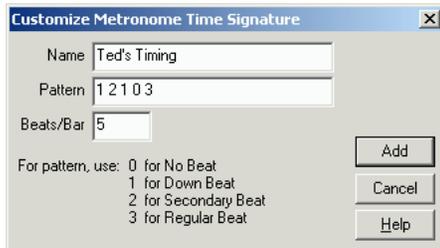
Enable Metronome Keeping perfect time is a cinch with Adobe Audition’s built in Metronome. Check the Enable Metronome box to hear the “ticks” of the metronome whenever you play or record something in Adobe Audition’s Multitrack view. (The metronome sound effects won’t be recorded – unless, of course, you’re using a microphone and your computer’s speakers are on.) Uncheck the box to be Metronome-free

Sound Set Use this drop-down list to select from different types of metronome sounds.

Volume Enter the desired volume of the metronome sounds in this text box. The default is -6dB. A higher number (such as -3dB) makes the metronome sounds louder, while a lower number (like -10dB) decreases the metronome volume.

Signature Select the appropriate time signature using this drop-down list. The current signature pattern is displayed in the non-editable Pattern field. Custom time signatures may be added using the Custom... button.

Custom Press this button to bring up the Customize Metronome Time Signature dialog.

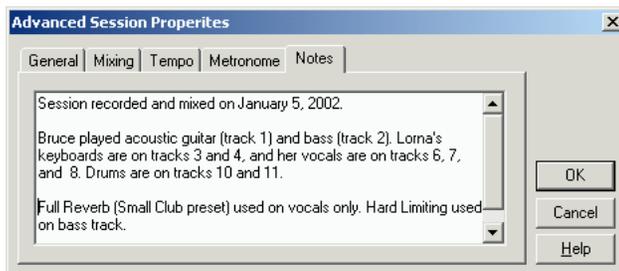


The Custom Metronome Time Signature dialog

- **Name:** Enter the name of the new time signature in this field.
- **Pattern:** Here's where you specify the pattern of the time signature. Use a 0 for no beat, a 1 for a down beat, a 2 for a secondary beat, and 3 for a regular beat. Spaces between the numbers are optional.
- **Beats/Bar:** The number of beats per bar goes in this text box. Press the Add button to place your new custom time signature to the Metronome's Signature list.

Pattern The current metronome pattern is shown in this non-editable text box.

Notes Tab



The Notes tab of the Advanced Session Properties window

You can enter any freeform text about the session here. For instance, the names of the musicians, who played what instrument, and notes about settings.

Chapter 16: Edit Menu – Multitrack View

The Edit Menu contains the options for editing waveform blocks in Adobe Audition's Multitrack View.

T Undo

As in Adobe Audition's Edit View, Undo allows you to back up through previous edits. In the multitrack environment, this includes anything you might do in the way of editing or arranging waveform blocks, such as moving, splicing, and copying. Unlike Undo in the Edit View, there isn't an Enable Undo function governing the ability to undo actions in Multitrack. Edits done in the multitrack environment are non-destructive, and therefore don't create temporary files on disk.

Note: Multitrack Edits done with real-time effects don't go into the Undo list. However, they're totally reversible by simply removing the effect from the track's rack.

Edit Waveform (Wave Block Only)

If a single wave block is highlighted, this command will appear on the Multitrack View's Edit menu. Access the command to send the wave block to Adobe Audition's Edit View for modification.

Loop Properties (Wave Block Only)

If a single wave block is highlighted, this command will appear on the Multitrack View's Edit menu. Access this command to bring up Adobe Audition's Wave Block Looping dialog so that you can adjust the wave block's loop properties.

More information about the Loop Properties dialog can be found in the "Navigating the Multitrack View Screen" chapter.

Allow Multiple Takes (Wave Block Only)

If a single wave block is highlighted, this command will appear on the Multitrack View's Edit menu. When you enable this option by clicking on its menu command (causing a check mark to appear next to its name), a Take History is preserved for the selected waveform in the event that it's recorded over. Material is not overwritten when recording into an existing waveform block. Instead, a new instance, or "take", of the block is created and recorded into, preserving the previous waveform.

With Allow Multiple Takes disabled, recording is done into the same spot each time.

Note: The Punch In function will automatically enable Allow Multiple Takes for the selected Punch area.

Take History (Wave Block Only)

If a single wave block is highlighted, this command will appear on the Multitrack View's Edit menu. The Take History function is available for any track that has the Allow Multiple Takes command enabled. With multiple takes, material is not overwritten when recording into an existing waveform block, instead a new instance, or "take", of the block is created and recorded into, preserving the previous waveform. Take History is simply a list of each of these instances, allowing you to move between them, delete them, and in the case of a Punch In, merge them back into the surrounding material they once belonged to.

Take List This is a list of takes available as alternates for the selected waveform block.

Merge This Take Choose this command to merge a take completed with Punch In back into the surrounding material.

Delete This Take This removes the selected take, deleting it from your hard drive.

Note: Each take exists as an actual waveform on your hard disk, and will take up drive space until deleted. Punch In takes are automatically deleted when Merge Current Take is selected.

Adjust Wave Volume (Wave Block Only)

If a single wave block is highlighted, this command will appear on the Multitrack View's Edit menu. Click it to display a vertical slider used to adjust the wave block's volume, which is measured in dB.

Dragging the slider up (or clicking on the arrow at the top of the slider) increases the wave block's volume, while moving it downward (or clicking on the arrow underneath the slider) reduces the volume accordingly.

You may also specify the wave block's volume by directly entering the desired dB value in the text box above the Volume Slider.

Adjust Wave Pan (Wave Block Only)

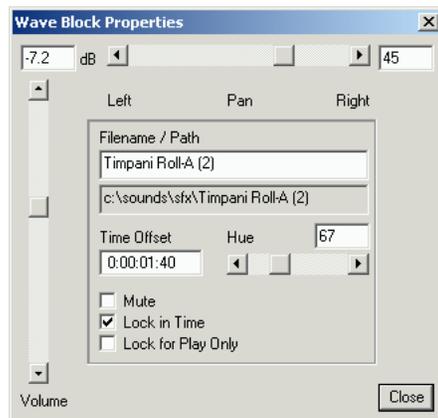
If a single wave block is highlighted, this command will appear on the Multitrack View's Edit menu. Click it to display a horizontal slider used to adjust the wave block's left/right volume balance.

Moving the slider to the left (or clicking on the arrow at the left of the slider) reduces the volume of the right channel, while dragging it to the right (or clicking on the arrow to the right of the slider) decreases the left channel's volume.

You can also set the wave block's panning parameter by directly entering the desired percentage in the text box to the right of the Panning Slider. Negative numbers (such as -100) denote a left pan, while positive numbers (such as 100) mean a right pan.

Wave Block Properties (Wave Block Only)

If a single wave block is highlighted, this command will appear on the Multitrack View's Edit menu. It brings up the Wave Block Properties dialog, which displays information on the selected waveform block. You can use this dialog to change waveform block properties, such as position, panning, volume, and more.



Wave Block Properties dialog

Volume Adjust the Volume slider or enter a decibel value in the box above to set an overall volume level for the waveform block.

Note: This volume setting is independent of the track's overall volume. Track volume is set in the Track Info dialog.

Pan Adjust the Pan slider or enter a pan amount (-100 is hard left, 100 is hard right) in the box to the right to set a pan level for the waveform block.

Note: This pan setting is independent of the track's overall pan. Track pan is set in the Track Info dialog.

Filename/Path The waveform's filename is displayed in an editable text box. You can change this name to whatever you want, which Adobe Audition will use to identify the waveform in Multitrack View. (For instance, "lornascratch-vocal.wav" could be renamed to the friendlier-looking "Lorna's Scratch Vocal".) The waveform's true filename isn't changed, and is shown along with its path in the non-editable space below.

Time Offset This box displays the offset, or beginning time, for the waveform block. You can enter a value directly to move the block to a specific time location. The time format used is that which is currently selected in View > Display Time Format.

Hue This sets the color of the waveform block. Use the slider or enter the hue value directly in the field provided. Any value of 0 to 255 is valid. Roughly, 0 is red, 25 is brown, 42 is yellow, 90 is green, 120 is aqua, 140 is blue, 185 is violet, 205 is purple, and 220 is fuchsia.

Mute Check this box to mute, or disable, the selected waveform block. The wave block will become gray in color, indicating that it won't play until unmuted. Remove this check to unmute.

Note: This Mute setting is independent of the track's overall mute state. Track Mute is set in the Track Info dialog, or by pressing the green "M" button in the track's console area.

Lock in Time If this box is checked, the waveform block is locked to its current time location in the multitrack session. Locking a waveform is useful when you've decided upon its position in time, and don't want to accidentally move the waveform when right-clicking on it. While a waveform which has been locked can't be moved in time (left or right), it may be vertically repositioned, allowing you to move it between tracks. To unlock a waveform, uncheck the Lock in Time box and it will be freed for horizontal movement.

Note: Locked waveforms are identified by an overlaid graphic that looks like a padlock in a circle.

Lock for Play Only Checking this box will set put the waveform block into "safe mode" so that it won't be recorded into. A locked-for-play-only waveform won't become red when position in a record-enabled track. To re-enable recording for the waveform block, uncheck this box, or to Edit > Lock for Play Only and remove its checkmark by clicking on it.

Transpose (MIDI Block Only)

If a single MIDI block is highlighted, this command appears on the Edit menu. Use Transpose and its submenu options to change the key of a MIDI block up or down by up to seven steps.

Set Tempo (MIDI Block Only)

If a single MIDI block is highlighted, this command appears on the Edit menu. It displays the Adjust MIDI Tempo box, where you can make changes to the tempo of the MIDI block using the Tempo field provided.

More information about the Set Tempo dialog can be found in the "Navigating the Multitrack View Screen" chapter.

Zoom in/Zoom out (MIDI Block Only)

If a single MIDI block is highlighted, this command appears on the Edit menu.

A MIDI note can be assigned to 127 note values. By default, Adobe Audition figures out which is the highest note in a MIDI block and which is the lowest and zooms in the note display in the block so that the highest note is at the top and the lowest is at the bottom. If you select Zoom Out, all the values are displayed on scale with the 127 note values, which gives you lots of extra space in the block.

Active Track (MIDI Block Only)

If a single MIDI block is highlighted, this command appears on the Edit menu. Its submenu lists all of the tracks in the MIDI block. Checkmark a track and Adobe Audition will playback only that track when the MIDI block is played. If you want to hear all of the MIDI block's tracks, checkmark the All item.

Set Controller 7 (MIDI Block Only)

If a single MIDI block is highlighted, this command appears on the Edit menu. The controller 7 value for a MIDI file is the master volume setting. This value gets sent out each time a MIDI file is played and tells the MIDI playback device what volume to play this file at. This option just lets you change the controller 7 value directly in case you have a MIDI file whose controller 7 value is too low or high.

Mix Down to File

This function creates a two-track stereo mix (unless a mono option is selected) of all or selected tracks and places the resulting waveform into Adobe Audition's Edit View. The contents of all or selected enabled (unmuted) tracks are combined, with track and waveform properties (such as volume and pan) affecting the way the final mix sounds. Session elements such as looping, images, and envelopes are all reflected in the mixed waveform.

All Waves Selecting this option will mix all unmuted waveforms in the session and place the resulting stereo waveform in the Edit View.

Selected Waves Choose this option to mix all highlighted waveforms in the session and put the resulting stereo waveform in the Edit View.

All Waves (Mono) Selecting this option will mix all unmuted waveforms in the session to a mono waveform, which will be placed in the Edit View.

Selected Waves (Mono) Choose this option to mix all highlighted waveforms in the session to a mono waveform, which will be opened in the Edit View.



Insert a submix back into your session using the File/Cue List in the Multitrack View's Insert menu. The original tracks that were submixed can then be muted or removed from the session.

Mix Down to Empty Track N (Bounce)

This command creates a two-track stereo or mono mix of all or selected tracks and places the resulting waveform into the next empty track in your session. Use the Mix Down to Empty Track N (Bounce) function to “bounce down” multiple tracks to a single track to speed up background mixing and clear up your work area.

The contents of all or selected enabled (unmuted) tracks are combined, with track and waveform properties (such as volume and pan) affecting the way the final mix sounds. Session elements such as looping, images, and envelopes are all reflected in the mixed waveform.

All Waves Selecting this option will mix all unmuted waveforms in the session and place the resulting stereo waveform into the selected track.

Selected Waves Choose this option to mix all highlighted waveforms in the session and put the resulting stereo waveform into the selected track.

All Waves (Mono) Selecting this option will mix all unmuted waveforms in the session to a mono waveform, which will be placed into the selected track.

Selected Waves (Mono) Choose this option to mix all highlighted waveforms in the session to a mono waveform, which will be opened into the selected track.

Group Blocks

This option will group any selected blocks together. Grouping can help with organizing the many blocks in your session. For example, you can have all of your guitar elements grouped together to easily identify and move them. Grouped blocks are designated by a different color than ungrouped blocks, as well as a special circular symbol overlaid each block.

- To group blocks together, select the members of the group by holding your Ctrl key down and left-clicking each desired block. Then, choose Group Blocks from the Edit menu, or from the right-click menu.
- To ungroup blocks, click on any member of that group to select it, and again choose Group Blocks from the Edit menu, or from the right-click menu. This will remove the checkmark from the menu item.

Grouped blocks will move together when selected and repositioned. Other wave operations such as Mute or Lock in Time will also affect all blocks in a group.

To select the color to use for a particular block group, right-click on any member of the group and choose Group Color.

Group Color

This command invokes Adobe Audition's Group Color dialog, which allows you to change the color of the current group of blocks. This function is documented in the "Navigating the Multitrack View Screen" chapter.

Block Color

This command invokes Adobe Audition's Block Color dialog, which allows you to change the color of the current block. This function is documented in the "Navigating the Multitrack View Screen" chapter.

Punch In

Punch In is a recording technique used to replace a section (perhaps a mistake of some kind) in an existing track segment. Replacing this segment usually entails starting and stopping record at exact moments in order to retain the surrounding audio, while material in-between these points is replaced with (hopefully) a better take.

Adobe Audition allows for multiple takes when Punching In; you can continually record over the original material and afterward choose between the best performance, or return to the original data. Punch In will work on one or more waveforms at a time.

To use Punch In:

- Highlight the area of the waveform block (or blocks) you would like to Punch into. For an exact selection, you can switch back to Adobe Audition's Edit View and use the Find Beats, Zero Crossings, and Snapping functions to make your selection, and then switch back to Multitrack View.
- Choose Punch In from the Edit menu, or from the menu that appears when you right-click on the waveform block. This will enable the selection for record and put the track into record mode.
- You can position the playback cursor sometime before the Punch In location and press the Record button. The session will begin playback, and upon hitting the Punch point, will record into the selected area.
- After recording, you can take another pass at the Punch. When finished, you can monitor the different takes by right-clicking on the waveform block and choosing them from the Take History menu option. Use the desired Merge Take command on the right-click menu when you've decided among them.



If possible, it's best to choose sections of the waveform where there are gaps in the audio for your Punch points. This will make it easier to achieve a natural sound, as if it were all one take.

Crossfade

Crossfades are used to create smooth transitions from the end of one audio segment to the beginning of another. They do so by creating a fade out and a fade in over the transition region. Waveforms to crossfade between should be positioned in different tracks (or can be in the same track, in a back-to-back manner), adjacent if possible.

To crossfade, make a selection with your mouse and include in it the beginning and ending sections of any waveforms you wish to include in the crossfade. The highlight should extend a bit beyond the end of the waveform(s) to fade from, and should start a bit before the beginning of the waveform(s) to fade to. Or, if Edit > Snapping > Snap to Blocks is enabled, the highlight will easily adjust itself to the beginnings and endings of the waveforms. (You can also adjust the highlight by holding the Shift key down when doing a left-click near the edge of the selection.)

After establishing a highlighted area, you need to select the waveforms to include in the crossfade by holding down the Ctrl key and left-clicking on the desired wave blocks. Next, choose Edit > Crossfade (or Crossfade from the right-click menu) and the type of crossfade you want to do to crossfade between the selected waveforms.

Three major types of crossfades are available. Choose Linear for an even grade, and Sinusoidal for a curved, sine-like slope to the fade. Logarithmic will fade the amplitude of a signal at a constant rate, producing a steeper slope at one end of the fade, depending on whether you are fading in or out. (Both Logarithmic In and Logarithmic Out options are at your disposal.)

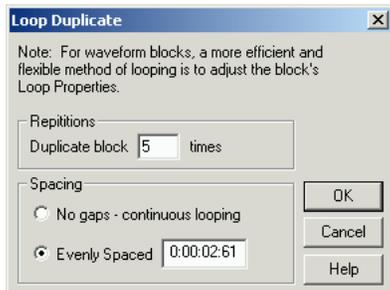
Note: *The Crossfade toolbar button uses Linear fades by default. Choose the Crossfade command from the Edit menu or right-click menu to specify a different crossfade type.*



The fade curves created with Crossfade can be modified by enabling the Show Volume Envelopes and Enable Envelope Editing options in the View menu.

Loop Duplicate

Loop Duplicate will create a specified number of consecutive Images of the selected waveform block creating, in effect, a loop of the waveform. Decide if you want no gaps between each duplicated wave form block, or if you wish them to be even spaced by a certain amount.



Loop Duplicate dialog

Note: For waveform blocks, a more efficient and flexible method of looping is to adjust the block's Loop Properties. You'll find this command on the menu that appears when you right-click on the wave block

Repetitions In this field, enter the number of times you'd like to duplicate the waveform block.

Spacing This determines the spacing at which to place each duplicated waveform block.

- **No gaps – continuous looping:** This option places each subsequent duplicate directly after the preceding block, for a continuous loop.
- **Evenly Spaced:** Choose this option if you would like to define a spacing to place between each block. The entry box to the right will display the spacing in the current Display Time Format. This value defaults to the length of the selected waveform block, which has the effect of functioning identically to the No gaps – continuous spacing option until changed. Entering a greater value will put space between each waveform block, while a lesser value will cause them to overlap.

 *Setting the Display Time Format to Bars and Beats makes it easy to place a waveform (such as a drum hit) at every other beat in your song, for example. Because each instance (other than the original) is an Image, you can place the instance throughout the entire song and only consume the drive space of the original occurrence.*

 *Find your loop points in Adobe Audition's Edit View with Find Beats, Snapping, and Zero Crossings. Using these functions you can make a selection (highlight) that starts and ends on zero crossings, which make musical sense. After doing so, choose Edit > Insert in Multitrack and switch back to the to the Multitrack View. You can now select the loop's waveform block and choose Loop Duplicate to create your loop.*

To create interesting time delay effects using the Loop Duplicate function:

- 1 Start by placing the source waveform into the Multitrack View.

- 2 Once entered, hold down the Shift key and right-click on the waveform and drag it to make an Image. The Image should be placed onto a new track (such as below the existing track)
- 3 Next, shift the Image in time, to the right of the original; small amounts to create phase shifts (small time offset), larger amounts to create delays (15-50 ms offsets) or echoes (50 ms or greater offsets).
- 4 Repeat this any number of times, shifting each Image slightly more to the right. You can even create in tempo delays by changing the Display Time Format to Bars and Beats and use the Snapping function to Snap To Ruler locations, placing the delayed Images at every beat, for example.

Try reducing the volume levels slightly for each Image, to soften the echoes (also, the combined waveforms will increase the track's overall level in the mix and could cause clipping). You can also create both Pan and Volume envelopes for each Image for truly dynamic delay effects that change over time. The great thing is this is all completely non-destructive!

Convert to Unique Copy

This command creates a copy of the selected waveform block. Copying a waveform block in the multitrack environment (as opposed to creating an Image) will create a separate audio file on disk, identical to the original in size and disk space consumption.

Creating a unique copy of a waveform block has the advantage of allowing for separate editing of each occurrence of the waveform (you can add reverb to the one placed in your intro, while the same sound placed in your verse can be dry).

Images have the advantage of not consuming any disk space, and depending on the situation, it may be an advantage to be able to affect all occurrences of a waveform block by editing the original once. (With Images, you can add a Flanger effect to one instance, and all 30 instances, for example would be flanged.).

 *To quickly add an effect to a waveform block within the Multitrack View, double-click on it (which will open up the waveform in the Edit View window). From there, you can use File > Save As... to create a copy of the file that can be transformed using any Effect function. The newly processed file can then be placed into the multitrack environment.*

To add multiple effects to an existing waveform non-destructively in Multitrack View:

- 1 Create a copy of the waveform using Edit > Convert to Unique Copy, or by saving the file under a new name using Save As... in the Edit View.
- 2 Once a copy of the original file has been made, add any processing or effect you like to it. For effects that have wet/dry balancing, set them to 100%, or close to 100% wet, so the result is primarily effect. (The original waveform that you copied will provide the dry element.)
- 3 Insert the processed copy into the multitrack environment, placing it into an adjacent to the original, in a different track. This way it will simultaneously playback both original file and the effected one.
- 4 You can then vary the effected waveform's (and the original's) volume and pan envelopes to bring the wet/dry in and out, or pan right and left.

Repeat this operation for other effects to have separate control over a number of effects (one for Flange, one for Reverb, etc.).

Mute Blocks

This command mutes or unmutes the selected block(s). This command is separate from Track Muting (pressing the Mute button in the Track Console area) which disables all blocks within a track. Muted blocks are ignored during playback and mixdowns, and are displayed as “grayed-out”.

Lock in Time

Lock In Time “locks” selected blocks to their current time location in the multitrack session. Locking a block is useful when you’ve decided upon its position in time, and don’t want to accidentally move the block when right-clicking on it. Locked blocks are identified by an overlaid graphic of a circled padlock.

While a block which has been locked can’t be moved in time (left or right), it may be vertically repositioned, allowing you to move it between tracks. To unlock a block, simply select it and choose Lock in Time to free it for horizontal movement.

Lock for Play Only

This command marks the selected block(s) as play only. Lock for Play Only disables the ability of a block to be recorded into, so while the track it resides in may be record enabled (have the red “R” button on the Track Console depressed), the marked block will not be recorded over. This is signified by the block not turning red when the track is record-enabled.

Split

The Split command functions similarly to a traditional tape splice; it “cuts” the block, segmenting it into parts. The cut is performed at the current cursor point, or selection (in this case the selection will be spliced out). Once a block has been split, each segment can be moved, deleted, or slid in time independently of the other segments. Splitting a block is non-destructive, as it does not alter the original file on disk.

Merge/Rejoin Split

Merge/Rejoin Split will recombine a split segment of a block with the original surrounding material, provided that the split segment sits at the same relative location as where it was split from, and in the same track as the original segment(s). The boundaries of the segments are joined together using a 30 millisecond crossfade. The Merge/Rejoin Split command can also be used to rejoin a take created using Punch In to the rest of the waveform.

***Note:** This merge operation is destructive in that it does alter your sound file. For example, when rejoining a Punch In segment, the file is rewritten with the new material.*

Align Left

The Align Left command (which is only accessible if two or more blocks are highlighted) to make the left-hand edges of all highlighted blocks the same as the left-hand edge of the last highlighted block.

Align Right

Click the Align Right command (which is only visible when two or more blocks are highlighted) to make the right-hand edges of all highlighted blocks the same as the right-hand edge of the last block you highlighted.

Adjust Boundaries

This function will adjust the boundaries of the selected block either inwardly or outwardly. After highlighting to define an area to adjust to, any material in the waveform block that lies within this highlight will be retained or exposed, removing that which is outside of the highlight (or including material inside the selection that is not currently revealed).

For example, with a five-second wave block, highlighting the third second and choosing Adjust Boundaries will trim off the surrounding material (the first two seconds and the last two seconds), leaving just the third second. If you decide you've trimmed off too much, you can highlight the middle second plus a little to the right, for instance, and again choose Adjust Boundaries to bring back that little bit to the right, thereby lengthening the wave block.

***Note:** To fine tune your adjustments in the preceding example, you could also have double-clicked on the trimmed block to bring up the Edit View, double-clicked to highlight the displayed area, then zoomed out to adjust the highlight boundaries to your satisfaction. When finished, switch back to Multitrack View (the highlight will be retained) and choose Adjust Boundaries to have the waveform block readjust to match your highlight.*

Trim

This Trim command works in the same fashion as Trim in the Edit View. All material will be removed from the selected waveform block except the portion within the highlight. Use Trim to remove unwanted material around a selection.

You may also trim an area using the Adjust Boundaries command. Trim differs from Adjust Boundaries in that Trim won't reveal any material within a highlighted region. Trim will only remove that which is outside of the highlight for the selected waveform block.

***Note:** Unlike the Trim command in Adobe Audition's Edit View, this version does not remove any data from the actual sound file. Trimming in Multitrack View simply tells Adobe Audition to "disregard this material" when playing back. You may return any trimmed block to its original state at any time by selecting it and choosing Edit > Full.*

Cut

Cut will remove the highlighted portion of a selected block. For example, if track 3 contains a wave block that is five seconds in length, selecting it, highlighting the third second, and choosing Cut will remove the third second, leaving the two-second portions on either side. If you want to remove unwanted material around a highlighted region, use Edit > Trim instead.

Note: Unlike Cut in the Edit Waveform View, this Cut command does not remove any data from the actual sound file, and places nothing on the clipboard. Using Cut in Multitrack View simply tells Adobe Audition to “disregard this material” when playing back. You may return any Cut block to its original state at any time by selecting it and choosing Edit > Full.

Full

This option returns any selected block to its original state. Any material removed using Trim or Cut (while in Multitrack View) will be replaced.

Note: Selecting Full for a block when its segments have been moved will return any missing material at that location, so this may result in having more than one instance of the same block portion throughout a session.

Remove Blocks

This command removes any selected (highlighted) blocks from a multitrack session. Removed waveforms will still be available in Adobe Audition’s Edit View. They’re also available to the session via Insert > Existing File/Cue until the audio files are closed from the File menu.

Destroy Blocks (remove & close)

This will remove the selected block(s) from the multitrack session, and close the files.

Select All Blocks

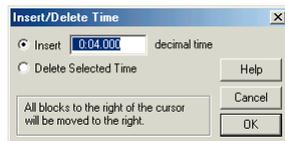
This command selects all blocks within the current session. Selected blocks appear brighter in color than their non-selected counterparts. If you’re using the default Adobe Audition color scheme, record-enabled blocks (red) will become bright red when selected, while regular unmuted blocks (green) will become bright green when selected.

Select All Blocks in Track N

This command selects all blocks in the currently selected track (the last one you clicked on, most of the time). Selected blocks appear brighter in color than their non-selected counterparts. If you’re using the default Adobe Audition color scheme, record-enabled blocks (red) will become bright red when selected, while regular unmuted blocks (green) will become bright green when selected.

Insert/Delete Time

Use Insert/Delete Time to introduce an amount of silence into the session at the cursor location, or to cut the selected area out of the session.



Insert/Delete Time dialog

Insert This option will shift all material (blocks or parts of blocks) to the right of the cursor position to a location after the time amount specified in the editable text field. Blocks will be split if necessary, and the specified amount of silence inserted.

Delete Selected Time If this option is chosen, the highlighted area will be removed, with all blocks to the right of the selected region shifted to the left.

Group Waveform Normalize

Group Waveform Normalize allows you to normalize the volume of multiple open waveforms using a three-screen batch process. (When you normalize a waveform, the loudest part of the waveform is set to a specified amplitude, thereby raising or lowering all other parts of the same waveform by the same amount.) If the volume is raised as part of the normalization process, Adobe Audition can apply limiting to prevent clipping.

Note: Complete documentation on the Group Waveform Normalize function can be found in the “Edit Menu – Edit View” chapter.

Snapping

Enable this function to have the edges of any block “snap” to certain time locations when being dragged. Snapping is used to align perfectly with time ruler locations when highlighting a region.

The Snap to locations (which vary depending on the Snapping option you have selected) act as magnets for any blocks dragged in their proximity, and have the effect of pulling the waveform toward them.

For example, if your Time Ruler’s format is hours:minutes:seconds, with seconds showing as the division at your current zoom level (1.0, 2.0, etc.), then dragging a block which starts at 1.6 seconds to the right will result in it snapping to the 2.0 location as it nears it. You can, of course, keep dragging beyond 2.0 in this case, if you wish.

Six snapping options are available from the pullout menu, and up to five of them may be checkmarked at one time. [Snap to Ruler (Fine) and Snap to Ruler (Coarse) can’t both be checked simultaneously.]

Snap to Cues With this option checked, the selection will snap to any time locations currently listed in the Cue List. Adobe Audition displays Cue markers in the Multitrack View as red and blue dotted lines.

Snap to Ruler (Fine) Check this option to have the selection or block snap to each subdivision in the current time display format (SMPTE, Bars and Beats, etc.) along the horizontal time ruler. Use this option for more sensitivity in Snapping. Zooming in will create further subdivisions to snap to, allowing you to Snap To with more precision.

Snap to Ruler (Coarse) Select this option to have the selection or block snap to each subdivision in the current time display format (SMPTE, Bars and Beats, etc.) along the horizontal time ruler. Use this option if you don't want too much sensitivity in Snapping. Zooming in will create further subdivisions to snap to, allowing you to Snap To with more precision.

Snap to Blocks This option snaps blocks to the beginning or ending points of other blocks. The beginning and ending points of blocks become the “magnets”.

Snap to Loop Endpoints If you select this option, blocks used as loops will snap to the beginning or ending points of other loops. The beginning and ending points of the loops act as “magnets”.

Snap to Frames (Always) When Snap to Frames (Always) is chosen, the selection or your cursor is always right on a frame boundary in whatever time format you're in – as long as the time format selected is measured in frames (such as Compact Disc and SMPTE). This option is especially handy for working on audio for CD.

Note: While dragging a block in Multitrack View, a white line will appear over the length of the tracks when Snapping points meet. For instance, if Snap to Blocks is selected, the white line will signal snapping as you drag a waveform block pass the beginning or end of other waveforms.

Refresh Now

Choosing Refresh Now instructs Adobe Audition to initiate the background mix from scratch. This will rebuild your session's output.

Check For Hidden Blocks

Run this command to see if you have any blocks completely hidden by other blocks in your session. If you do, the smaller blocks appear on top of the larger ones.

Chapter 17: Insert Menu – Multitrack View

Use the Insert Menu to import wave files, MIDI files, video files, and audio from video files into Adobe Audition's Multitrack View. Both active files (those already opened in Edit View) and inactive (not already opened) files may be easily inserted.

Wave from File

This command activates Adobe Audition's Open a Waveform dialog. It allows you to choose one or more supported wave files to be placed at the insertion point in the Session Display.

You can open several wave files at once by selecting them with the Shift key pressed (for contiguous selection) or Ctrl key pressed (for non-contiguous selection). Multiple waveforms are placed at the insertion point, each on its own track.

Note: More information on the Open a Waveform dialog can be found in the File Menu – Edit View dialog.

MIDI from File

Use this command to initiate the Choose a MIDI file dialog (which is very similar to a standard Windows Open dialog) which can be used to pick a MIDI file (files with a .mid or .rmi extension). Once you press the dialog's Open button, the MIDI file is placed at the Session Display's insertion point.

Note: Only one MIDI file may be inserted at a time, but you're welcome to re-run this command as many times as desired to place multiple MIDI files in a session.

Video from File

This command brings up the Choose a movie file dialog (which strongly resembles a typical Windows Open dialog) so that you can pick one video file for insertion into Adobe Audition's session window. A filmstrip graphic that represents the video appears on one track, while the audio from the video is placed in the track below. If you want to actually see the video's visual data, make sure Adobe Audition's Video Window is open (View > Show Video Window).

Note: Keep in mind that once your video file is inserted, the video block and its soundtrack are completely independent entities in the session. Moving or changing one will have no direct effect on the other. Also, the soundtrack wave block is considered to be a newly created audio file since it was extracted from the video, and will be treated as such.

Note: Just one video file may be open in Adobe Audition at a time. If you try to insert another one, you'll be prompted to close the currently open video file.

Audio from Video File

Insert the soundtrack from one or more supported video files to the session window using this command, which displays Adobe Audition's Choose a video file dialog. (This dialog is virtually identical to a normal Windows Open dialog.)

Note: You're limited to inserting audio from just one video file at a time, but feel free to re-run this command as often as need to insert addition video soundtracks into your session.

Currently Open Waveform List

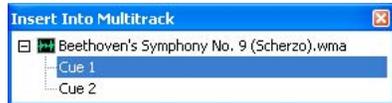
Entries for up to 10 open waveforms are listed.

Note: If you have more than 10 waveforms active in Edit View, only the last 10 opened will appear on the Multitrack View's Insert menu. Think of this section of the Insert menu as a MRU ("Most Recently Used") list of open waveforms. If you need to access another open waveform whose name isn't listed, use the Switch To... command or Adobe Audition's Organizer window.

File/Cue List

This command activates Adobe Audition's Insert Into Multitrack dialog. This window lists all of the files that are currently open in Edit View.

If a file has cues in it, a plus sign will appear next to its name to let you expand that file out and see all the cue ranges in it. This allows you to insert either the whole file, or expand it and grab one of the cue ranges directly.



Insert Into Multitrack Dialog

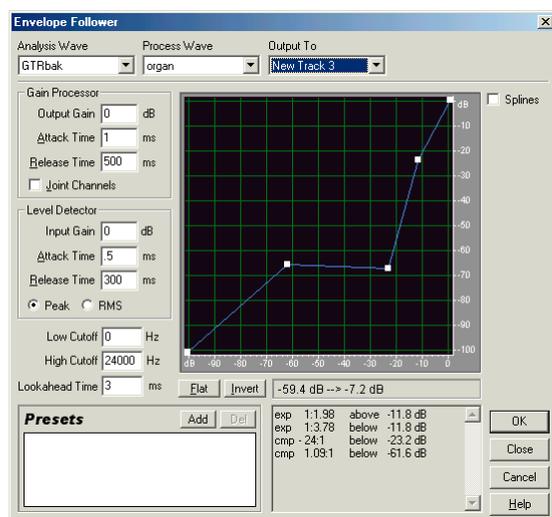
To place an item from the list at the session window's insertion point, just click on it. If you want to insert an item anywhere in the session window, just drag it off the list and drop it on the desired track. No matter which insertion method you choose, the Insert into Multitrack window automatically closes.

Chapter 18: Effects Menu – Multitrack View

This menu contains all of Adobe Audition’s “exclusively multitrack” effects.

Envelope Follower

The Envelope Follower varies the output level of one waveform, based on the input level of another. The amplitude map, or envelope of one waveform (the Analysis wave) is applied to the material of a second waveform (the Process wave), which results in the second waveform taking on the amplitude characteristics of the first waveform. This lets you, for example, have a bass guitar line which only sounds when a drum is being hit. In this example you would have the drum waveform as the Analysis wave, and the bass guitar waveform as the Process wave.



Envelope Follower Dialog

In addition to applying an amplitude envelope to a waveform, you can also alter the dynamic properties of the result with a variety of settings to otherwise expand, gate, compress, or limit the resulting signal.

To initiate Adobe Audition’s Envelope Follower effect:

- Position the wave blocks so that the sections you want to process together are lined up.
- Make a selection that covers the part of both wave blocks you want to process.

- Ctrl+Left-click each wave block you wish to process to give them focus.
- Access the Envelope Follower from the Effects pull-down menu, the Effects toolbar, or the Effects section of the Organizer window.

The following are the major components of the Envelope Follower dialog:

Analysis Wave Choose the waveform from which to “read” the amplitude envelope. Any waveform in the session window may be selected from this drop-down list.

Process Wave Select the waveform containing the material to apply the amplitude to. Any waveform in the session window may be chosen from this drop-down list.

Output To Choose the track to output the resulting waveform to. This entry defaults to the next available track.

Gain Processor Set the following:

- Output Gain: the gain (measured in dB) applied to the output signal. It’s the last step performed on the audio.
- Attack Time: the attack time (measured in milliseconds) applied just before output. Attack Time determines the time in milliseconds that it takes for the processed output signal to reach its specified output volume. If there’s suddenly a quiet portion that drops 30dB, it will take the amount of time specified in this field before the output actually drops to its corresponding volume level. If the sum of the Attack and Release times is too short (less than 20 milliseconds total), audible effects (such as a “vibrating” sound) can be heard at around 1000 Hz/ms total. For example, if the Attack and Release times are each set to 5 milliseconds (making 10 milliseconds total), then a vibrating sound at 100Hz will be heard. Thus, a total value of about 30 milliseconds is about as low as you can go without introducing these effects.
- Release Time: the release time (measured in milliseconds) applied just before output. Release Time is the time it takes the end of a previous output level to reach the specified output volume. For example, where the Attack is the time it takes for the start of a pulse to reach the desired output volume, the Release is the time it takes for the end of the pulse to reach the desired level.
- Joint Channels: With stereo audio files, each channel can be compressed independently, sometimes causing the surrounding background noise to get louder on one channel at a time – which can sound strange. For instance, a loud drumbeat in the left channel will make the background noise sound louder in the right than in the left. If Joint Channels is checked, both channels are used to find a single input dB value, and both channels will be amplified to by the same amount, thus preserving the stereo center-channel image. This means a loud drumbeat on the left channel will also cause the right channel to be reduced in level by an equal amount.

Level Detector Set the following:

- Input Gain: the gain (measured in dB) added to the signal before it goes into the Level Detector (the section that detects the current level). This essentially “pushes” the graph up or down by the gain given.

- **Attack Time:** the attack time (measured in milliseconds) applied when retrieving the current amplitude information. Attack Time determines the time in milliseconds that it takes for the processed output signal to reach its specified output volume. If there's suddenly a quiet portion that drops 30dB, it will take the amount of time specified in this field before the output actually drops to its corresponding volume level. If the sum of the Attack and Release times is too short (less than 20 milliseconds total), audible effects (such as a "vibrating" sound) can be heard at around 1000 Hz/ms total. For example, if the Attack and Release times are each set to 5 milliseconds (making 10 milliseconds total), then a vibrating sound at 100Hz will be heard. Thus, a total value of about 30 milliseconds is about as low as you can go without introducing these effects.
- **Release Time** is the release time (measured in milliseconds) applied when retrieving the current amplitude information. Release Time is the time it takes the end of a previous output level to reach the specified output volume. For example, where the Attack is the time it takes for the start of a pulse to reach the desired output volume, the Release is the time it takes for the end of the pulse to reach the desired level.
- **Peak/RMS:** Two methods of graph interpretation are provided for this effect: Peak and RMS.
 - Peak Mode is a little more difficult to use than RMS, and is considered by many to be outdated. It equates to the RMS value times two. That is, if the RM value is -20dB, then the equivalent peak value will be -40dB. This occurs because the RMS value calculated was mapped to a peak sample value for output. This method is basically here for backward compatibility.
 - RMS is the graph interpretation method that more closely matches the way people hear volume. This mode causes the output to be exactly the RMS amplitude that's specified in the graph. For instance, a limiter (flat horizontal line) at -10dB causes the RMS amplitude of the result to average -10dB (where 0dB is a maximum amplitude sine wave without clipping).

Low Cutoff This is the lowest frequency that dynamics processing will affect. You can define a band, or range, to which compression or expansion is applied, within the current frequency range. The Low Cutoff point is the bottom boundary in defining this band. For example, with values of 1000 for Low Cutoff and 5000 for High Cutoff, dynamics processing only affects audio in the frequency range of 1kHz to 5kHz. To use the entire frequency range of the source material, leave this setting at 0.

High Cutoff This is the highest frequency that dynamics processing will affect. You can define a band, or range, to which compression or expansion is applied, within the current frequency range. The High Cutoff point is the bottom boundary in defining this band. For example, with values of 2000 for Low Cutoff and 9000 for High Cutoff, dynamics processing only affects audio in the frequency range of 2kHz to 9kHz. To use the entire frequency range of the source material, this setting should be a one-half of the current sample rate (24000 for 48kHz, 11025 for 22kHz, etc.).

Lookahead Time Lookahead Time is used to handle sharp spikes that might occur at the onset of a louder signal. At times, and for brief instances, these onsets can go beyond the limits of your compressor settings, which may be desirable in certain compression scenarios since it can enhance the impact of, say, a drum hit. However, this is obviously not desirable if you're using limiting in order to reduce the maximum amplitude of the audio.

Note: The spikes occur because it takes a little time to determine (the Level Detector's attack value) and react (the Gain Processor's attack value) to the current signal level, so Lookahead Time will actually cause the attacks to start before the audio gets loud, instead of right on top of the transient. Otherwise, with a Lookahead Time of 0, a spike will stay loud until all of the attack times have elapsed.

Graph The graph depicts input levels along the x-axis (left and right) and the new output level along the y-axis (up and down). A line that flows directly from the lower-left to the upper right (which is the default) depicts a signal that's been left untouched, since every input value goes to the exact matching output value.

Adjusting the shape of the line will adjust the input or output assignments, thereby altering the dynamic range. For instance, you can boost all input that has a level of around -20dB, leaving everything else unchanged.

You can also draw an inverse line (a line from upper-left to lower-right) that will dramatically boost low amplitudes while dramatically surpassing high amplitudes. (In other words, all quiet sounds will be loud, and loud sounds will be quiet.)

Note: More information on Adobe Audition's graph controls can be found in the "Looking at the Work Area" chapter.

Splines Check this option to generate a smoother, best fit curve instead of a straight line between control points on the graph. When you use Spline Curves, the line will not ordinarily travel directly through the control points; rather, the points control the shape of this curve.

To get the curve closer to a control point, click to create more control points near the point in question. The more control points there are clustered together, the closer the spline curve will be to those points.

Use Spline Curves when you want very smooth curves instead of straight lines (with their discontinuities at the control points).

Flat Press this button to remove all control points with the exception of the points at each end of the graph line, thereby straightening out the line.

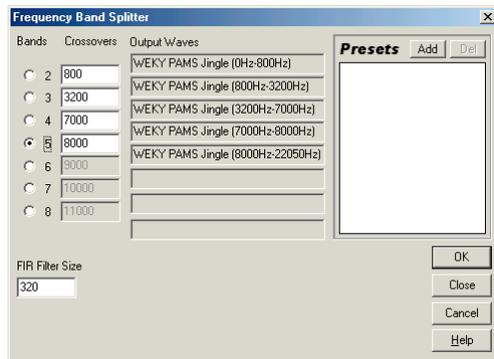
Invert The Invert button changes the graph to one that will function as the exact opposite. For example, if an effect function with a compressor characteristic is being displayed, pressing Invert changes the graph to one with the corresponding expander characteristic.

Note: For a graph to be invertible, it must have points in the two default corners (-100, -100 and 0, 0) and its output level must always increase from left to right (i.e., each control point must be higher than the one to its left).

Presets The Envelope Follower effect supports Presets for storing and recalling your favorite settings. More information about Presets can be found in the "Looking at the Work Area" chapter.

Frequency Band Splitter

Use the Frequency Band Splitter to take a selected waveform block (or a highlighted portion thereof) and make up to 8 copies of it, with each copy assuming a different frequency range of the original. Split points are determined by the crossover frequencies specified. Each copy of the waveform is placed in its own track in the session window.



Frequency Band Splitter Dialog

For instance, using the default setting of 3 bands with crossover values of 800 and 3200, three copies of the selected waveform are made: one that has the frequencies of the selected wave from 0 to 800Hz, one from 800 to 3200Hz, and one from 3200 to 22050Hz (or whatever the maximum frequency present is, based on the sample rate).

To initiate the Frequency Band Splitter dialog:

- Make a selection that covers the part of the wave block you want to process.
- Ctrl+Left-click the wave block you wish to process to give it focus.
- Access the Frequency Splitter from the Effects pull-down menu, the Effects toolbar, or the Effects section of the Organizer window.

Note: If you have more than one wave block selected, the Frequency Band Splitter effect will be unavailable.

The following are the major components of the Frequency Band Splitter dialog:

Bands Choose the number of split points here. The original waveform will be copied “this many times” with each copy having a different frequency range, as determined by the Crossovers.

Crossovers These are the crossover frequencies that are used for split points. If you don’t want to use Adobe Audition’s default values, enter different crossover frequencies in these boxes. The number of active text boxes is based upon the number of Bands chosen.

Output Waves Adobe Audition automatically assigns each copy of the original waveform block a new name. This is usually the original name of the waveform plus its frequency range. The names of the copies are listed in this column.

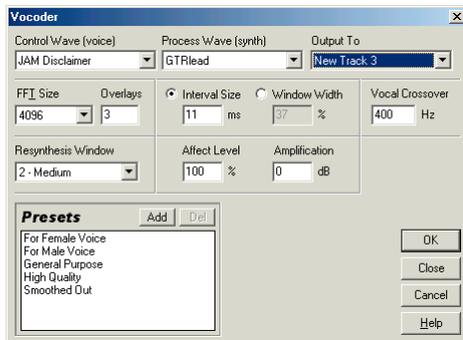
Max FIR Filter Size Adobe Audition’s Frequency Splitter effect uses an FIR (Finite Impulse Response) filter, which maintains phase errors over the response curve. This is unlike IIR filters, which can have phase error (often audible as a ringing quality). Higher numbers create higher accuracy in the frequency filtering that the effect does.

The default FIR Filter Size of 320 should work most of the time. However, if you get any distortion or ringing in the filtered waves, try increasing this number to get a more accurate filter.

Presets The Frequency Splitter effect supports Presets for storing and recalling your favorite settings. More information about Presets can be found in the “Looking at the Work Area” chapter.

Vocoder

A vocoder is a special effect which takes two inputs, usually an instrument of some kind and a voice, and modulates one signal (the process signal, usually the instrument) with the input of the other (the control signal, usually the voice). Specifically, the amplitude of given frequencies in the process signal are made to fluctuate in response to the same amplitude changes in the control signal, allowing for the sound of one instrument to be “controlled” with the other. In the example above, the instrument (the process sound) could be made to “sing” by affecting it with the voice (the control sound).



Vocoder Dialog

To initiate Adobe Audition’s Vocoder effect:

- Position the wave blocks so that the sections you want to process together are lined up.
- Make a selection that covers the part of both wave blocks you want to process.
- Ctrl+Left-click each wave block you wish to process to give them focus.
- Access the Vocoder from the Effects pull-down menu, the Effects toolbar, or the Effects section of the Organizer window.

The following are the major components of the Vocoder window:

Control Wave From this drop-down list, choose the waveform (from any that are already active in the session) to act as the control signal. This is usually a vocal of some kind.

Process Wave Use this drop-down list to select the waveform (from any currently open in the session window) to act as the process signal. This is normally some kind of synthesized sound to use to replace the vocal chords.

Output To Choose the track to output the resulting waveform. This entry defaults to the next available track.

FFT Size The FFT Size parameter specifies the size of the FFT to use, which can affect processing speed and quality.

- Higher values (like the default value of 4096) mean cleaner-sounding filters. Even larger FFT sizes should produce extra smooth results, but will take a bit longer to process.
- Smaller numbers make for a faster processing time, but the results may sound choppier.

Overlays This is the number of FFTs that are overlapped in processing the vocoder. More overlays can produce smoother results, but will take longer to process.



Values of 3 to 12 work well.

Interval Size This sets the time interval (measured in milliseconds) per FFT taken. Values between 10 and 30 milliseconds usually work best, but higher overlay settings may require you to vary the interval size used.

Note: *Smaller values can produce a hum, while large values can be too blocky sounding.*

Window Width As an alternative to setting the Interval Size, you can simply choose a percentage to use for Window Width in specifying the interval used per FFT taken.



A value of 90% will generally produce good results.

Vocal Crossover This frequency setting is used to filter out, or separate, the source waveform's underlying base frequency (voice) from the vocal formants. With higher values, more formants and less of the actual vocal tone from the source is used. As this value gets lower, more of the speaker's voice characteristics get vocoded in. (This shouldn't be confused with the speaker's formant characteristics.) Ideally, you want none of the source voice itself carried over, but all of the formant information, so the synthesizer "talks".

Resynthesis Window Choose the size of the window used for resynthesis of the vocoded signal. Narrower windows will make hard consonants sound clearer, and with higher overlay settings, narrow resynthesis windows will give better time resolution in the event that the vocoded signal sounds too smoothed out. The resynthesis window number should never be larger than or equal to the number of overlays.

Affect Level This sets the amount of processed (vocoded) signal that ends up in the resulting waveform. Set this value to 100% for full vocoding, while 50% keeps more of the original waveform.



A value of 15% will produce a subliminal effect, as it will just barely affect the process wave with the voice.

Amplification You can set an amplification amount in dB to adjust the final waveform by. Usually this can be zero, but if the results are too quiet or loud, raise or lower this to compensate.



For ease of use, try selecting Window Width and set it to about 90%, use 3 or 4 overlays, a resynthesis window of 1 or 2, and FFT sizes from 2048 to 6400.

Chapter 19: Options Menu – Multitrack View

The Options menu contains commands for customizing Adobe Audition's settings for the Multitrack View.

Loop Mode

The Loop Mode option toggles Adobe Audition's Play control between Loop and Play states.

- When Loop Mode is checked, pressing the Play control will continuously loop the current selection (or the entire session, if nothing is selected).
- When there is no checkmark, the Play control plays the selection (or session) once and then stops playback.

Monitor Record Level(s)

Checking this option will activate the Level Meters and start monitoring the recording source, which is useful for setting levels before recording.

To quit monitoring, press the Stop control or uncheck Monitor Record Level(s) on the Options menu.

Show Levels on Play and Record

This item enables or disables Adobe Audition's Level Meters used to visualize audio levels during recording and playback.

MIDI Trigger Enable

By checking MIDI Trigger Enable, any of Adobe Audition's shortcuts that have been assigned to MIDI events can be called from a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command.

Disable this option if you don't wish to have shortcuts respond to MIDI events.

***Note:** Before attempting to enable MIDI triggering, you'll need to choose a device for MIDI In that's recognized by Windows, such as a soundcard's built-in MIDI interface or a MIDI interface card. To do this, go to Options > Device Properties and click on the MIDI In tab. Use the drop-down list to select the device to be used for triggering.*



Use the keyboard shortcut of F6 to quickly toggle the status of the MIDI Trigger Enable function.

MIDI Panic Button

Sometimes MIDI devices get out of sync and get "stuck notes" where a note won't stop playing. Select MIDI Panic Button to send an All Notes Off command to every MIDI device that Adobe Audition knows about.

Synchronize Blocks with Edit View

When Synchronize Blocks with Edit View is checked and you switch to Adobe Audition's Edit View from Multitrack View, the current selection and view information will be transferred to the Edit View. For example, if you're only looking at seconds 4 through 27 on a wave block when you double-click on it in Multitrack View, you'll be looking at seconds 4 through 27 of that wave in the Edit View. The same goes with selection information if you highlight an area, then hit F12 or click the leftmost button on the File toolbar to switch views.

SMPTE Slave Enable

With this option turned on, Adobe Audition can synchronize its playback to SMPTE time code (MTC) generated from another device, such as a MIDI sequencer, or with the appropriate hardware, a VCR or tape deck. This is commonly referred to as slaving.

With SMPTE Slave enabled, Adobe Audition will update its current sync status in the lower left of the Status Bar to Opened MIDI Input Device when waiting, Synchronizing when establishing lockup, and Playback Synchronized when actually established.

***Note:** Before attempting to enable SMPTE sync, you'll need to choose a device for MIDI In that is recognized by Windows (such as a soundcard's built-in MIDI interface, a MIDI interface card, or other hardware device options.) To do this, go to Options > Device Properties to bring up the devices window. Select the MIDI In tabbed window, and then select your MIDI input device from the SMPTE Slave Device drop-down list.*

SMPTE Master Enable

Besides being able to slave to time code from another source, Adobe Audition can generate "master" time code for another program or a piece of hardware to follow.

After checkmarking SMPTE Master Enable from the Multitrack View > Options menu, you need to select a SMPTE time code format for the ruler. This can be done by right-clicking on the ruler and selecting Display Time Format. Then, selecting any format prefixed with "SMPTE" ("SMPTE 30 FPS", for example).

***Note:** The SMPTE code that Adobe Audition generates must be output through a MIDI port. To configure the MIDI port for your system, go to Options > Device Properties to bring up the devices window. On the MIDI Out tab you'll see a SMPTE Output drop-down list box. Adobe Audition will output the SMPTE time code to whichever device is selected in this box.*

SMPTE Start Offset

This command brings up the General tab of Adobe Audition's Session Properties window, where you can enter a time offset to use when synchronizing to SMPTE time code.

This is the time location that Adobe Audition will wait for before playback begins. Locations can be entered in either of the two formats listed to the right of the SMPTE Start Time Offset text box.

Note: Before attempting to enable SMPTE sync, you'll need to choose a device for MIDI In that is recognized by Windows (such as a soundcard's built-in MIDI interface, a MIDI interface card, or other hardware device options.) To do this, go to Options > Device Properties to bring up the devices window. Set MIDI In (Sync/Trigger) to the input to be used for SMPTE sync (commonly YourDevice: Sync).

Sample Accurate Sync

Some sound cards are capable of sending sample-accurate sync timing information to Adobe Audition. Engage this option to take advantage of this sample-accurate sync when slaving to SMPTE through such sound cards

Pause Background Mixing

Enable this option to halt Adobe Audition's background mixing process.

Background mixing pre-mixes audio before the audio is played. This can relieve some of your computer's processing load at playback time if some audio can be mixed before you even hit the Play button. Adobe Audition will function with background mixing turned off, but in most cases you'd be better off with it enabled.

However, you may want to pause background mixing if you are making lots and lots of edits, and don't want the hard drive trying to keep up with the background mixing only to have you make more edits which invalidate the previous mixing.

Metronome

This command brings up the Metronome tab of Adobe Audition's Session Properties window.

Note: The Metronome function is fully documented in the "View – Multitrack View" chapter.

Windows Recording Mixer

Select this command to display the Record Control part of the Windows Mixer (a.k.a. Volume Control) so that you adjust sound card recording levels.

Settings

You can customize Adobe Audition's colors, use of memory and hard disk space, multitrack behavior, SMPTE options, and miscellaneous other program characteristics in Options > Settings.

Note: Complete documentation on the Settings dialog can be found in the "Options Menu – Edit View" chapter.



Pressing the F4 key is a fast way to display the Settings dialog.

Device Properties

The Device Properties dialog is where you tell Adobe Audition which devices to use for playback and recording of waveforms, as well as input and output of MIDI signals.

Note: Complete documentation on the Device Properties dialog can be found in the “Options Menu – Edit View” chapter.

Device Order

The Device Ordering Preference window allows you to set the order in which Adobe Audition uses your system’s playback, recording, MIDI output, and MIDI input devices. The order of preference chosen determines which tracks get assigned to which devices when loading multitrack sessions on systems with different sound card configurations.

Note: The Device Order dialog is fully documented in the “Options Menu – Edit View” chapter.

Shortcuts (Keyboard & MIDI Triggers)

Shortcuts are used to speed up the editing process by allowing you to execute almost any command in Adobe Audition from your computer and/or MIDI keyboard. For example, instead of using your mouse to go to the Edit menu and selecting Cut to cut the highlighted portion of a wave block, you can simply press Ctrl+X.

Likewise, you can execute commands from a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command. This type of shortcut is referred to as a MIDI Trigger. An example of this would be assigning the Play command in Adobe Audition to the C4 note on your MIDI keyboard.

Note: The Keyboard Shortcuts dialog is fully documented in the “Keyboard Shortcuts” chapter.

Chapter 20: Multichannel Encoder

Adobe Audition includes the Multichannel Encoder, a self-contained dialog where you can access the tracks of any existing Multitrack session. In the Multichannel Encoder dialog, you can individually pan each track to your multichannel setup, preview the current mix and then Export the session as an encoded WMA Pro version 9 Multichannel file. It is also possible to export your session as 6 Mono Wave files or as one interleaved 6-channel Wav file for use with an external Multichannel Encoder such as a Dolby or DTS encoder.

To use the Multichannel Encoder, first open an existing Adobe Audition session, or create a new session in the Multitrack window. Once all your tracks are added, you next want to achieve a basic stereo mix balance with your desired track volume, stereo pan and FX settings. Then launch the Multichannel Encoder from the View menu.

Multichannel Encoder Requirements

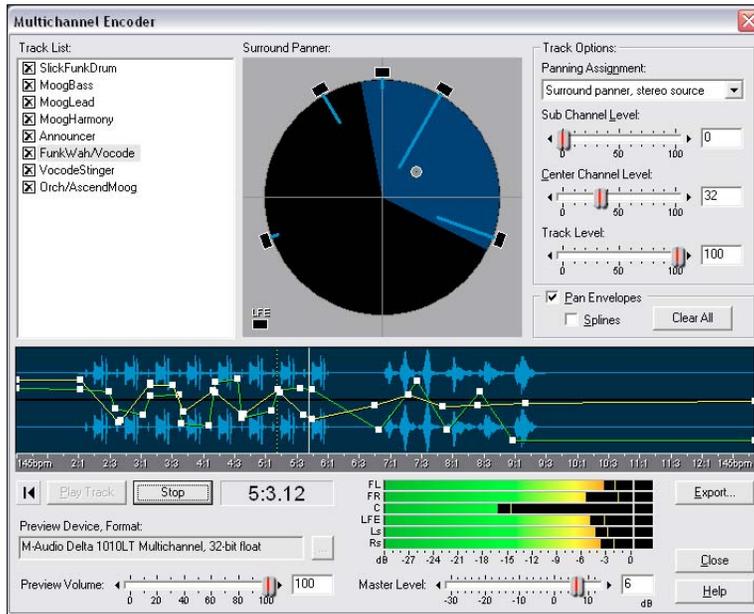
To achieve proper “5.1 Surround” preview playback from the Multichannel Encoder, it is necessary to meet the following requirements:

- A sound card installed and selected that offers at least 6-channel analog output, as well as a special interleaved device driver that is compatible with the Microsoft DirectSound multichannel format
- Microsoft Direct X 8.0 or later installed

If your PC does not meet these requirements then you may receive a warning dialog, and your Play Track and Play All buttons will not be accessible. To be able to Export and Encode your project to a 6-channel Windows Media file, you must have the Windows Media 9 runtime installed. If you have an earlier Windows Media version on your PC, the Encode to WMA9 option will not be available. The latest Direct X and Windows Media Updates are available on the Microsoft website.

Elements of the Multichannel Encoder

The Multichannel Encoder dialog box contains several elements, which are described in detail below.



Multichannel Encoder dialog box.

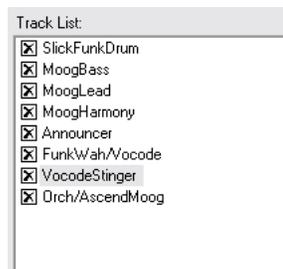
Track List

At the top left of the dialog is the track list. This list will show all the tracks and buses in use in the current Adobe Audition multitrack session. You select which tracks and/or bus outputs you want to be included in the multichannel mix by checking the box at the left of each track. Any track that is unchecked here will be removed from the multichannel preview and will not be included in the export of the multichannel project.

Panning Tracks and Buses separately Note that if you have routed a track's output to a bus in the Multitrack that you will likely not see that track appear in this track list. Instead, the bus that the track was assigned to will appear in this track list, and the bus output (which consists of all track signals routed to that bus) can be selected and panned as one mono or stereo signal. Additionally, it is also possible to access the "dry" track signal and pan it separately as well. This is achieved by going into the Adobe Audition bus tab at the left of the track, or into the Adobe Audition multitrack Mixer Window, switching to the Bus Mixer tab and adjusting the balance of the wet and dry values. Once the dry value is set greater than 0, this creates a selectable track entity that will appear in the Multichannel Encoder Track List the next time you launch it.

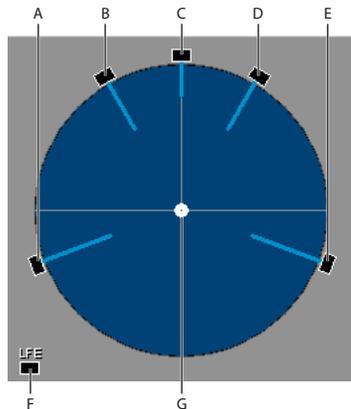
For an example, consider that track 1 of a session consists of a vocal wav file. You might have several other vocal tracks in this session as well and therefore you might add one Reverb effect on Bus A and set each vocal track's output to Bus A to have all be effected by this Reverb. But for track 1 you might wish to achieve a different surround panning effect where a non-effected track signal is panned to the front channels, while the effected bus output signal is sent only to the rear surround channels. If you set the track 1 bus wet and dry values each to a value such as 50 then you will be able to pan these two signals separately. Once these settings are all made within the multitrack session, launch the Multichannel Encoder dialog and you will then see that both Track 1 and Bus A are available as selectable entities that can be panned individually. This type of routing can be very useful for adding ambience and dimension to your multichannel mixes.

To perform panning adjustments on any track, select any one track in the list by clicking on the track name so that it becomes highlighted. For example, the VocodeStinger track is selected in the following example:



Track List

Surround Panner



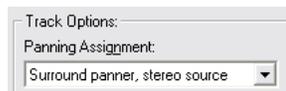
Surround Panner options:

A. Left Surround B. Front Left C. Center D. Front Right E. Right Surround F. Low Frequency Effects (Sub Bass) G. Panner Point

The Surround Panner allows you to position the selected track anywhere in the main speaker sound field by left clicking and dragging the white Panner Point. Note that as you position the Panner Point within the circular sound field the light blue Power Indicator lines coming from the speakers will change in length. This indicates the power balance of your sound source coming from each of the five main channels. Additionally, a portion of the sphere will appear dark blue in color to indicate the image of the sound field. That is, when seated in the center of the speakers, the blue area indicates where the listener would perceive the sound coming from.

You can also drag the Panner Point outside the sound field directly on top of one of the five main speakers or on top of the LFE speaker. Once the Panner Point is in any one of these speaker locations, the audio from the currently selected track is summed to a mono signal and sent discretely to this one speaker channel. This is an easy way to send the complete track signal all to one channel.

Panning Assignment Selector



Panning Assignment Selector

At the top right of the dialog is the Panning Assignment selector where you can choose to either use the Panning Interface to position your track sound source, or to make fixed panning assignments for your track. To follow is a description of each of the options in this list:

Surround panner, stereo source This option allows you to use the Panning Interface to position your sound source by dragging the Panner Point to the desired position. It also keeps your stereo left and right signals from your track discrete when panning in the sound field. For example, if your track includes a stereo file, the left stereo signal will be sent to the Front Left and Left Surround channels, your track's right signal will be sent to the Front Right and Right Surround channels. The Center channel always receives a summed to mono (L + R) signal. Therefore, as you pan in the five channel sound field, these stereo sources will retain their "stereo image" while being routed to the multiple channels.

Surround panner, summed to mono This option allows you to use the Panning Interface to position your sound source. However, this option always sums the track's signal to a mono signal. In this mode, panning the sound source to any location in the sound field will result in the summed mono signal being fed to all channels.

LFE only Sends the entire track signal to the LFE channel. Your monitoring system will apply the proper crossover frequency cutoff for reproducing the audio sent to the LFE channel. Typically most LFE components in 5.1 surround playback systems are set to a cutoff of < 80 Hz or < 120 Hz. The Adobe Audition Multichannel Encoder itself does not apply any filter to the LFE channel audio.

FL + FR, stereo Sends the selected track's signal as a stereo source directly to only the Front Left and Front Right speakers in a 50/50 stereo balance.

Ls + Rs, stereo Sends the selected track's signal as a stereo source directly to only the rear Left Surround and Right Surround speakers in a 50/50 stereo balance.

Center + LFE, stereo Note that this option is most useful with a stereo source file – when this option is selected for a stereo track, it will route the track’s left channel signal to the Center channel and the track’s right channel signal to the LFE channel discretely. If this option is selected for a track containing Mono source file, then the same signal will be sent equally to both the Center and LFE channels.

Center only, FL only, FR only, Ls only, Rs only These options each sum the selected track’s audio to a mono signal and send it all to the selected channel. This is the same as dragging the Panner Point directly onto one of the five main speakers in the Surround Panner interface.

Sub Channel Level Slider



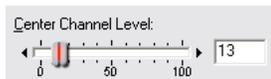
Sub Channel Level Slider

When in one of the Surround Panner modes, the Panner Point positions the sound source balance between the five main channels. However, you can also raise the amplitude of this slider to additionally send the track’s signal to the LFE channel. With this slider set to zero, no track audio is sent to the LFE channel.* If the currently selected track is assigned to only the LFE channel, this slider will attenuate the amount of this track’s output sent to the LFE channel.

It is also worth noting that Adobe Audition’s Multichannel Encoder does NOT apply filtering to audio sent to the LFE channel, nor does it apply any filtering during preview, exporting or encoding. This allows you to apply your own specific filtering as needed for your projects. Therefore, any low-pass filtering needed for your final LFE channel content should be applied to your audio within the Adobe Audition Multitrack, or on your Exported wav files.

It is recommended that you utilize a Bass Management circuit in your monitoring setup to ensure that you hear the representative mix levels that might be reproduced in an end listener’s playback system. Often home receivers and LFE equipped speaker systems will employ a bass management circuit on playback that will automatically route ALL sub-120 Hz or lower frequencies from all channels to the LFE speaker. Therefore, it is possible to overdrive the end listener’s system by sending too much overall sub-120 Hz audio to the LFE channel. Further, some playback systems and even encoder systems will apply a + 10 dB boost to LFE content. The use of a bass management circuit in your monitoring system will therefore allow you to compensate for these playback scenarios and adjust your mix accordingly.

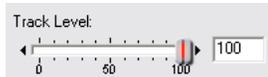
Center Channel Level Slider



Center Channel Level Slider

When in the Surround Panner modes, the Center slider determines the balance of the FL, Center and FR channels. With this slider at 0, no signal is sent to the Center channel. With this slider at 100, the Center channel receives an equal % of signal as the FL and FR. The position of the Panner Point then determines the positional panning according to this FL, Center, FR balance ratio.

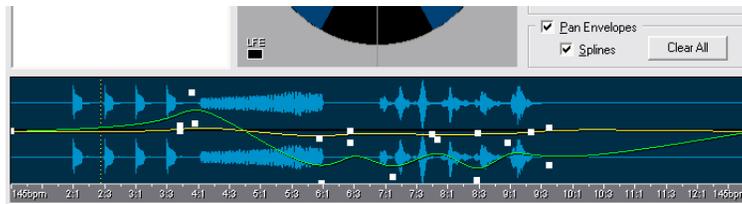
Track Level Slider



Track Level Slider

In any selected pan mode, this level slider controls the amplitude level of the currently selected track within the Multichannel mix.

Pan Envelope Automation



Pan Envelope Automation

When the Pan Envelopes checkbox is selected, two envelope lines are shown in the waveform display window. The yellow envelope line controls the Left/Right balance and the green envelope controls the Front/Surround balance. These envelopes are interactive with the positioning of the Panner Point in the Surround panner interface. It is possible to create dynamic panning over time by using these envelopes.

To create a dynamic pan on a track, follow these steps:

- 1 Select and check the box for one of your tracks in the Track List.
- 2 In the Panning Assignment menu, select either the Surround Panner, stereo source or Surround Panner, summed to mono options.
- 3 At the top right of the Waveform display section, check the Pan Envelopes checkbox. You will see two envelope lines appear in the Waveform display. The yellow line starts on top of the green line, though, so you may only see the yellow line until you change the pan position.
- 4 Left click in the waveform display at the desired time location where you would like to set a pan destination for the sound source. This moves the vertical cursor to this time location.

5 Now move the Panner Point to the desired position in the five channel sound field. Note that as soon as you move the Panner Point that two “handle” points are created on the Envelope lines within the Waveform display. These handles now move with your positioning of the Panner Point.

6 You can also left click directly on either of the envelope lines to create additional adjustable “handles” for shaping the envelope lines.

7 These Envelope handles are now also able to be edited by left click dragging them to any position, and the Panner Point will move in tandem to show you the relative position in the sound field during playback. If you wish to delete any single handle, simply drag it up or down beyond the boundary of the waveform display area.

8 To clear all envelope handle points and reset the track to flat envelopes, click on the Clear All button at the top right of the waveform display area.

9 Select the Splines checkbox if you prefer the envelope to use rounded Spline curves for smoother transitions between points.

10 Place the playback cursor back to the start of the track and select one of the Play buttons. Watch the Panner Point position and listen for the dynamic pan setting you just created.

If you prefer to keep your track panned to a fixed point throughout the duration session, then simply deselect the Pan Envelopes checkbox. This removes the Envelopes from the waveform display and allows you to set the Panner Point to any static position you like. You can toggle the Pan Envelopes setting on/off and any envelope points you have created for this track will be retained. Note that if the Pan Envelopes checkbox is not selected, you can drag the Panner Point during playback and hear your static pan positioning in real-time.

Zooming

There are several options for zooming in and out within the Waveform Display. To access the four available Zoom commands, place the mouse cursor over the time ruler which runs across the bottom of the waveform display and right click to bring up the context menu. The four available options here are:

- Zoom In
- Zoom Out
- Zoom Full
- Zoom to Cursor

You can also zoom in to a specific time area by right click-dragging the desired area on the time ruler itself. To Zoom back out again, right click and choose Zoom Out or Zoom Full from the context menu described above.

Another handy Zoom method is to place the mouse pointer anywhere within the waveform display and turn the mouse wheel. This will zoom into the time area directly beneath the mouse pointer. Reversing the mouse wheel rotation will zoom back out incrementally.

Transport Controls



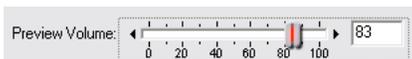
Transport Controls

Rewind to Start This left arrow button will place the cursor back at the start of the track.

Play Track This starts preview playback from the cursor location and only plays the currently selected track. Playback will always play to the end of the track, regardless of the current zoom level.

Play All This starts preview playback from the cursor location and plays the multichannel mix with all tracks that are checked in the Track List. Playback will always play to the end of the session, regardless of the current Zoom level.

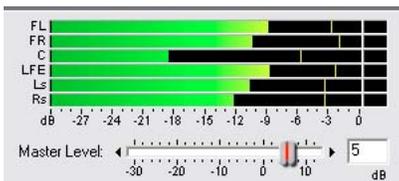
Preview Volume Slider



Preview Volume Slider

Controls the volume of the preview playback – note that this slider will not change the amplitude of the exported WAV or encoded WMA files that are created from the Multichannel Encoder, nor will it affect the levels measured by the 6-channel Output Meter. Use the Master Level slider for that use. This slider should be used to adjust your playback volume without affecting the amplitude of the exported files.

Output Meters



Output Meters

This set of six meters displays the output of each of the channels during Preview. During Play Track, the meters will display the output of only the selected track, and during Play All the meters will display the output of the complete 5.1 mix. These levels are what the actual levels will be for your exported wav or wma files from the session. The overall 6-channel level can be attenuated by the Master Level slider beneath the meters.

Master Level Slider



Master Level Slider

The Master Level slider will affect the audible level of your preview playback; however, this slider is primarily offered to adjust the amplitude of the exported or encoded files. Use this slider and reference the meters to optimize the overall peak amplitude of the 5.1 channel mix so that none of the channels are clipping.

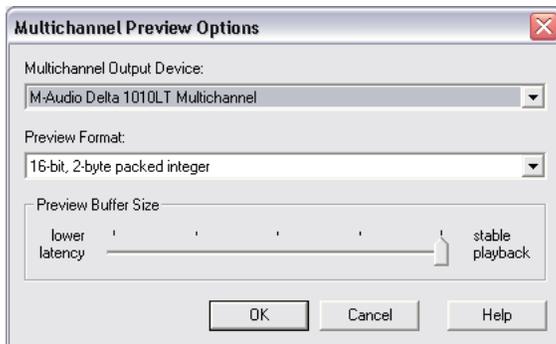
Preview Device Selection



Preview Device Selection

The Preview Device field displays the currently selected device that Adobe Audition will route its 6- channel output to. This field will also show the currently selected bit rate for preview playback. To make changes to these device and bit selections, press the Change button to the right of this field. This will launch the Multichannel Preview Options dialog:

Note: For information about device requirements, please see “Multichannel Encoder Requirements” on page 329.



Multichannel Preview Options dialog box

Some sound cards that offer 5.1 playback, such as the Creative Labs Audigy, will display only one device driver listed, and this is the device that should be selected since the sound card’s driver will route the 6 channels of audio to the correct speakers.

Some professional sound cards such as the M-Audio Delta 1010/1010LT or 410, Echo Layla24, Frontier Design Dakota and EgoSys WaMi units offer a single Multichannel Device driver, in addition to several individual stereo drivers. For these cards that offer this special “interleaved” multichannel driver, you should select this from the list. For example, for the M-Audio Delta 1010LT sound card that offers this type of driver, it shows up as “M-Audio 1010LT Multichannel, 16 bit.” These driver types will accept the 6-audio input from Adobe Audition and automatically route it to the standard Microsoft 5.1 channel configuration listed below.

Channel Order

When connecting your speakers to the analog outputs of a Multichannel sound card that offers the required interleaved multichannel driver, you should ensure that your speakers are connected in this order to correctly reproduce the 5.1 surround preview:

- Out 1 – Front Left speaker (FL)
- Out 2 – Front Right speaker (FR)
- Out 3 – Center speaker (C)
- Out 4 – LFE - Sub Woofer speaker (LFE)
- Out 5 – Left Surround (rear) speaker (Ls)
- Out 6 – Right Surround (rear) speaker (Rs)

Note that the above channel order is also the order in which your Exported 6-channel .wav and 6-channel .wma files will be created. If your project requirements include a different channel order (such as if you are submitting a master for use in a DTS encoded project or other surround format), then you should choose the Export to 6 mono wav files option.

Preview Format Selector

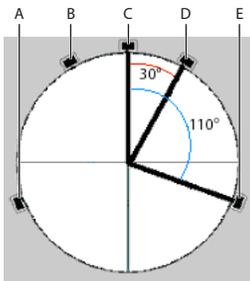
Use this drop down menu to select the bit rate of the preview playback material that is sent to your sound card. This can be kept at 16 bit, but if your session includes higher bit rate files, and if your sound card supports higher bit rate playback, you can select one of the 24 or 32 bit rates here to allow the preview to more accurately play back the higher bit rate content of your session.

Preview Buffer Size Slider

This slider is offered to allow adjustment of the size of the buffers used for both the Play Track and Play All functions. The slider can be adjusted to the right to create larger buffer sizes, which can allow for a more stable preview playback at the cost of raising the latency. A higher latency means that it will take longer for you to hear the result of changes made to any sliders or panning adjustments made while previewing. Adjusting this slider farther left will reduce the buffer size that can result in lower latency, but at the cost of less preview stability. If dropouts occur during preview playback, then the best action is to try moving this slider to the right. If you have a fast PC and/or if the session you are working on is a small one with few effects and panning envelopes, it may be possible to reduce the buffer size and take advantage of lower latency.

Speaker Placement

The recommended physical speaker placement is also important for accurate monitoring. Ideally, the five main speakers should be placed at ear level and each at an equal distance from your listening position. The LFE/Sub Woofer is a non-directional speaker, therefore, it can be placed anywhere in the room on the floor, but it would be practical to try moving it to slightly different floor positions to find the most accurate response within your environment. The following graphic illustrates the recommended setup for 5.1 speaker placement.

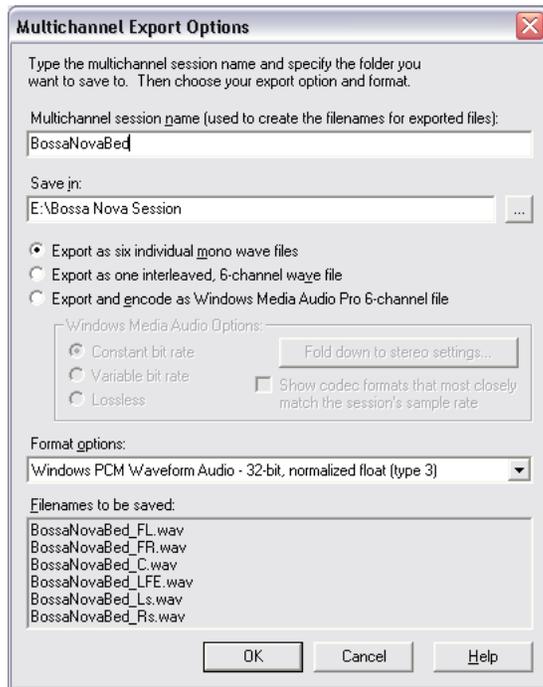


Recommended 5.1 speaker placement:

A. Left Surround B. Front Left C. Center D. Front Right E. Right Surround

Export options

Once you have completed mixing your multichannel project you will then want to export it to your desired file format. Adobe Audition includes the ability to encode directly to an interleaved 6-channel Windows Media 9 Pro (WMA) file or to export into two WAV formats. The field in the main dialog indicates the currently selected format, and this format is retained from your last used export option. To select your export/encode options, select the Export button at the bottom right of the window. This launches the export dialog:



Multichannel Export Options dialog box

Multichannel Session Name field The Multichannel Session Name field will automatically be completed with the name of your Adobe Audition Multitrack session if you had already saved the session and named it. You can type a new name in this field if you like. Since it is possible to Export to one of three different options (6-Mono Wav files, one interleaved 6-channel Wav file or encoded as one interleaved WMA 6 channel file), there are also different naming conventions for each. The following section explains the file naming conventions used for each scenario.

The text entered in the name field will be used as a common prefix for saving each exported file or files. Once the name text is entered into the name field, you can then select the desired Export format, and in the lower Filenames to be saved the full filename(s) will be listed to show you how the Exported file(s) will actually be saved. For example, if the text “BossaNovaBed” were entered in the name field, the following would be how the files would actually then be named for each Export format:

- For the 6-Mono Wav Export option, six files will automatically be rendered and saved with these names:
 - 1 – “BossaNovaBed_FL.wav” (Front Left channel file)
 - 2 – “BossaNovaBed_FR.wav” (Front Right channel file)
 - 3 – “BossaNovaBed_C.wav” (Center channel file)
 - 4 – “BossaNovaBed_LFE” (LFE channel file)

- 5 – “BossaNovaBed _Ls.wav” (Left surround channel file)
- 6 – “BossaNovaBed _Rs.wav” (Right surround channel file)
- For the Interleaved 6-channel Wave Export format, one file will be rendered and saved as “BossaNovaBed.wav”.
- For the WMA 6-channel Export format, the file will be encoded in the WMA format and saved as “BossaNovaBed.wma”.

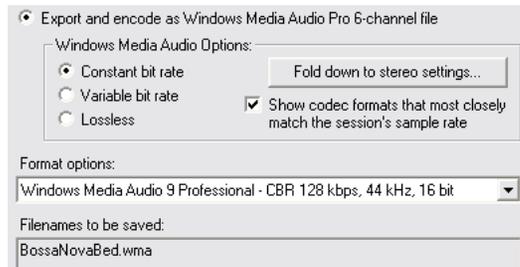
Save In This field lists the currently designated directory where the Export files are to be saved. To select a different directory, you can either type in an existing directory path, or click on the button to the right of this field and browse to the destination of your choice.

Export Formats Here you select the desired Export format. Below is a detailed description of each:

- Export as six mono wav files: If you want to export to a format that can then be utilized by another audio software, or by a software or hardware surround encoder, such as to encode your project as Dolby Digital or DTS, then the best option is to Export the project either as one interleaved 6-channel WAV file, or as 6-mono WAV files. The six mono wav files option will create standard Windows PCM .wav mono files that typically can be used by any Windows audio application.
- Export as one interleaved 6-channel wav file: The Windows PCM .wav format spec also allows for a single file to contain multiple channels of audio, however, not all Windows audio applications are capable opening or playing back wav files that are not mono or stereo. This interleaved 6-channel wav file type is commonly used by many hardware and software encoding systems. Choosing this option will save the multichannel session as one 6-channel file following the Microsoft “Wave Format Extensible” (which is also the SMPTE & ITU specification) channel order, which is defined as:
 - 1 - Front Left
 - 2 - Front Right
 - 3 - Center
 - 4 - LFE
 - 5 - Left surround (Rear Left)
 - 6 - Right surround (Rear Right)

This channel order is also the typical one utilized as the desired input for Dolby Digital encoders. However, if you are planning on preparing your project for a third party or specific encoding process, it is recommended that you ask the recipient exactly which format they require since channel order requirements can differ for different encoding systems. If a different channel order is required, then use the Export to 6 Mono WAV files when you export.

- Encode as WMA 6-Channel: Multichannel WMA files are capable of being played back by anyone that has installed the Windows Media Player version 9 or later, assuming of course they also have a multichannel output sound card and 5.1 speaker setup. Media Player 9 also requires that the user is running Windows XP as their operating system. If the end user has the Windows Media 9 player on any other Windows version, then the multichannel file will automatically be “folded-down” to play as a stereo file. For more information on the Windows Media 9 technologies, please visit the Microsoft Windows Media website at: <http://www.windowsmedia.com>



Export options

The WMA format utilizes a perceptual compression scheme and allows you to select from several different quality settings, including Constant Bit Rate (CBR), Variable Bit Rate (VBR) and Lossless. Just as with stereo WMA files, the higher quality setting you select, the larger the file size, and vice-versa. The Lossless option will compress to a smaller file size than wav, but will result in no fidelity loss whatsoever. To encode your project as WMA, select the WMA option and then the desired quality setting in the Format Options drop-down menu.

The checkbox option Show codec formats that most closely match the session's sample rate will limit the list of selectable WMA kbps option to those that are only the same sample and bit rate as the multitrack session's files. If you uncheck this box, the complete list of WMA options will be presented.

***Note:** As mentioned above, if your project is to be sent out to a mastering, duplication or other outside service with the intention of being encoded into other specific surround or media formats, you should inquire with the recipient as to the format specifics. Channel ordering differs between surround formats. Also, the crossover frequency points vary among different formats as well. For example, DTS typically employs a crossover of 80 Hz, meaning that all frequency content of your channels lower than 80 Hz can be routed to a sub woofer, and all frequency content greater than 80 sent to mains. This differs from the Dolby Digital system that utilizes a crossover point of 120 Hz. Some systems also employ a boost of 10 dB for the LFE channel, automatically assuming your LFE content will be approximately this much lower in power than the main channels. Therefore, these components should be accounted for in your mix before you deliver your master files to the recipient. It is best to inquire with the recipient about all such requirements to ensure that the audience will hear your project the same as you are hearing it on your monitoring system!*

- **Fold-Down to Stereo options:** If the 6-channel WMA file is played back on a non-Windows XP system, or on a system that does not have a 5.1 playback setup, Windows Media Player 9 will automatically perform a “fold down” or “down-mix” of the 6-channel playback to a stereo playback. The Windows Media Player 9 does this by accessing channel attenuation instructions that Adobe Audition's WMA encoder writes into the file. These attenuation parameters allow you to control how the levels of the Center, Surround and LFE channels get mixed down with the front stereo channels and played back on a stereo output system.



Fold down to stereo dialog box

To access the Fold down to stereo dialog, click on the Fold down to stereo settings button within the Multichannel Export Options dialog. The three numerical fields will default to -3, -3 and -12 dB as shown above. The defaults are usually good settings for most files, but you can enter any value in any of these 3 fields between 0 and -144 dB as desired.

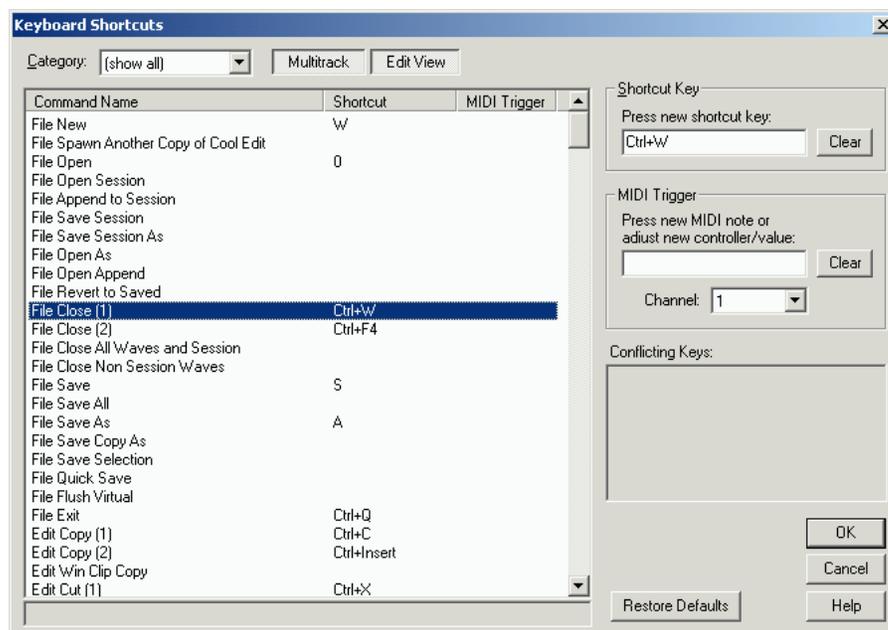
Opening Encoded and Exported files back in Adobe Audition

Once you have performed your Export from the Multichannel Encoder dialog and created the new WMA or WAV file(s), you may wish to open the files back into Adobe Audition. Adobe Audition is capable of opening 6-channel WMA or WAV files. To open a multichannel file in Adobe Audition's Edit View, simply go to File-Open as usual and this will open the multichannel file as six mono files. Note: 6-channel WMA files will only open as 6 mono files in Adobe Audition if you are running on Windows XP. If you are using Windows 98/Me/2000, the 6-channel wma files will be automatically folded down to a stereo file utilizing the Fold-down to stereo parameters mentioned in the previous section. This is a restriction of the Windows Media file format imposed by Microsoft on these operating systems. This restriction is not imposed on 6-channel WAV files.

Once these 6 mono files are created within Adobe Audition, they can then be edited and saved back as just as any mono file within any audio application that supports wav files. If you wish to use Adobe Audition to save the files back as one single, interleaved multichannel WMA or WAV, you can open the exported 6-channel file in Adobe Audition. This will automatically split the file into six mono files you can then insert them into six tracks of an Adobe Audition multitrack session. To "save" these files back as one interleaved multichannel file, it is necessary to launch the Multichannel Encoder dialog and assign each file back to its own individual channel and perform an export again as WMA Pro or 6-channel WAV as desired. You should of course be sure to set the pan assignment to the correct channel order if you wish to keep the same channel order as the opened 6-channel file.

Appendix A: Keyboard Shortcuts

Adobe Audition offers a wide variety of built-in keyboard shortcuts that can be used to speed up the editing process. Many users find these shortcuts help them to work a lot faster than using the corresponding menu commands, toolbar icons, or controls.



The Shortcuts (Keyboard & MIDI Triggers) dialog

The following list details the default shortcut commands that can be executed from your computer's keyboard. You don't however have to use the hotkey combinations we've created. Nearly all of these shortcuts, as well as commands that can be given using the keys on a standard MIDI keyboard instrument, may be changed using the dialog that appears when you access the Options > Shortcuts (Keyboard & MIDI Triggers) command (documented later in this chapter). Plus, there are dozens of other Adobe Audition functions that can be given keyboard shortcuts.

 Go to Options > Shortcuts (Keyboard & MIDI Triggers) and assign one-key shortcuts to the effects and functions you use most. For example, you might want to have the "N" key assigned to the Normalize effect, the "Q" key set to activate the Quick Filter dialog, and the "R" key programmed to bring up the Noise Reduction window. You'll be surprised how much quicker editing can be with one-key shortcuts.

File Menu Shortcuts

Ctrl+N	Create a new waveform or session
Ctrl+O	Open a new waveform or session
Ctrl+W	Close a waveform or session
Ctrl+F4	Close a waveform or session
Ctrl+S	Save a waveform or session
Ctrl+Q	Exit <i>Adobe Audition</i>

Edit Menu Shortcuts

Ctrl+C	Copy the waveform or selection to the clipboard
Ctrl+Insert	Copy the waveform or selection to the clipboard
Ctrl+X	Cut the waveform or selection to the clipboard
Shift+Delete	Cut the waveform or selection to the clipboard
Ctrl+V	Paste the clipboard's contents into the Wave Display or Session Display
Shift+Insert	Paste the clipboard's contents into the Wave Display or Session Display
Ctrl+Shift+V	Display the Mix Paste dialog
Ctrl+Z	Undo the last action
Alt+Backspace	Undo the last action
Ctrl+Y	Redo the last action
Ctrl+T	Trim the waveform so that only the highlighted part remains
Delete	Delete the selection
F11	Display the Convert Sample Type dialog
Ctrl+M	Insert the waveform into the Multitrack View's Session Window
1	Mark Intro Time
2	Mark Sec Tone
Ctrl+Shift+N	Paste the contents of the active clipboard to a new waveform
Ctrl+1	Set the active clipboard to internal clipboard 1

Ctrl+2	Set the active clipboard to internal clipboard 2
Ctrl+3	Set the active clipboard to internal clipboard 3
Ctrl+4	Set the active clipboard to internal clipboard 4
Ctrl+5	Set the active clipboard to internal clipboard 5
Ctrl+6	Set the active clipboard to the Windows clipboard
F8	Add the cursor position to the Cue List
Shift+F8	Add a track marker
Ctrl+F8	Add an index marker
F2	Repeat the last command (its dialog is shown)
F3	Repeat the last command (no dialog is shown)
Ctrl+B	Enable both channels
Ctrl+L	Enable the left channel only
Ctrl+R	Enable the right channel only
Up Arrow	Adjust the channel left or track up
Down Arrow	Adjust the channel right or track down
F5	Refresh the display of blocks
Shift+F5	Check for hidden blocks
Shift+Backspace	Destroy the selected block
Ctrl+Shift+Backspace	Destroy the selected block now
Ctrl+G	Group/Ungroup blocks
Ctrl+Delete	Clear Volume Envelope points
Ctrl+Shift+Delete	Clear Pan Envelope points
,	Nudge the selected block to the left
.	Nudge the selected block to the right
Ctrl+Shift+Up Arrow	Block color (next)
Ctrl+Shift+Down Arrow	Block color (previous)
Ctrl+K	Choose color for block(s)
Ctrl+Up Arrow	Select previous block in track
Ctrl+Down Arrow	Select next block in track

Scroll/Select Shortcuts

H	Adjust selection left side to the left
J	Adjust selection left side to the right
K	Adjust selection right side to the left
L	Adjust selection right side to the right
Ctrl+Home	Scroll to the beginning of the waveform or session when the view is zoomed in
Ctrl+End	Scroll to the ending of the waveform or session when the view is zoomed in
Ctrl+Page Up	Scroll the view one "page" to the left
Ctrl+Page Down	Scroll the view one "page" to the right
Ctrl+Left Arrow	Scroll the zoomed-in view to the left
Ctrl+Right Arrow	Scroll the zoomed-in view to the right
Home	Move the play cursor to the beginning of the waveform or session
End	Move the play cursor to the ending of the waveform or session
Page Up	Move the play cursor one "page" to the left
Page Down	Move the play cursor one "page" to the right
Left Arrow	Move the play cursor to the left
Right Arrow	Move the play cursor to the right
Escape (Esc)	Move the play cursor to the start of the waveform or session
Shift+Home	Extend the selection to the beginning of the waveform or session
Shift+End	Extend the selection to the end of the waveform or session
Shift+Page Up	Extend the selection one "page" to the left
Shift+Page Down	Extend the selection one "page" to the right
Shift+Left Arrow	Extend the selection to the left
Shift+Right Arrow	Extend the selection to the right
Ctrl+Shift+A	Select the current view
Ctrl+A	Select the entire waveform in Edit View; select all blocks in Multi-track View
Ctrl+Up Arrow	Select the previous block in the currently selected track

Ctrl+Down Arrow	Select the next block in the currently selected track
Shift+I	Adjust Zero Cross in
Shift+O	Adjust Zero Cross out
Shift+H	Adjust Zero Cross left side to the left
Shift+J	Adjust Zero Cross left side to the right
Shift+K	Adjust Zero Cross right side to the left
Shift+L	Adjust Zero Cross right side to the right
Shift+[Find the next beat to the left
Shift+]	Find the next beat to the right
[Anchor the selection to the left when playing
]	Anchor the selection to the right when playing

View Menu Shortcuts

F12	Toggle between Multitrack View and Edit View
Alt+9	Toggle display of Organizer window
Alt+8	Toggle display of the Cue List
Alt+6	Toggle display of the Sel/View controls
Alt+3	Toggle display of the Session Properties window
Alt+2	Toggle display of the Mixer window
Alt+5	Toggle display of the EQ window
Alt+4	Toggle display of the Track Properties window
Alt+1	Set focus to the main display
Alt+,	Set focus to the previous docked window
Alt+.	Set focus to the next docked window
Alt+Page Up	Activate the previous floating window
Alt+Page Down	Activate the next floating window
Alt+/	Flash the window that's currently in focus
Alt+Right Arrow	Zoom in horizontally

Alt+Left Arrow	Zoom out horizontally
Alt+Home	Zoom in to the left edge of the selection
Alt+End	Zoom in to the right edge of the selection
Alt+Up Arrow	Zoom in vertically
Alt+Down Arrow	Zoom out vertically
Ctrl+P	Display Wave Properties in Edit View only; display Advanced Session Properties in Multitrack View

Analyze Menu Shortcuts

Alt+Z	Toggle the display of the Frequency Analysis window
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Options Menu Shortcuts

F10	Toggle the monitoring of the input signal in the Level Meters
F4	Display the Adobe Audition Settings dialog
F6	MIDI Trigger Enable
F7	SMPTE Slave Enable
P	MIDI Panic (reset all MIDI devices)
Alt+K	Open the Shortcuts (Keyboard & MIDI Triggers) dialog

Transport Shortcuts

Alt+P	Play normally
Alt+S	Stop playback or recording
Space	Toggle between Play and Stop
Ctrl+Space	Toggle between Record and Pause
Shift+Space	Toggle between Play to End and Pause
Ctrl+Shift+Space	Toggle between Play All and Pause
Ctrl+Alt+Home	Move Play cursor to previous cue or beginning of waveform or session
Ctrl+Alt+End	Move Play cursor to next cue or end of waveform or session

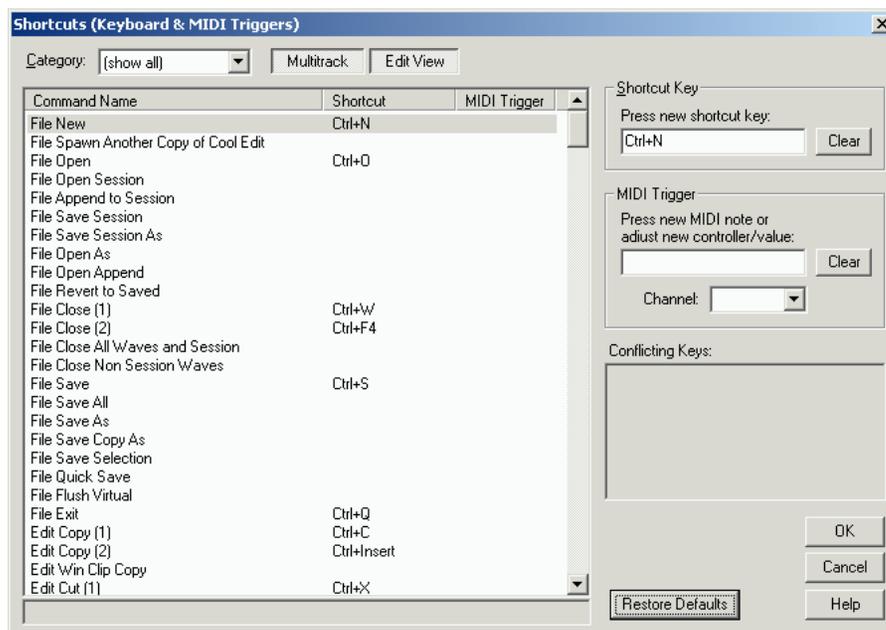
Help Menu Shortcuts

F1

Display the Help file's Quick Reference section

Keyboard Shortcuts Dialog

The Keyboard Shortcuts dialog (Options > Keyboard Shortcuts) is where you can customize Adobe Audition's existing keyboard shortcuts, as well as create new ones. One of Adobe Audition's best features is that most functions can have a keyboard shortcut assigned to them, which can make editing much faster.



Not only can you assign commands to your PC's keyboard, but you may also execute commands from a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command. This type of shortcut is referred to as a MIDI Trigger. An example of this would be assigning the Play command in Adobe Audition to the C4 note on your MIDI keyboard.

Note: Before attempting to enable MIDI triggering, you'll need to choose a device for MIDI In that is recognized by Windows, such as a soundcard's built-in MIDI interface, a MIDI interface card, or another similar hardware device.



When using Adobe Audition with a MIDI sequencer, you can trigger audio from your MIDI sequence by embedding a controller event at a specific location, and assigning playback in Adobe Audition to that event. This works well for shorter samples, but for longer audio segments you should use SMPTE to ensure synchronization.

Keyboard Shortcuts List All of Adobe Audition's commands and functions that can be assigned a keyboard shortcut are displayed in this list, along with any currently assigned shortcuts and MIDI triggers. Click on a command to highlight it, and then give it a shortcut (or edit its existing one) using the Shortcut Key and/or MIDI Trigger area of the Keyboard Shortcuts dialog.

Category Nearly every aspect of Adobe Audition can have a keyboard shortcut assigned to it – we're talking a lot of shortcuts. That's why you'll find the Category drop-down menu handy. It allows you to filter the Keyboard Shortcut list by category, which helps you quickly find the command or function to which you want to assign a shortcut.

Multitrack If the Multitrack button is depressed, you'll see all of Adobe Audition's keyboard shortcuts for the Multitrack View. Release the button to hide all Multitrack View shortcuts.

Edit View If the Edit View button is depressed, all of Adobe Audition's keyboard shortcuts for the Edit View are displayed. Release the button to hide all Edit View shortcuts.

Shortcut Key Use this section if you want to be able to execute the currently highlighted command from your computer's keyboard. Adobe Audition gladly accepts most single key shortcuts. (The most notable exceptions are the Print Scrn, Scroll Lock, Num Lock, Caps Lock, Tab, and Enter keys.) However, you can use the Ctrl, Shift, and Alt keys (or any combination of the three) to act as the first part of a multiple key shortcut.

With your text cursor in the Press new shortcut key field, press the desired key(s) to use for the currently highlighted command's shortcut key. Use the Clear button to remove an entry from the field.



Many Adobe Audition users find single key shortcuts (such as "N" for Normalize) faster to use and easier to remember.

MIDI Trigger Use this section if you want to be able to execute the currently highlighted command as a MIDI event. With your text cursor in the Press new MIDI note or adjust new controller/value field, press the desired MIDI note or adjust the new controller/value. Use the Clear button to remove an entry from the field.

You also may select the MIDI channel to receive the trigger from the drop-down Channel list to the right. (The default channel is 1.)

Conflicting Keys If a command's keyboard shortcut is already in use, you'll be notified here.

Restore Defaults Press this button to restore all keyboard shortcuts to Adobe Audition's default assignments.

Appendix B: Glossary

64-bit Doubles See DBL.

8-bit Signed See SAM.

ACM The Microsoft ACM (Audio Compression Manager) is part of all 32-bit versions of Windows. The ACM enables many Windows audio programs to compress and decompress files in a variety of formats, such as DSP Group TrueSpeech and GSM 6.10. Some of these formats come as a standard part of Windows, while others are provided by third parties. ACM formats are sometimes acquired when you install other software.

ADPCM ADPCM (Adaptive Differential Pulse Code Modulation) is an audio compression scheme which compresses sound files from 16 bits to 4 bits, yielding a 4:1 compression ratio. There are many varieties of ADPCM, such as the IMA (Interactive Multimedia Association) DVI standard, and versions from Microsoft, Dialogic, etc.

A-Law A-Law (or CCITT standard G.711) is an audio compression scheme common in telephony applications. It is a slight variation of the u-Law compression format, and is found in European systems. This encoding format compresses original 16-bit audio down to 8 bits (for a 2:1 compression ratio) with a dynamic range of about 13 bits. Thus, A-law encoded waveforms have a higher s/n ratio than 8-bit PCM, but at the price of a bit more distortion than the original 16-bit audio. The quality is higher than you would get with 4-bit ADPCM formats. Encoding and decoding are rather fast and generally widely supported.

Adapter A passive cable, plug, or jack that enables you to connect two audio or video devices together.

ActiveMovie See DirectX.

ADAT A digital 8-track tape deck manufactured by Alesis Corporation that has been very popular in recording studios.

Aliasing Noise that occurs from sampling a high frequency sound at a sample rate that is less than what is required to represent that frequency. A certain sampling rate can represent a frequency equal to half its rate (the Nyquist Frequency).

Amiga IFF-8SVX See IFF or SVX.

Amplitude Amplitude represents the loudness of a signal. A waveform's amplitude is measured by its distance from the center line (which represents a 0 amplitude value). There are different standards for measuring amplitude, with decibels (dB) being the most common.

Analog Recording This is the recording of the ongoing changes of an audio waveform. The analog recording of audio is a continuous curve, as opposed to digital recording, which is based on discrete samples.

AIF See AIFF.

AIFF This is Apple's standard wave file format. Like Windows WAV, AIFF files support mono or stereo, 16-bit or 8-bit, and a wide range of sample rates. Adobe Audition only supports the PCM encoded portion of the data, even though this format (like Windows WAV) can contain any one of a number of data formats. The AIFF format is a good choice for PC/Macintosh cross-platform compatibility. Extensions are .aif or .snd.

ASCII Text Data Audio data can be read to or written from files in a standard text format, with each sample separated by a carriage return, and channels separated by a tab character. The file extension is .txt. An optional header can be placed before the data. If there is no header text, then the data is assumed to be 16-bit signed decimal integers. The header is formatted as KEYWORD: value with the keywords being: SAMPLES, BITSPERSAMPLE, CHANNELS, SAMPLERATE, and NORMALIZED. The values for NORMALIZED are either TRUE or FALSE.

Attack The first part of the sound that you hear. Some sounds (like pianos and drums) have a very fast attack; the loudest portion of the sound is reached very quickly. A sound with a slow attack rate (such as a soft string section) slowly increases in volume.

Attenuate To make smaller, such as to reduce volume or signal level.

AU This format is the standard found on NeXT and Sun computers, and has many data types. Adobe Audition supports the CCITT mu-Law, A-Law, G.721 ADPCM, and linear PCM data variants. Like Windows WAV and AIFF files, this format can support mono or stereo, 16-bit or 8-bit, and a wide range of sample rates when saved as linear PCM. The most common use for the AU file format is for compressing 16-bit data to 8-bit mu-law data. AU is used quite extensively for distribution on the Internet, and for inclusion in Java applications and applets. AU files can have either an .au or .snd extension.

Audio File Format The method used for writing audio data to disk for storage is known as a File Format. Adobe Audition supports many file formats, which themselves support a variety of properties such as sample rate, compression, etc. Some file formats may not be compatible with other systems. Windows PCM (.wav) is found most often in the Windows environment. File formats can be chosen in File > Save As...

Audition Loop See CEL.

Automation The process of making pan and volume changes in a mix that will happen every time the file plays back. In many hardware mixers, this is done with “automatic” volume and pan sliders that can be programmed to remember timing information and move themselves appropriately during playback. Automation is handled by Envelopes in Adobe Audition.

Background Mixing This is the process that Adobe Audition uses to combine waveforms for monitoring while in the multitrack environment. As its name implies, background mixing occurs behind the scenes, and works much of the time to reflect changes to your session, such as a moved or deleted waveform, a volume change, or new material recorded into a track. An update of the background mixing process can be seen in the Mix Gauge.

Band Pass Filter A filter or DSP effect that allows some audio frequencies to pass, but not others.

Basic Cue One of the four types of Adobe Audition Cues. Basic Cues are used to mark important sections of the waveform for later reference (such as to remind yourself of an editing point). They’re also used to specify stop and start positions for the Adobe Audition’s Play List.

Beat Cue One of the four types of Adobe Audition Cues. Beat Cues are just like Basic Cues, but they’re specifically for marking musical beats.

Beats Per Minute (BPM) The number of steady even pulses that occur in a piece of music every 60 seconds. The BPM value defines the tempo of the musical selection.

BIOS Basic Input Output System. It's the most basic system for controlling a PC. BIOS routines are usually stored on hardware on the motherboard of a computer system (i.e., stored in a ROM chip). It's the BIOS that handles the instructions for starting a computer, initializing the hardware, and launching the operating system.

Bit Bits are part of the numbering system used in digital equipment, such as computers. A bit makes up a larger number, called a word, which is used to represent the voltage level (amplitude) of an analog signal. The number of bits used to make up a word determines the bit resolution, which in turn determines the number of voltage levels possible in representing amplitude. Therefore, greater bit resolutions make for higher dynamic range.

Bit Resolution (a.k.a. Bit Depth) The number of bits used in measuring amplitude for a sample. Choosing 8-bit resolution will provide 256 possible unique "volumes", while choosing 16-bit resolution will provide 65,536 possible unique volumes, which results in a 96 dB signal-to-noise ratio. Obviously, a much greater dynamic range can be reproduced at 16-bit resolution than at 8-bit resolution, which only has a 48 dB dynamic range. Compact disc players have 16-bit resolution. Some sound cards support resolutions greater than 16-bit. Adobe Audition supports up to 32-bit sample rates. It's best to remain at the 32-bit level while working in Adobe Audition, and convert down for output, in order to best represent the audio after transforms have been performed.

Block A segment of audio in Adobe Audition's Multitrack View that can be dragged separately. A block can be one of several data types: a wave block, a MIDI block, or a video block.

Brown Noise Brown noise has a spectral frequency of $1/f^2$. This means, in English, that there's much more low-end, and there are many more low-frequency components to the noise. This results in thunder- and waterfall-like sounds. Brown noise is so called because, when viewed, the wave follows a Brownian motion curve. That is, the next sample in the waveform is equal to the previous sample, plus a small random amount. When graphed, this waveform looks like a mountain range.

Bus An electrical conductor that makes a common connection between several circuits. In the case of Adobe Audition's Multitrack View, a bus lets you make a common connection between several tracks.

Bus Effects Configuration This dialog shows tabs for each effect on a given bus. It's where you actually set the properties for that effect, such as the delay length.

Bus Effects Rack The Bus Effects Rack dialog is the initial dialog used for deciding which real-time effect(s) you wish to add to or remove from a bus.

Burn Burning is the act of writing to a CD-R or CD-RW disc.

CD-R A CD-R is a blank, recordable compact disc that can be written to once. It typically holds at least 650 MB of data or 74 minutes of audio. CD-R can also refer to the computer drives that write, or burn, CD-R discs.

CD-RW A rewritable CD-R disc. A CD-RW also typically holds at least 650 MB of data or 74 minutes of audio, but unlike a CD-R, a CD-RW disk can be erased and written to again.

CEL Audition Loop files are essentially MP3 files with a .cel extension, but with a header that contains two important items. The first is the loop info, just as it's saved with a .wav file and is set under View > Wave Properties > Loop Info. The .cel file also works to avoid a problem with MP3 files. It's in the nature of MP3 files that a very small amount of silence is added to the beginning and/or end of an MP3 file when it's encoded. When working with a loop, though, it's enough to throw the entire loop off. As it saves a .cel file, Adobe Audition calculates how much silence is going to be added to the MP3 file and writes this information into the .cel header. Then, when Adobe Audition loads a .cel file, it reads this silence information and automatically removes the silence from the file so that it'll loop smoothly.

Chorus A delay effect that simulates several voices. It does this by adding multiple delays with a medium amount of depth and a mild amount of feedback.

Click Track An audio track comprised of clicks that occur on the beat, like a metronome. They are often used at the beginning of a session as timing information for musicians, and then removed from the session before production is complete.

Clipping When the amplitude of a signal exceeds the maximum level for the available current conditions (for example, 256 in 8-bit audio), a process known as clipping occurs. Clipping causes the signal to distort, and appears in the display as a "chopping-off" of the top of the waveform. If you experience clipping, try lowering recording input or source output levels.

CODEC Stands for Coder/Decoder. It's an abbreviation used often when speaking of multimedia compression modules, such as ACM modules (often called ACM CODECS), MPEG compressors, QuickTime, AVI, and the combined A-D-D-A modules on some soundcards.

Compressor Compressors reduce the dynamic range of an audio signal. In the real world, it's generally an amplifier with two gain levels: the gain is unity for input signal levels below a certain threshold, and less than unity for signals with levels above the threshold. For example, compressors can be used to eliminate the variations in the peaks of an electric bass signal by clamping them to a constant level, thus providing an even, solid bass line. Compressors can also be useful in compensating for the wide variations in the level of a signal produced by a vocalist who moves frequently or has an erratic dynamic range.

Creative Sound Blaster See VOC.

Crossfade Fade from one audio track to another.

Crosstalk Undesired leakage of audio from one track to another, which was common on analog tape. This is impossible in Adobe Audition since each track is stored as a separate digital audio file.

Cue List A list of time offsets, or locations defined within an audio file. A cue can be either a point that specifies a cursor position, or a range that specifies a selection. You may define and save an unlimited number of cues in Adobe Audition's Cue List for later recall, or for assembling in Adobe Audition's Play List.

DAC Digital to Analog Converter. The hardware responsible for converting a digital audio or video signal into an analog signal that can be played back.

DAT (Digital Audio Tape) A standard two-track digital audio tape format. DAT tapes are sampled at 16 and 24 bits, and 32,000, 44,100, and 48,000 samples per second. (The latter is considered DAT quality).

DC Offset Many sound cards record with a slight DC offset, which means that direct current is introduced into the signal by the card, causing the center of the waveform to be off from the zero point in the waveform display. This is seen as the waveform being “shifted” above or below the center line in the Wave Display, and can cause a click or pop to be played at the beginning and the end of the file. To compensate for DC Offset, you can choose the DC Bias Adjust setting in Effects > Amplitude > Amplify.

DBL The 64-bit doubles wave format (.dbl) is 8-byte doubles in binary form—8 bytes per sample mono, or 16 bytes per sample stereo interleaved. Just as raw PCM has no header and is just audio data, so does 64-bit doubles format.

Decibel (dB) In audio, a decibel (dB) is a logarithmic unit of measurement used for amplitude (specifically, to compare sound pressure with a reference pressure). In digital audio, 0dBFS is the maximum possible amplitude value (as in 256 for 8-bit audio). The value in dBFS does not relate directly to the original absolute sound pressure level of the audio measured in dB. Many areas of Adobe Audition refer to, or can refer to, amplitude in decibels.

Delay A time-shifted signal which can be mixed with the original, non-delayed signal to give it a fuller sound or to create echo effects. Adobe Audition offers a variety of Delay Effects such as Reverb, Chorus, and Echo.

Destructive Editing Destructive editing means simply that edits (cut, paste, effects, etc.) and changes are applied to the wave file itself, so that the original audio data itself changes, unlike nondestructive editing. For example, in destructive editing, a change in audio volume would actually alter the amplitude of a wave file. In Adobe Audition even “destructive” edits are not applied directly to the wave file in use until you save the file.

Device Devices are used to send data into and out of Adobe Audition. This includes both Wave and MIDI devices. Wave devices generally take the shape of sound card Inputs and Outputs, and are used for the recording and playback of audio material, while MIDI devices are used to send performance and sync information to Adobe Audition and other MIDI-enabled programs or hardware. Devices are selected in Options > Device Properties.

Dialogic ADPCM See VOX.

DiamondWare Digitized See DWD.

Digital Signal Processing (DSP) The process of transforming a digital audio signal from one form to another via complex algorithms. Examples include filtering or creating a sound-simulation effect, such as a reverb or echo.

DIMM Dual Inline Memory Module. A type of RAM.

DirectX DirectX (formerly ActiveMovie) is a development platform designed by Microsoft for 32-bit versions of Windows which provides an open standard for audio plug-ins. DSP modules and other audio tools based on this standard can be used by any application that supports the ActiveMovie/DirectX architecture (such as Adobe Audition).

Dither Dithering is the process of adding small amounts of noise to a digital signal so that fainter audio (below the noise you add) can be heard that would otherwise be truncated away when converting from a higher bit rate to a lower one (e.g., converting 24-bit to 16-bit).

DMA Direct Memory Access. A method of transferring data directly from memory to a device without passing through the CPU. Hardware that uses DMAs can pass data much more quickly. DMA addresses can not be shared between multiple devices.

Dry Used to describe an audio signal without any signal processing (such as reverb); the opposite of Wet.

DSP See Digital Signal Processing.

DVD Digital Video (or Versatile) Disc. A storage medium similar to Compact Disc (CD), but with much higher bandwidth and storage capabilities. Audio stored on DVD movies is generally 96 kHz/24-bit.

DVI See IMA ADPCM.

DWD This is the audio format used by DiamondWare's Sound Toolkit, a programmer's library that lets you quickly and easily add high-quality interactive audio to games and multimedia applications. It supports both mono and stereo files at a variety of resolutions and sample rates.

Echo A distinct repetition of a sound, due to the sound reflecting off a surface. Adobe Audition offers two Echo Effects (Echo and Echo Chamber), each with several settings.

Envelopes Adobe Audition uses several types of envelopes (Pan, Volume, Wet/Dry, and FX Parameter) for mix automation. The envelopes can be drawn directly on the blocks in the Multitrack view, and represent the pan, volume, wet/dry, and/or FX parameter settings at any point along the track. For instance, when the volume envelope is at the top of the wave block, the audio is at full volume, and when it is at the bottom, it's at zero volume.

Equalization (EQ) Equalization is the process of increasing or decreasing the amplitude of audio signals at a specific frequency band relative to the signals at other audio frequencies.

Expander Expanders are used to expand the dynamic range of an audio signal. (They are basically the opposite of a compressor). An expander can also be considered an amplifier with two gain levels: the gain is unity for input signal levels above a certain threshold, and less than unity for signals with levels below the threshold. An expander boosts the high-level signals and attenuates low-level signals.

Fast Fourier Transform An algorithm based on Fourier Theory that Adobe Audition uses to perform its filtering functions, as well as Spectral View and Frequency Analysis functions.

Flange An interesting audio effect caused by mixing an approximately even amount of a varying short delay with the original signal. Adobe Audition's Flanger effect can be found in Effects > Delay Effects.

Flushing The process Adobe Audition performs when it copies the audio data from a waveform file to Adobe Audition's temp folder so that the original file can be closed. This allows the file to be renamed, deleted, or opened exclusively by another application. Flushing sometimes occurs when a modified waveform is saved on top of its original file.

Fourier Theory Fourier Theory states that any waveform can be made up of an infinite sum of sin and cos functions. By using Fourier Theory, waveforms can be quickly analyzed for their frequency and amplitude content.

Frequency Measured in Hertz (Hz), a sound's Frequency is the rate at which it vibrates, or its cycles per second. A cycle consists of movement from a starting point (0) through both positive and negative amplitudes, eventually returning to the starting point. A sound's frequency determines its pitch.

FX An abbreviation for the word effects.

Hertz (Hz) This is a unit of measurement used to measure the frequency of a sound. Hertz are equal to cycles per second.

IDE Integrated Drive Electronics. A type of hard drive and CD-ROM interface that's the most common interface (as opposed to SCSI) in PCs.

IMA ADPCM The International Multimedia Association (IMA) flavor of ADPCM compresses 16-bit data to 4 bits/sample (4:1) using a different (faster) method than Microsoft ADPCM, and has different distortion characteristics, which can give better or worse results depending on the sample being compressed. As with Microsoft ADPCM, it's best to save to this format from 16-bit rather than 8-bit. This compression scheme can be a good alternative to MPEG; it provides reasonably fast decoding of 4:1 compression, and it degrades sample quality only slightly. The files produced using this format have an extension of .wav.

IFF The Amiga 8SVX format (which can have an extension of .iff or .svx) is an 8-bit mono format from the Commodore Amiga computer. A variety of sample rates are supported.

Image An Image is a block in the multitrack environment that does not singly represent a sound file, but rather acts as a shortcut. Images are simply pointers which, when encountered in a session, direct Adobe Audition to the same sound file on disk. For example, you may have a drum hit which occurs 30 times within a session. If these 30 instances are entered in the session as Images, the disk space required for the drum hits amounts to the same as for one instance. Keep in mind that because Images represent the same file, any alteration to one Image (like a cut, or transform) will affect all instances of the Image in the session. As an alternative, a copy of a block will create a separate sound file on disk; this consumes disk space, but will allow for separate editing.

Index Cue One of the four types of Adobe Audition Cues. Indexes are markers within a CD track. (Some CD players offer controls for cueing indexes.) Also, the time between the track marker that begins a track and the first index in that track will show up on the player as "negative time".

Impulse An impulse is the data by which every other sample in your waveform will be multiplied when using the Convolution effect (Effects > Special > Convolution). Impulses are like "amplitude maps". For example, if the impulse is a single sample of a full volume "tick", then the convolution of that impulse with any audio data would just be that audio data itself. Should the tick be at half volume, then the original audio data will be reproduced at half volume. If there are several ticks (such as one tick every 100 milliseconds), descending in amplitude over time, with each being half as loud as the previous tick, then the result of convolution with some audio will be that sound echoed with 100ms between each echo, and each echo at half the volume of the previous one.

Interpolate To estimate the values of data points between known data points. Interpolation is used in functions where new data must be generated from known data to fill in areas where values are unknown, such as the final error correction stage in the playing of CD audio.

Level (or VU) Meters Adobe Audition's Level Meters are found by default along the bottom of the main window, and are used to monitor the volumes of incoming and outgoing signals. The clip indicator to the right of the meters will light up, and remain lit, when levels exceed the maximum of 0dB. Clicking on the clipping indicator at any time will reset it. When displaying stereo audio or in a multitrack session, the top meter represents the left channel, and the bottom, the right.

Limiter Limiters reduce or “limit” input signals that exceed a specified threshold level so that the output does not increase in gain beyond that point. In other words, a limiter only allows the dynamic range at its input to increase up to a certain point, which is determined by the threshold setting. Beyond this level, as the input continues to increase in gain, the output level remains relatively constant and does not increase in volume.

Loop With regard to waveform material, a Loop is simply the repeating of a certain range, either indefinitely or for a certain number of repetitions. There are many looping functions in Adobe Audition, such as Loop Duplicate or loop “painting” in the multitrack environment for looping waveform blocks. If you insert a file that has loop information in its header, just grab the bottom right corner of the block and drag to loop it over time.

Microsoft ADPCM The Microsoft ADPCM format consists of 4-bits-per-channel compressed data, which provides 4:1 compression. Files saved in this format will automatically be expanded to 16 bits when loaded, regardless of their original resolution. For this reason, it’s best to save to this format from a 16-bit waveform rather than 8-bit, as the quality will be much greater. The file extension is .wav.

MIDI MIDI stands for Musical Instrument Digital Interface, and is a way of communicating performance information from one piece of software or hardware to another. This performance information can take the simple shape of a note instruction, or it can transmit detailed information on things such as timing or sound patch data. Windows provides a way of transmitting MIDI information internally between programs, plus you can transmit MIDI information into and out of your computer to and from external devices (such as a MIDI Keyboard) through the MIDI Port of a sound card, or other MIDI Interface device.

MIDI Time Code (MTC) MIDI Time Code (MTC) is a method of sending timing information between MIDI-capable devices. For instance, SMPTE, which is an audio signal, can be converted to MTC to sync to and control Adobe Audition’s transport from a device such as a VTR or tape deck.

MIDI Trigger A MIDI Trigger is simply a shortcut which can be called via a MIDI event (such as a Note On or Pitch Wheel). MIDI events can be sent from a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command.

Millisecond (ms) One thousandth of a second. (There are 1,000 milliseconds in a second.)

Miniplug A common name for 1/8-inch plugs and jacks, sometimes known as minijacks. Miniplug jacks are the most common interface for a sound card’s analog inputs and outputs.

Mixdown Mixdown is the process of combining the output of all enabled tracks (or selected tracks) in Multitrack View into a new stereo waveform. When doing a Mixdown, track properties such as Volume and Pan are reflected in the resulting waveform. Mixdown is generally the final operation, which is done when you’re happy with the way your session sounds. A Mixdown can also be used to create submixes, which are mixes of selected tracks (such as all of the drums). Submixes are useful for cleaning up your workspace.

Mixing Mixing is the process of combining multiple audio sources (or tracks) together for output as a single source. Output is generally in the form of a stereo pair of channels (a right and a left), though mixes may be directed to more than two channels for output (as in drums to one set of outputs, and everything else to another pair of outputs). These types of mixes are often referred to as submixes.

Mix Gauge The Mix Gauge is found beneath the Track Console in the Multitrack View. It's an indicator of the amount of background mixing that Adobe Audition has completed. Each time something changes in a session that alters the way it will play back, this bar will "empty" and fill back up as the mix is re-processed. The further this bar progresses to the right, the more complete the mix of your session. The bar becomes a bright color when fully mixed. You don't need to wait for the Mix Gauge to completely fill up before playing the session.

Mono Mono (derived from "Monophonic") means that there's only a single sound source, or the signal came from a single sound source.

MPEG A set of audio and video compression schemes created by the Motion Picture Experts Group.

MP3 MPEG Layer 3. A subset of the MPEG standard invented by the Fraunhofer institute that has become very popular due to the very high compression ratios (10 to 1 or better) with little audible loss. Adobe Audition has MP3 support built in, so there's no need to purchase an MP3 plug-in.

mu-Law mu-Law (or CCITT standard G.711) is an audio compression scheme and international standard in telephony applications. mu-Law is very similar to A-Law, a variation of mu-Law found in European systems. This encoding format compresses original 16-bit audio down to 8 bits (for a 2:1 compression ratio) with a dynamic range of about 13 bits. Thus, mu-Law encoded waveforms have a higher s/n ratio than 8-bit PCM, but at the price of a bit more distortion than the original 16-bit audio. The quality is higher than you would get with 4-bit ADPCM formats. Encoding and decoding is rather fast and generally widely supported.

Noise Gate A Noise Gate is a special type of expander that can be used to reduce or eliminate noise below a threshold level. It does this by heavily attenuating signals with levels that fall below the threshold. A noise gate is often used to totally cut off the signal level during a musical pause so as not to pass background noise. It can also be used to silence the pauses in speech.

Noise Shaping This technique is used to shift the frequency of dithering noise to minimize its audibility in the wave file.

Nondestructive Editing Nondestructive edits and changes are those which don't alter the sound file on disk in any way. For example, in nondestructive editing, a volume change would not actually alter the amplitude of the waveform, but would be an instruction to "get louder at this point", which is then acted out upon playback. The Adobe Audition Multitrack View is a nondestructive editing environment.

Normalize Normalization adjusts the highest peak of the waveform to a certain percentage, thereby raising or lowering all other peaks accordingly. In most cases, this is 100%, but Adobe Audition allows you to normalize a waveform to any percentage.

Nyquist Frequency The Nyquist Frequency (also called Nyquist Rate) is one-half of the current sampling rate. For example, the Nyquist Frequency for 44,100 Hz audio is 22,050 Hz. This frequency designates the highest reproducible frequency for that sample rate. Thus, in order to reproduce a signal with an 11 kHz frequency range, you would need to select a sample rate of at least 22 kHz. In nearly all analog-to-digital converters, frequencies that exceed the Nyquist Frequency are filtered out before reaching the actual analog-to-digital conversion process using analog bandpass circuitry. If not filtered out, frequencies above the Nyquist Frequency would produce aliasing distortion. It's best to record at higher sample rates and convert down if needed.

Organizer Window The part of the Adobe Audition interface that lets you see a tabbed list of open files, effects, and favorites. The Organizer Window isn't shown by default. Activate it by pressing the F9 key or by selecting View > Show Organizer Window.

PCM Pulse Code Modulation. PCM is the standard method of digitally encoding audio. It's the basic uncompressed data format used in file types such as Windows .wav and Apple AIFF.

Peak Files Peak Files are files with the extension .pk that enable Adobe Audition to load, save, and redraw audio files more quickly than it could do without them. You can safely delete Peak Files, or turn their creation off altogether in the Display tab of Options > Settings. However, keep in mind that without Peak Files, larger audio files will take longer to reload.

Phase As a wave vibrates, it cycles through from its maximum to minimum amplitude, and then back to its maximum. The phase is the position of the wave on this cycle. When you hear two sounds simultaneously, their amplitudes are summed together to form the total sound we hear. Two sounds will cancel each other out when one's high points coincide with the other's low points. These two waves are said to be out of phase.

Pink Noise Pink noise has a spectral frequency of $1/f$ and is found mostly in nature. It's the most natural sounding of the noises. By equalizing pink noise, you can simulate rainfall, waterfalls, wind, a rushing river, and other natural sounds. Pink noise falls exactly between brown and white noise (which is why some people used to call it tan noise, but pink was more appealing). Pink noise has a fractal-like nature when viewed. When zoomed in, the pattern looks identical to the way it appears when zoomed out, except at a lower amplitude.

Play List The Play List is an arrangement of Cue List entries that you can play back in any order and loop a specified number of times in nondestructive fashion. Adobe Audition saves the Play List in the .wav file format header.

Plug-in A Plug-in is a software component that can be added to another piece of software to increase its functionality. For example, Adobe Audition supports third-party DirectX audio Plug-ins, which seamlessly integrate into Adobe Audition's interface.

Portion Bar Vertical and horizontal bars that are used for zooming and scrolling in both Edit and Multitrack Views. Their name comes from their ability to allow you to see just a portion of the current waveform or session.

Preset Most functions in Adobe Audition supports Presets. A Preset is a group of function settings saved under a particular name for later recall. Functions that support Presets will have a list where you can click on a Preset name to recall the settings, as well as an area for creating and naming new Presets, as well as a way for removing unwanted ones.

Preview Many functions in Adobe Audition support real-time Preview, which means that changes to the function settings may be monitored while they're being made, from within the function dialog. Those that offer Preview will have a Preview button in the function's dialog. The Preview quality is dependent upon your system's performance.

Punch In Punch In is a recording method used to record "into" a certain region of an existing waveform. Punching is usually used to replace an undesirable section within a longer audio segment. Adobe Audition allows for multiple "takes" when Punching In; you can continually record over the original material and afterward choose between the takes for the best performance, or return to the original data. Punch In is only available in the multitrack environment.

Quantization Quantization takes place as a part of the analog to digital conversion process as the values of an analog wave are turned into steps. Quantization Noise naturally occurs during this time, caused by sample levels being changed to conform to standard Quantization levels. This noise is heard more often in sounds recorded at low bit resolutions, especially at lower amplitude.

RAM Random Access Memory. The memory required to load and run programs. This is what you refer to when you say that your computer has 128 megabytes of memory.

RCA Cable This type of cable (sometimes called a phono cable) features RCA plugs or jacks at either end. RCA cables are normally used to connect stereo system components, like receivers, CD players, and cassette decks.

Real-Time In computer-based audio, Real-Time generally refers to an operation or function's ability to act, or react, on demand. That is, there is no "wait time" for processing to occur, but rather things are seemingly done in an instantaneous fashion (though this is never really the case when dealing with digital audio, as there is always some amount of processing done). Adobe Audition provides Real-Time effects functionality with most of its effects in the Multitrack View, and a Real-Time preview of effects before they are applied in the Edit View.

Resample To recalculate a sound file's samples using a different rate than which the file was originally recorded.

Reverb The persistence of a sound after its source has stopped emitting it. (Reverb is a shortened version of the word Reverberation.) Many discrete echoes arrive at the ear so closely spaced in time that the ear can't separate them. Adobe Audition offers three Reverb Effects (Quick Verb, Reverb, and Full Reverb), each with many settings.

Rip Ripping is the process of digitally extracting the audio from a compact disc and turning it into a waveform that's usually an identical copy of the original CD track(s). Most newer CD-ROM, CD-R, and CD-RW drives support digital audio extraction.

ROM Read Only Memory. Memory that cannot be written to after it is constructed. Most computers have a small amount of ROM that holds software required to boot the computer. Unlike RAM, information in ROM stays when the computer is turned off.

Ruler The numbered controls to the right and below the Wave and Session displays. The Horizontal Ruler can measure time, CD frames, time code, samples, and bars and beats in both Edit and Multitrack Views, and includes scrolling and zooming functionality. In the Edit View, the Vertical Ruler measures sample values, normalized values, decibels, or percentage. In Multitrack View, use the Vertical Ruler to scroll through tracks, as well as to control the number of tracks displayed on-screen at once.

S/N Ratio Signal-to-Noise Ratio (S/N Ratio) is the difference in level between the average signal level and the average level of the noise floor.

SAM The 8-bit signed wave file format (.sam) is popular for building MOD files, since audio in MOD files is 8-bit signed. Many MOD editors allow samples to be inserted from files, or exported to files in this format. 8-bit signed raw format data with the .sam extension is assumed to be 8-bit signed raw data with no header. The sample rate is assumed to be 22050Hz, but the actual sample rate can be changed once the file is loaded using Edit > Convert Sample Type.

Sample In digital audio, a signal isn't captured in a complete continuous fashion, as in traditional recording. To digitize a sound, a number of snapshots, or samples, are taken every second (this number is a product of the sampling rate used) and are changed into numeric representations. A sample is a single snapshot of the sound, if you will. The term sample can also refer to the digitally recorded sound itself, though this nomenclature tends to be used more when referring to digitized sounds used by hardware samplers than in computer-based audio editing.

Sample Rate The sample rate determines the number of times per second to take a snapshot of the audio. Higher sample rates produce higher-resolution audio, but at the expense of requiring more disk space. Frequencies of up to one-half the sample rate can be produced effectively. So, to reproduce a frequency of 10Khz, a sample rate of at least 20Khz must be chosen. (See Nyquist Frequency.)

Sampler A device that records and plays digital sounds (samples) many times with features to edit and store the samples.

SCSI Small Computer Systems Interface. Pronounced "skuzzy", SCSI is a standard for attaching components (such as hard drives and CD-ROMs) to a computer. It allows for multiple devices to be daisy-chained to the same controller, and for very fast data transmission.

Sequencer A programmable electronic device which can play a stepped order (sequence) of musical events, such as samples, pitches, and rests.

Session A Session in Adobe Audition is a multitrack project. Actually, "session" can refer to the project itself (as in "your session") or to the act of working on the project (as in "during your session"). A Session is stored as a file on your hard disk with the extension .ses, and contains the details of the project, such as what tracks contain what audio files, volume levels, and envelope information, among other things. Session files don't actually store the audio data in them; instead they point Adobe Audition to where it can find and load the sound files used in the project.

Session Display The workspace where audio data shows up and can be manipulated in Adobe Audition's Multitrack View.

SMP Audio files with an .smp extension are used by Turtle Beach's SampleVision program. The format supports only mono 16-bit audio. If your data is in a different format, you'll be asked to convert it before saving. SMP format also supports loop points, which can be edited using Adobe Audition's Cue List.

Shortcut Shortcuts are used to speed up the editing process by allowing you to execute almost any command in Adobe Audition from your computer keyboard. For example, instead of using your mouse to go to the Edit menu and selecting Cut to remove the highlighted portion of a waveform, you can simply press Ctrl+X. In Adobe Audition, you can execute commands from a MIDI keyboard, a sequencer, or any other device capable of issuing a MIDI command. This type of Shortcut is referred to as a MIDI Trigger. An example of this would be assigning the Play command in Adobe Audition to the C4 note on your MIDI keyboard.

SMPTE Start Offset The offset to the start of a multitrack session. The Start Offset determines when playback and recording will trigger while Adobe Audition is syncing to SMPTE time code.

SMPTE Time Code SMPTE (Society of Motion Picture and Television Engineers) time code is a timing reference used to synchronize two devices to the same clock. SMPTE time is broken down into hours, minutes, seconds, and frames.

SND Waves with this file extension are usually Apple AIFF or Next/Sun files. See AIFF or AU.

Sound Card A device in your PC that allows your computer to play and record audio.

Stereo The reproduction or recording of two or more channels, where the position of discrete left and right sounds can be determined.

Stripe To copy SMPTE time code to a single track of a multitrack tape. The SMPTE is then used to reference time on the other tracks for purposes of syncing those tracks to other devices.

SVX The Amiga 8SVX format (which can have an extension of .iff or .svx) is an 8-bit mono format from the Commodore Amiga computer. A variety of sample rates are supported.

Tempo Tempo is the rate at which music moves, normally measured in BPM (Beats Per Minute).

Time Code An audio or digital signal designed to synchronize time between multiple devices. The most common forms are SMPTE and MIDI time code.

Track Tracks in Adobe Audition are where audio is positioned and arranged in a session. A track can contain one or more blocks, and can be configured with its own volume, pan, track name, EQ, effects, and input/output assignments. Tracks are found in the Session Display of the Multitrack View, and each session can have up to 128 tracks.

Track Cue One of the four types of Adobe Audition Cues. They indicate a split in tracks for an audio compact disc.

Track Controls The Track Controls appear to the left of Adobe Audition's Multitrack View's Session Window. There you'll find controls for viewing and altering track properties such as pan, EQ, and level; adding real-time effects; assigning multiple tracks to a bus; and adjusting the input/output routing (sound card assignment) of each track.

Track Effects This term refers to Adobe Audition's non-bus real-time effects. They are accessed through the Track Effects Rack dialog.

Track Effects Configuration This dialog shows tabs for each effect on a given track. It's where you actually set the properties for that effect, such as the delay length.

Track Effects Rack The Track Effects Rack dialog is the initial dialog used for deciding which real-time effect(s) you wish to add to or remove from a track.

TXT See ASCII Text Data.

Unity Gain When a device is set to Unity Gain, it means its output will be at its specified line level. Because there are different line levels (-10 dBV and +4 dBu being the most common), a device could take a voltage of one level, turn it up to a different level, and pass that as its unity gain.

VOC This is the Sound Blaster and Sound Blaster Pro voice file format. Adobe Audition supports both old and new style .voc files. The older .voc format only supports 8-bit files. Mono files can have a sample rate of up to 44.1 kHz, and stereo up to 22 kHz. The newer type of .voc file supports both 8-bit and 16-bit audio. VOC files can contain information for looping and silence. If the file you're loading contains loops and silence blocks, they'll be expanded while loading.

Vocoder An Adobe Audition Multitrack Effect that will modulate one signal with another.

VOX The Dialogic ADPCM format is commonly found in telephony applications, and has been optimized for low sample rate voice. It will only save mono 16-bit audio, and like other ADPCM formats, it compresses to 4 bits/sample (for a 4:1 ratio). This format has no header, so any file format with the extension .vox will be assumed to be in this format.

WAV Files with the .wav extension typically are Microsoft Windows uncompressed PCM audio files. However, Adobe Audition also supports compressed variations of .wav files such as A-Law, mu-Law, ACM, DVI/IMA ADPCM, and Microsoft ADPCM.

Wave File The term Wave File usually refers to any audio file format which contains soundwave data. Windows users sometimes just think of a Wave File as an audio file with the .wav extension, but a Wave File could very well be an .au, .aif, or .mp3 file.

Waveform The term Waveform is used to describe the visual representation of a signal (amplitude over time) displayed in Adobe Audition. Waveform can also be used to refer to the signal itself, as in “listen to the waveform”. In this case, the term usage is referencing sound waves, and is being used in an acoustical sense.

Waveform Block Any waveform entry in the Multitrack environment. A Waveform Block may be an actual representation of a waveform, an Image (a pointer to another waveform block), a copy of another waveform, a spliced section of another Waveform Block, etc. Any edits made to a Waveform Block are nondestructive in nature. You can right-click on a Waveform Block for a menu containing editing commands, or double click on it to “jump into” its source in the Edit View.

Wave Display The Wave Display is the part of Adobe Audition’s Edit View in which you view and manipulate audio data. By default this audio material is in the shape of a waveform, but it can be displayed as spectral data by choosing the View > Spectral View option.

Wet Used to describe an audio signal that has had signal processing (such as reverb) applied; the opposite of Dry.

White Noise White noise has a spectral frequency of 1. In other words, equal proportions of all frequencies are present. Because the human ear is more susceptible to high frequencies, white noise sounds very hissy. Adobe Audition generates white noise by choosing random values for each sample.

Windows WAV The Microsoft Windows WAV format supports both mono and stereo files at a variety of resolutions and sample rates. It follows the RIFF (Resource Information File Format) specification, and allows for extra user information to be embedded and saved with the wave file. WAV files contain PCM coded audio – which is pure, uncompressed Pulse Code Modulation formatted data.

Zero Crossing A place in time where the wave crosses the zero amplitude line. By placing edit points at zero-crossing points, it’s much less likely that the edit will create a pop or click after the edit since there won’t be a sudden jump in amplitude after the edit.

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