ADOBE AUDITION 1.5

FOR AUDIO CONVERSION AND MANIPULATION

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Abstract

Saat ini banyak aplikasi di pasar yang digunakan untuk mengolah data baik itu data suara maupun video. Salah satu aplikasi yang cukup populer saat ini adalah Adobe Audition sebuah aplikasi yang diproduksi oleh sebuah perusahaan dengan nama Adobe Corporation. Aplikasi ini dapat digunakan untuk membuat sebuah data suara baik langsung dari sumbernya yang direkam langsung maupun diambil dari sebuah sumber berupa CD musik ataupun sebuah suara yang ada pada sebuah data video. Aplikasi ini juga memiliki efek-efek suara untuk manipulasi suara yang biasanya dilakukan dengan menggunakan prosesor suara secara piranti keras. Jurnal ini berisi tentang kelebihan dan kekurangan aplikasi tersebut serta juga langkah-langkah yang dibutuhkan untuk melakukan pembuatan suara serta manipulasi terhadap sebuah data suara.

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Introduction

Before digital technology is widely used, the analog technology was the best common way to process any manipulation to sound or audio. Audio signal generated by an instrument or any musical device is possible for manipulation. To get a new kind of audio signal, it requires passing any filter that will process the audio to new form of audio signal. Many effects may be added to an audio signal such as reverberation, delay, chorus, flanging, compression, etc.

A device required to transform sound is called signal processor device. The analogy to any altered sound is the same way as adding any spices in cooking or any make-up in the theater. The important thing to consider is to have a good understanding of the essential characteristics of the audio signal and what a signal processor can do to it. To process the audio signal while using a lot of instruments or devices is not economic. Having two forms of new audio signal may need at least two devices while processing the signal.

One of the software for the audio process manipulation is Adobe Audition, formerly Cool Edit. It offers many features to manipulate an audio signal. There is no need to buy any signal processor device to manipulate a sound instead of just using this software, which can be installed in a Personal Computer (PC). The version of Adobe Audition used in this journal is Adobe Audition 1.5.

Brief History of Adobe Audition

A company called Syntrillium released audio software before year 2000 called Cool Edit 2000 and Cool Edit Pro; Figure 1 shows the logo of the software. They were successful audio software in the market. In May 2003, another big company, Adobe Corporation, bought the Syntrillium and altered the name of the software to Adobe Audition with its first version, version 1.0. Figure 2 shows the new logo of the Adobe Audition software.

Figure 1. Logo of the Syntrillium software

Figure 2. New logo of the software

Things to Make Sure Before Starting

A PC commonly is equipped with an audio card in order to process any audio data either to have it as an output or converted to another audio variation. To start audio manipulation in a PC the important things to ensure is this audio card is plugged already and detected by the Operating System (OS) used for the PC. The minimum requirements for the system are Intel® Pentium® III or 4 or Intel Centrino™ processor, Microsoft® Windows® XP Professional or Home Edition with Service Pack 2, 512MB of RAM (1GB recommended), 1,024x768 display (1,280x1,024 recommended), Sound card with DirectSound, DVD-ROM drive, CD-RW drive for audio CD creation, and Speakers or headphones [3].

The minimum specs mentioned above are things to make sure before starting audio process. A PC may have more than one audio card. Adobe Audition supports selection of any available audio card plugged to the computer. This may be done by setting the device used for manipulating the audio. The source of the audio may be directly from any instrument or device connected to the audio card either through the line-in or microphone. The larger the size of the memory used the quicker the application processes any given task instead of just the minimum memory size.

Benefit of Using the Application

Multimedia is the seamless integration of text, sound, images of all kinds and control software within a single digital information environment [2]. This application is classified as a multimedia application because it is involved in processing combination of the sound and video and the moving pictures or images. Before changing its name to adobe audition, it can only deal with the task of audio but in this new face, it integrates with popular audio technology such as ReWire and Virtual Studio Technology (VST) [4] and video applications like Adobe Premiere Pro and Adobe After Effects.

Adobe Audition supports up to 32-bit files and sample rates up to 10 MHz as long as it is equipped with 32-bit internal processing, Its other feature such as powerful effects, restoration, and pitch correction tools enhanced the new audio. It also puts all the tools needed to get work done quickly and efficiently. An intuitive interface shorten the processing time such as editing, mixing, and streamlining the audio workflow through CD burning. Even though integration of audio and video exists in this application, it is mainly focused on the audio.

In early technology to manipulate the audio as a part of multimedia component, tape had important role in the nature of the multimedia systems. New form of multimedia emerges when the media moves from the outside of the computer to inside. There were two major changes, which are digitization of the media and the drastic price reduction of digital storage systems [2].

Inside the application the necessity of usage of manual devices or components in minimized. All tasks done by it involve digital process. It requires no external device for adding any effect to the audio, all fully digital.

Inside the Adobe Audition

To run the application click on the link to the program located in Start-All Programs as seen in figure 3.

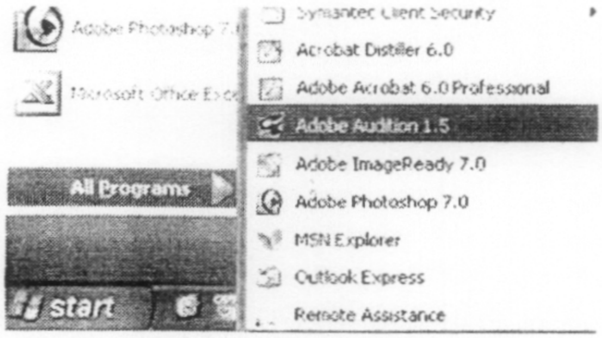


Figure 3. The link to run the application

Once it is opened the main working area is displayed. It is divided into three main work areas: Edit View, Multitrack View, and CD Project View. This intended to help focusing on the major tasks of editing audio files, mixing sessions, and burning CDs. Figure 4 shows the main working area.

All three views have a similar user interface, including the following components: Menus, Toolbars, Windows, and Display Window. Before starting any project, setting up the application is the first to do to ensure the good audio output. The setup tasks can be divided into some categories, which are setting up the devices being used, customizing internal preferences, and managing the size of temporary files. Computer may have multiple sound cards, or a single card that has multiple inputs and outputs. To get the right component in the right track devices need to be specified either for purpose of playback or recording. It also may need setting up other device such as MIDI devices, external controllers, and ReWire connections.

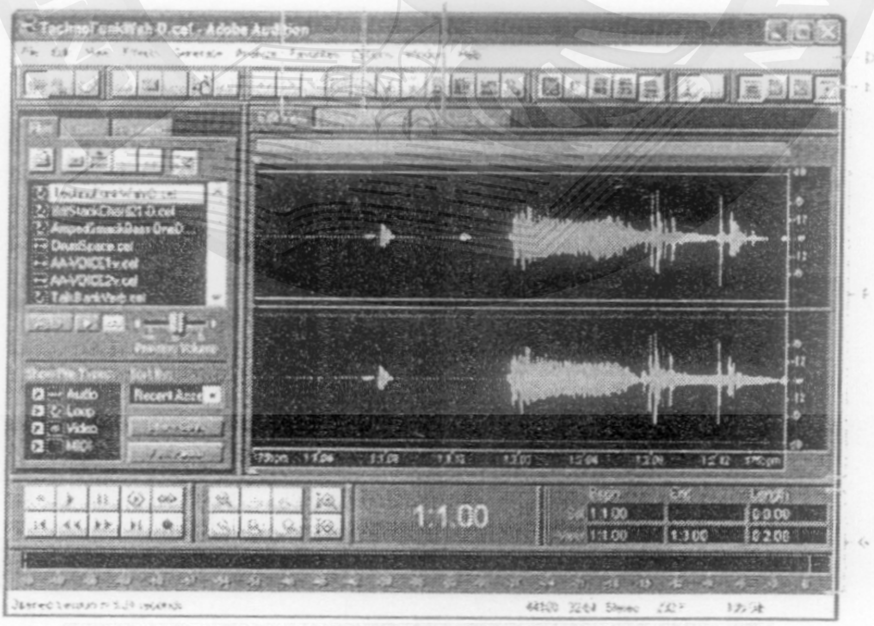


Figure 4.
Application Working Area A. Edit View tab B.
Multitrack View tab C. CD Project View tab D. menus
E. toolbars F. display window G. various windows

The size of temporary files is limited only by the amount of disk space that is available. It is recommended to have a dedicated amount of space in the hard disk separated from the main hard disk used for the main task of operating systems. However, shall any problems emerge while working with very large files (or when you have many files open at the same time) for example running the disk space low, deleting the unused files will be the solution.

There are wide range of devices may be used with Adobe Audition. Sound card inputs bring audio signals into Adobe Audition through sources such as microphones, tape decks, and digital effects units. They also monitor audio signals through sources such as speakers and headphones. MIDI ports connect Adobe Audition to MIDI keyboards and synthesizers. You can also synchronize Adobe Audition with ReWire applications and hardware or software components that support Society of Motion Picture and Television Engineers (SMPTE) [5] / MIDI Time Code (MTC) time code.

To designate the devices the Device Order dialog box is used. When working in Edit View, one stereo output device can designate to use for playback and one stereo input device for recording. When working in Multitrack View, different input and output devices can be assigned to each audio track. If audio system includes MIDI devices, they can also be designated for MIDI input and output devices to use. For example, a MIDI keyboard can be designated to use for triggering commands and a MIDI synthesizer channel for playback.

Steps to designate the devices to use are choosing Options > Device Order, click the tab for the type of device to designate: Playback, Recording, MIDI Output, or MIDI Input. Move the devices to use into the Multitrack Device Preference Order list by selecting devices in the unused list and clicking Use. Remove unwanted the devices by selecting devices in the Multitrack Device Preference Order list and clicking Remove. This application may support up to 16 stereo devices or 32 mono devices in the Multitrack Device Preference Order list. See figure 5.

Figure 5. Menu to select the device(s)

Designate the device in Edit View by selecting the device and clicking Use in EV. [EV] appears after the device name. Adjust the order of devices for use in Multitrack View by selecting a device and clicking Move Up or Move Down. The first device in the list is the default device. This means that, by default, the first playback device is assigned as the output for all audio tracks in a session and the first recording device is assigned as the input for all audio tracks. Likewise, the first MIDI Out device is assigned as the output for all MIDI tracks. However, you can easily reassign the devices for a track. (See Using the Track Properties window and Importing and mapping MIDI files.) If desired, click a different tab to set up ordering for another type of device. Click OK to finish.

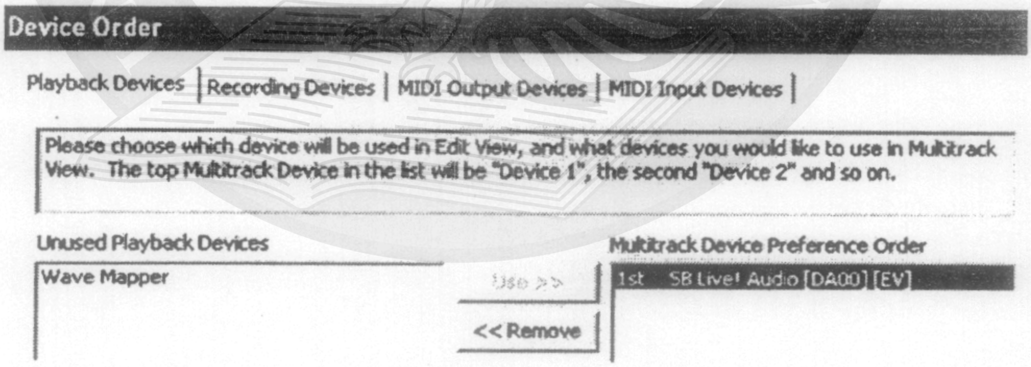


Figure 6. Device Order Window

Through Device Properties dialog box Adobe Audition's parameters for playing back waveforms is configured. The properties for each output either using multiple sound cards or a single card that has multiple audio outputs is set in this dialog box. Steps to set are choosing Options, Device Properties, click the Wave Out tab and selecting a device from the list at the top of the dialog box.

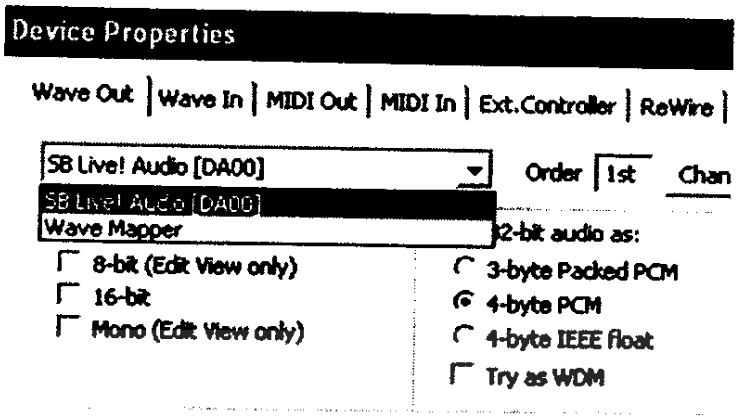


Figure 7. Device Properties Window

The capabilities of the selected output device are shown in the Supported Formats table. Shown in table 1. A Yes or No indicates 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.

	8-bit		16-bit	
Rate	Mono	Stereo	Mono	Stereo
8K	Yes	Yes	Yes	Yes
11K	Yes	Yes	Yes	Yes
16K	Yes	Yes	Yes	Yes
22K	Yes	Yes	Yes	Yes
32K	Yes	Yes	Yes	Yes
44K	Yes	Yes	Yes	Yes
48K	Yes	Yes	Yes	Yes
96K	Yes	Yes	Yes	Yes
24-bit (3-byte packed) supported				
32-bit (4-byte PCM) supported				
32-bit (IEEE Float) supported				
WDM "Extensible' Supported				

Table 1. Supported Formats

Order displays the order of the device for use in Multitrack View. To change the order of the devices click change button. The application will use the device to play waveforms in Edit View if the option in figure 6 is ticked.

Figure 8. Option to display device in Edit View

The option limit the playback (see figure 9) is used to compensate for limitations imposed by hardware. For example, if the sound card desn not handle 32-bit audio correctly, use this to limit the playback of 32-bit files to either 16-bit or 8-bit. The option Send 32-bit Audio As specifies how Adobe Audition sends 32-bit audio data to the output device. (See figure 9) This option is unavailable when selecting a Limit Playback To option. If the output device supports it, sending 32-bit audio as 3-byte Packed PCM, 4-byte PCM, or 4-byte IEEE float is possible to do.

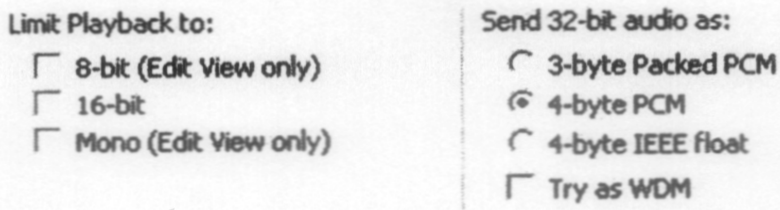
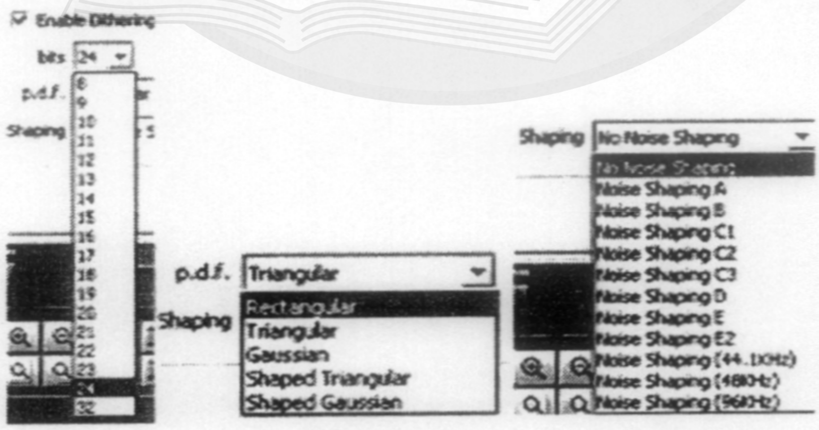


Figure 9. Options to suit the hardware

Enabling Dithering activates dithering when playing back audio at a limited bit depth. See figure 10a. Deselecting this option will truncates the audio data instead. This means that bits that aren't used are simply chopped off and discarded. Enabling dithering is recommended when working with audio files that have a higher bit depth than sound card supports. Its options are available when dithering is enabled. Maximum bits to specifies the number of bits to dither to is 32. If using a 20-bit sound card, for example, and requiring dithering to 20 bits since any more bits will not be used by the card. Even for 16-bit-only sound cards, choosing to dither to 16-bit will improve the quality when playing back 32-bit audio. probability distribution function (p.d.f) (figure 10b) controls how the dithered noise is distributed away from the original audio sample value. Usually one of the Triangular p.d.f. Functions is a wise choice, because it gives the best trade off between Signal-to-Noise Ratio (SNR) , distortion, and noise modulation. Shaping specifies a noise-shaping curve for moving noise to different frequencies (figure 10c). To obtain no shaping just choose no noise shaping from the available options.



a. b. c.

Figure 10. a. Bit Dithering b. p.d.f options c. Shapping

Starting a Project

To start a new project, choose the menu File and New Sesssion (see figure 11) or simple press Ctrl+N on the keyboard.

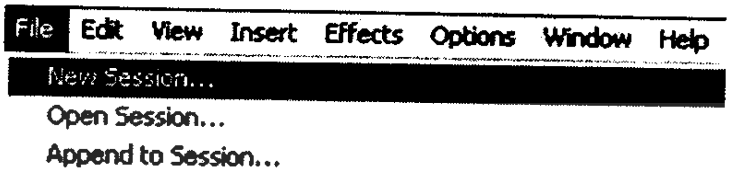


Figure 11. Starting a new project

The window for sample rate option will display (figure 12). Sample Rate determines how many frequencies can be encoded in the audio signal. Higher sampling rates mean wider bandwidth.

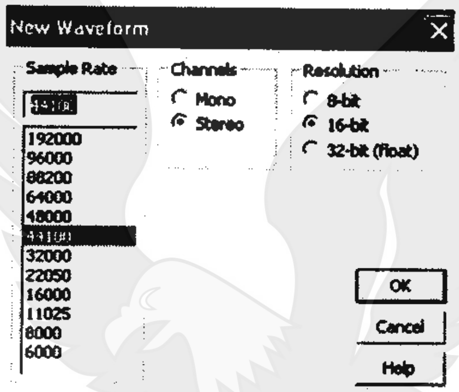


Figure 12. Sample rate of the audio

During the sampling process, an incoming analog signal is sampled at discrete time intervals. Each interval of analog signal is momentarily observed, and thus, each represents a specific, measurable voltage level. A mathematical conversion generates a digital series of numbers that represent the signal level at that particular point in time. The generated data can be digitally stored or processed.

The sample rate is the number of samples (or snapshots) that are taken of an audio signal per second. For example, a sample rate of 44,100 Hz means that 44,100 samples are taken per second. Since sampling is tied directly to the component of time, a system's sample rate determines a system's overall bandwidth—in other words, how many frequencies can be encoded within the audio signal. Higher sample rates generally yield a better quality waveform. The most common sample rates for digital audio editing are as follows:

- 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

To create output that can be played using a common CD player choose the CD Quality sample rate which is 44,1KHz.

Channels (figure 10) determine if the waveform is mono or stereo. Select Mono to create a waveform with just one channel of audio information. This option works well for a voice-only recording. Select Stereo to create a two-channel waveform with separate right and left channels. This option is usually best for a music recording because they contain twice as much data; stereo waveforms consume twice the storage space of mono waveforms.

Resolution (figure 10) determines the number of unique amplitude levels Adobe Audition can use to represent a sound. The 32-bit level is best while working in Adobe Audition, and you convert down for output if necessary. Older sound cards might not be able to play 32-bit files properly. To check the capabilities of sound card, choose Options > Device Properties. If sound card doesn't support 32-bit files, convert the files to a lower bit rate (such as 16-bit) for playback.

Some options are available to open files in Edit View (see figure 13).

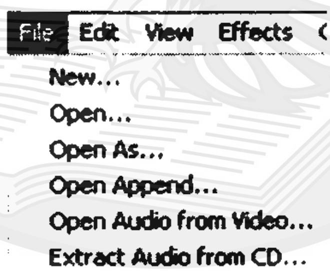


Figure 13. Open File Options

To create a new audio file, choose the menu File and New, if the audio file exists already previously the other options may be chosen. Recording audio can be done from a microphone or any signal plugged into the Line In port of a sound card. Adjustment of the input signal is important to obtain the optimum recording and signal-to-noise levels.

Adobe Audition doesn't directly control a sound card's record levels (input gain) and playback levels (output volume). Adjusting levels is necessary if recordings are too quiet (causing unwanted noise), too loud (leading to clipped, distorted sound), or not audible when played in Adobe Audition. Just open the Audio Control for Windows (as seen in figure 14)

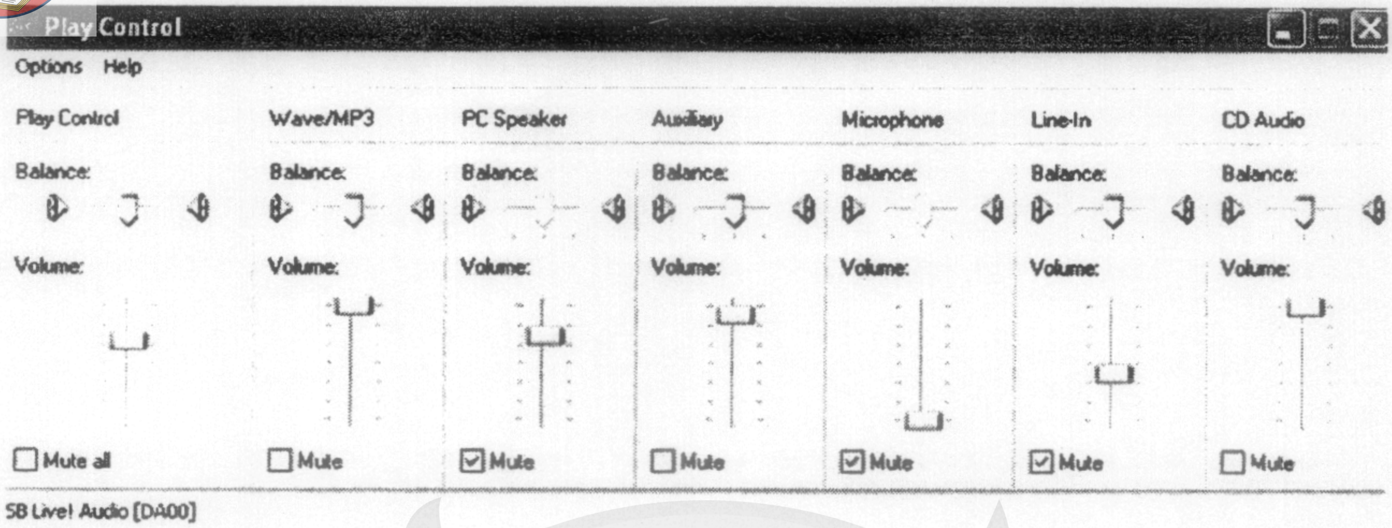


Figure 14. Windows Audio Control

By default, Adobe Audition displays waveforms in real time while recording. However, if the recorded audio is choppy, deselect Live Update During Recording in the General tab of the Settings dialog box. Once the source of the audio is plugged to either the microphone or line-in port of the sound card, the recording process can be done by clicking the record button (figure 15), button with red dot. To stop recording is simply by clicking the stop button, button with green square in the middle.

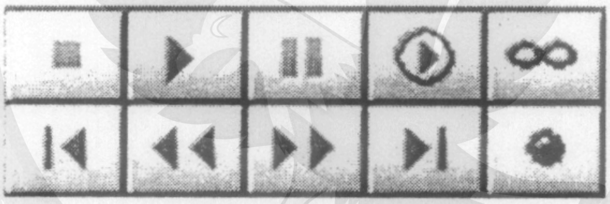


Figure 15. Operation Button

The recorded signal is will be displayed directly after clicking the stop button or the record button as seen in figure 16.

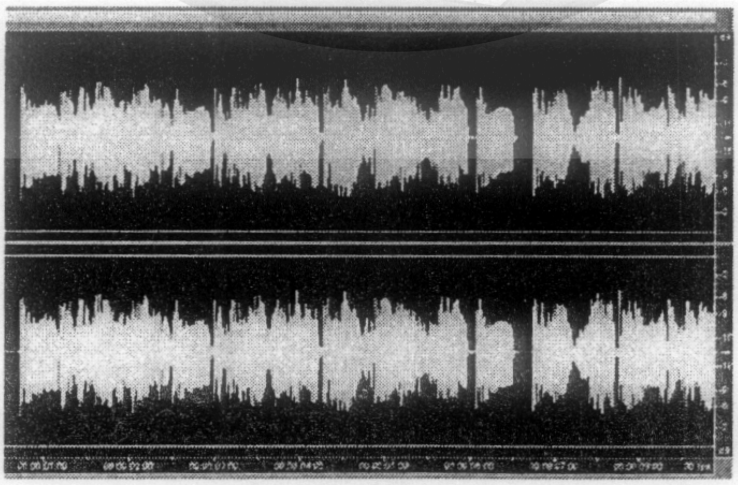


Figure 16. The recorded audio signal

The highest limit of the audio or amplitude is shown in the graphics. It is important to ensure that the level of the audio recorded is not beyond the top line. If a mono type audio is recorded then only one graphic is dislayed. Figure 17 shows a bad example of the recorded audio. The level of the source is too high so the adjustment of the source is needed.

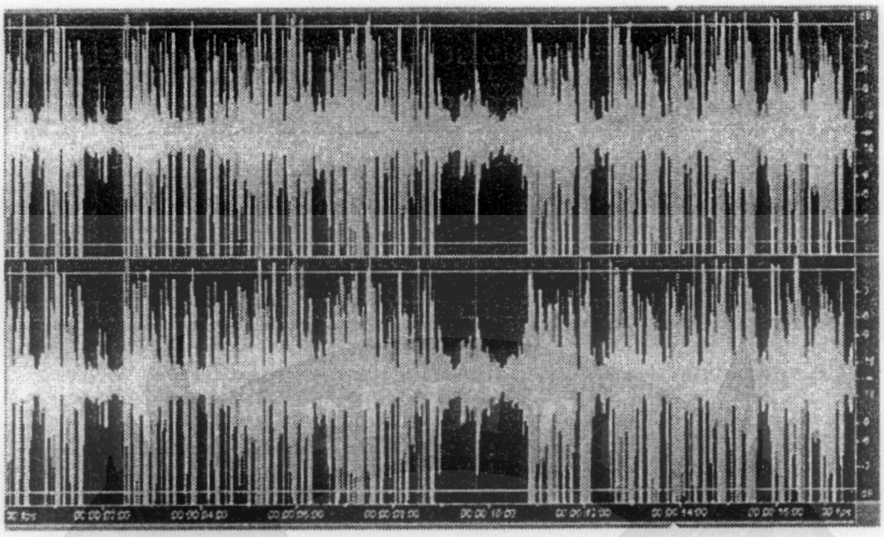


Figure 17. Bad Audio Source Level

Once the recording phase is finished, the data can be save to the default format of audio standard used by Windows, WAV. To save data to a file select the File and Save AS menu as seen in figure 18.

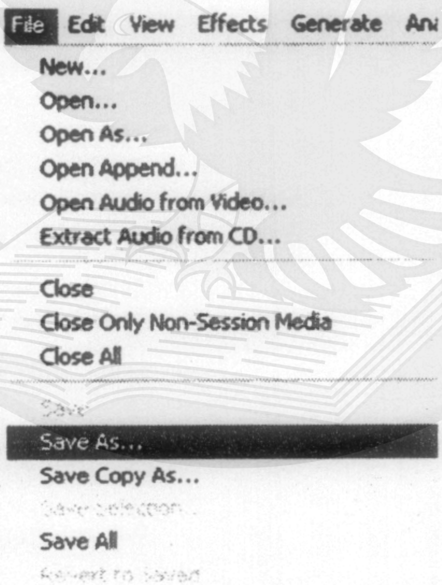


Figure 18. Save the Audio Data

Next window will appear to give option where the file need to save to (figure 19). Select the place to save the file in the option Save In. Right side of the window shows available free space of each storage attached to the computer, and the default format of the audio file is WAV. Click the Save Button to finish the saving process. Once the audio file exists the other Open Menu (figure 13) options may be selected. It is also possible to get the audio from any audio CD or any audio attached to a video file by selecting the two open menus from the button of the Open Menu, Open Audio from Video or Extract Audio from CD.

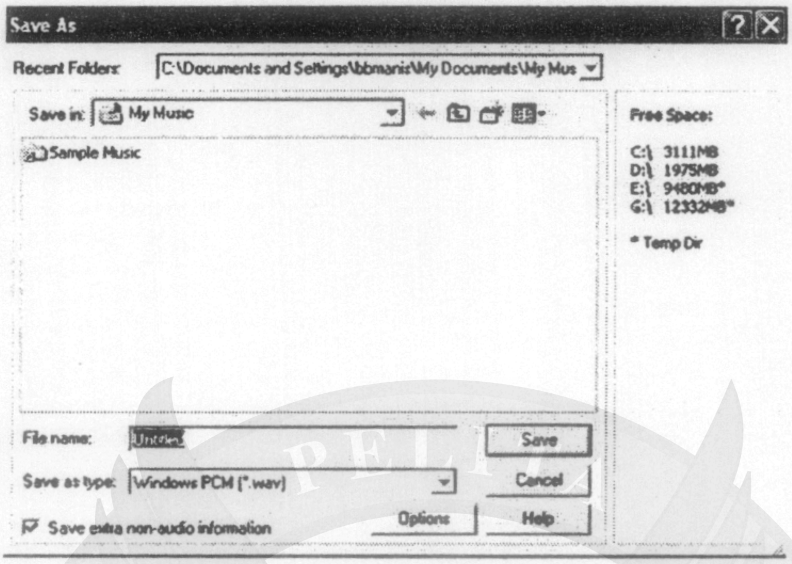


Figure 19. Save As Windows

Converting Audio

The Audio can be converted either by its sample rate or the end format of the audio file and it is also possible to convert the audio to its reverse version. A file's sample type determines its sample rate and bit depth, as well as the channel format (whether the waveform is mono or stereo). Converting the sample type to change any of these attributes is possible. When converting the sample type of a file, Adobe Audition directly processes the samples within the file, or resamples the data, so that the audio retains the same pitch and duration as the original file.

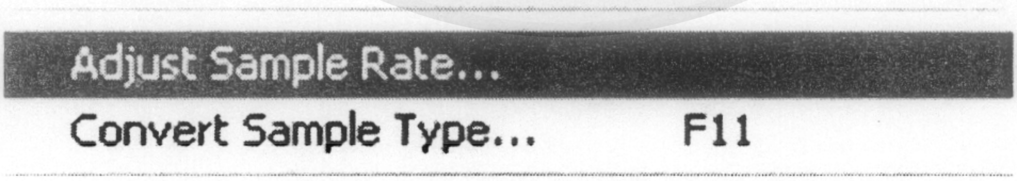


Figure 20. Menu to convert the audio

To convert an audio or a current processed audio file, find the menu at the bottom of the Edit Menu. To only adjust the sample rate select Adjust Sample Rate, but for the converting the sample including its channel, resolution and dithering feature, select the lowest menu of the Edit Menu.

Converting the format of the audio is also possible to meet any requirement. For example, to put the audio file for a website, the WAV format is absolutely is not the option to choose. The available bandwidth of data to transfer data across the network especially internet is the limitation to choose the format. Many other platforms are also available such as Macintosh and Unix environment. They also use different for of the audio. To accommodate this, the conversion is a must. The most popular audio type used by most people is MPEG Audio Layer 3 (MP3) format. MP 3, more commonly referred to as MP3, is a popular digital audio encoding and lossy compression format invented and standardized in 1991 by a team of engineers working in the framework of the ISO/IEC MPEG audio committee under the chairmanship of Professor Hans Musmann (University of Hannover, Germany). It was designed to greatly reduce the amount of data required to represent audio, yet still sound like a faithful reproduction of the original uncompressed audio to most listeners. In popular usage, MP3 also refers to files of sound or music recordings stored in the MP3 format on computers. [6].

MP3 is a lossy format; it is able to provide a number of different options for its "bit rate"—that is, the number of bits of encoded data that are used to represent each second of audio. Typically rates chosen are between 128 and 256 kilobit per second. By contrast, uncompressed audio as stored on a compact disc has a bit rate of about 1400 kbit/s. MP3 files encoded with a lower bit rate will generally play back at a lower quality. With too low a bit rate, "compression artifacts" (i.e., sounds that were not present in the original recording) may appear in the reproduction. A good demonstration of compression artifacts is provided by the sound of applause: it is hard to compress because it is random, therefore the failings of the encoder are more obvious, and are audible as ringing.

Manipulating Audio

Sometimes it is not enough to have just the original audio recording. A new recording needs impact or perhaps clean up. There are several type of audio enhancement and restoration effects. Noise reduction effects remove unwanted hiss, hum, clicks, or pops. Filter effects that let you change overall tonal balance, from rumbling bass tones to sparkling highs. Amplitude effects precisely control audio volume for increased radio impact, detailed fade outs, and more. All of these effects are available in Edit View, but some don't exist in Multitrack View because the two views are linked, however, this limitation can be easily overcomed . If a multitrack clip requires noise reduction, for example, simply double-clicking the clip will process it in Edit View.

Mastering describes the complete process of restoring and enhancing audio files for a particular medium, such as radio, video, CD, or the Web. In Adobe Audition, mastering is either individual audio files in Edit View or groups of files in a batch process. (Batch processing is particularly useful for burning a group of files to CD.

The mastering process consists of several stages, which are usually performed in the following order : Analysis, Noise reduction, Equalization, Compression, and Normalization To ensure that the loudest sounds reach the highest possible level that digital systems allow—0 dBFS. Reversing the order of the equalization is also possible. In Edit View, the frequency, phase, and dynamic range of an audio file can be analyzed. These analysis options are particularly helpful when used in conjunction with the many enhancement and restoration effects in Adobe Audition. For example, the Frequency Analysis window can be used to identify problematic frequency bands, and then correct them with a filter effect. Similarly, Waveform Statistics dialog box can be used to determine dynamic range and then compress it with an amplitude effect.

Adobe Audition contains a wide variety of effects for changing stereo imagery, adjusting pitch, and adding delay (for example, reverb and echo). Dialog boxes for these effects share many common options, such as graphs, spline curves, presets, and previews. Effects provide much of the functionality in Adobe Audition. For example, the use of effects to remove noise, optimize volume, change pitch, and add reverb. If Adobe Audition doesn't provide the effect required, it is also possible to purchase a plug-in effect to do the job. Adobe Audition can change the apparent location, or stereo imagery, of sounds coming from the speakers. For instance, moving a sound from the center to the left or right speaker or even make sounds seem to circle a listener's head. Note that all of the stereo imagery effects except the Doppler Shifter effect work only on stereo files.

Chorus, flanger, and phaser effects can thicken sound or make it outrageous. They range from the Chorus effect's ability to make a single instrument or vocalist sound like a group playing or singing in unison, to the wilder sounds of the Flanger effect and the phaser effects. They can be applied not only in stereo for the most dimensional results, but also mono sound as well. The Chorus effect adds richness as if several voices or instruments are played at once. It's a great way to add a degree of presence to a track. It can be used to give a stereo effect to a mono sample (where the left and right channels are identical) or to add harmony or "thickness" to a vocal track. It also can be used to create some truly out-of-this-world special effects. Adobe Audition uses a direct-simulation method of achieving a chorus effect, meaning that each voice (or layer) is made to sound distinct from the original by slightly varying the timing, intonation, and vibrato. The Feedback setting add extra detail to the result.

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 has a phase-shifted, time-delay effect, characteristic of psychedelic recordings of the 1960s and 1970s. The Flanger dialog box creates a similar result by slightly delaying and phasing a signal at predetermined or random intervals.

Similar to flanging, phasing introduces a variable phase-shift to a split signal and recombines it, creating psychedelic effects first popularized by guitarists of the 1960s. The Sweeping Phaser effect sweeps a notch- or boost-type filter back and forth about a center frequency. A phase is similar to a flange except that instead of using a simple delay, frequencies are phase-shifted over time. If a phase is used on stereo files, the stereo image can be dramatically altered to create some remarkably interesting sounds.

Conclusion

Adobe audition provides intuitive interface for the user to create or modify audio file. It offers variety of effects to complement either any audio that exist previously or that has been created newly. To create an audio file is easy task by this application. Audio conversion and manipulation is not a hard task any more because adobe audition provides ease of use to the user who needs another format of audio files or perhaps enhanced audio files for many purpose.

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