Document No.: Digital Music Record Convert Burn Station Help Document

Digital Music Record Convert Burn Station

Digital Smart Software

http://www.audioeditor.us

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Introduction

Digital Music Record Convert Burn Station is the most reliable and complete audio software available in its price range. It includes powerful audio tools such as audio converter, audio recorder, CD burner and CD grabber, CD burner .

Comprehensive, well designed, and easy & fun to use, Digital Smart Software offers the best value in audio software.

Five applications are provided in Digital Music Record Convert Burn Station, they are:
Audio Recorder
Audio Converter
Audio CD Burner
Audio MP3 CD Burner
Audio Grabber

If you need some help, please read this User Guide, if you can not find something here or you need some explanation or assistance, please feel free to e-mail us via E-mail or visit our website - http://www.audioeditor.us.

The user interface (Overview)

Digital Music Record Convert Burn Station is the most reliable and complete audio software available in its price range. It includes powerful audio tools such as audio converter, audio recorder, CD burner and CD grabber, CD burner .

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How to buy Digital Music Record Convert Burn Station

As soon as you make your order, our resellers will verify it. Your order will most likely be processed within less than 1 hour, but in some VERY rare cases it may take resellers more than 24 hours to process your payment.

The registration key will be automatically generated at our server and e-mailed to you immediately after we receive payment confirmation from our e-commerce reseller.

Please do not worry if you haven't received the registration information right away. Delays usually occur due to the high security settings of spam filters used by our clients. Our message may be rejected as a spam message by the mail service you use.

If you haven't got the registration message within several hours, feel free to contact our Support Teamvia email.

If you have questions concerning our software, please send e-mail to: support@audioeditor.us. **We always do our best to help you!**

Why Digital Music Record Convert Burn Station?

Save Time

a. It is outstanding both in **speed** and **audio quality**.
b. The **easy-to-use interface** helps you catch on to the system quickly.
c. According to surveys, it saves **35%** time in audio conversion.

Save Money

a. It is your **one-way ticket** to audio converting, creating and burning.
b. **30-day money-back guarantee** if you are not satisfied with it!

All in one tool

Do more with your digital media. Audio recorder, audio converter, CD burner, MP3 CD burner and CD grabber are all supported by this powerful tool.

Easy-to-use

With step-by-step manual, Whether you are an experienced user or a beginner, the software will make the task of converting audio files a breeze!

What will you have after purchase?

Full version of Digital Music Record Convert Burn Station

Fun in unlimited audio converting, audio recording, CD grabbing and audio burning.

Customer care

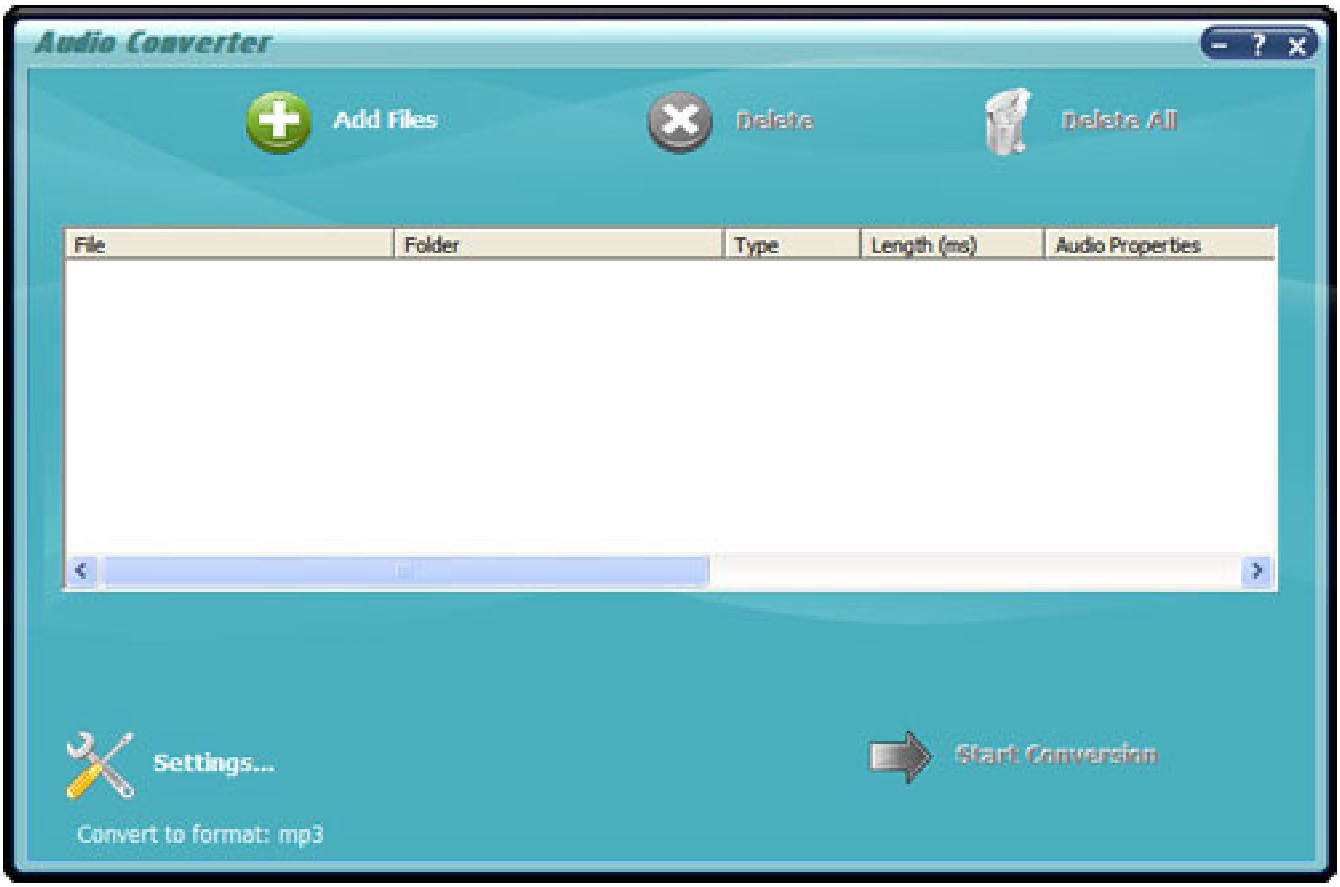
We are pleased to offer our care to meet your needs. We promise that any customer question will be replied within 1 business day!

Is my order secure?

We promise the most secure purchase as we did for years.
ShareItand RegNow, all these Digital Smart Software partners passed strict certification. We truly believe in 'Only by benefiting our customer can we benefit ourselves'. So your purchase security in Digital Smart Software is our top priority! Digital Smart Software has been involved in E-commerce for years. And through these years, we built up a secure online shopping system. You can enjoy the speed and convenience we offer.

Introduction

Audio Converter enables you to easily convert your audio files from one format to another. You can convert single files or batch convert a list of files. It supports a wide range of audio formats such as MP3/MP2, WMA, WAV, OGG, MPC, VOX, RAW, G723, G726.



Step1-Getting started

Use the following buttons to create a list of files you want to convert:

Button	Description
Add File(s)	Adds one or several files selected from the specific folder to the file list.
Delete	Removes the highlighted file from the file list.
Delete All	Removes all of the files from the file list.

Right click the file list, you can find an additional menu, which allows you to perform the operations listed above, you can also select all of the files by clicking the Select All or using the Ctrl+A shortcut. You can select files of different formats and from various locations in the file list.

Step2-Selecting output file format

Select the output format and specify the output parameters such as Frequency, Bit rate and Channels by clicking the **Settings** button.

Note: it is possible to specify fine tuning parameters for MP3 formats when you click the Advanced button. You can find the detailed information about these settings in the Appendix section. See also:

Available MP3 Parameters Combinations
Available MP2 Parameters Combinations
Available Ogg Vorbis Parameters Combinations

Step3-Selecting output file location

Audio Converter will automatically save all of the output files in **My Music** folder on your computer. To change output file location, please click the Settings button, and then click the Browse button to select desired output location.

Step4-Converting process

If everything looks all right, please click the **Start Conversion** button to start converting process.
Note: Audio Converter offers the option to close the program (or shut down windows) after converting process is finished.

You can visually judge the completion of the converting process with the Progress Bar.

You can also cancel converting process by clicking the Cancel button.

Selecting the desired file quality

Before you start converting, you have to select the desired sound quality in the Audio Converter Settings window.

MP3 Preset Quality:

The easiest way of choosing a level of compression for your MP3 files is to use the Preset Quality slider.

The Preset Quality slider allows you to choose from several levels of quality and resultant file sizes. Higher quality results in larger file size; lower quality results in smaller file size.

Preset	Bit rate	Mode	LAME
Phone	16 kbps	Mono	X
Shortwave	24 kbps	Mono	X
AM Radio	32 kbps	Mono	X
FM Radio*	96 kbps	Stereo	X (J-stereo)
Voice	64 kbps	Mono	X
Radio	112 kbps	X	Stereo
Tape	128 kbps	Stereo	J-stereo
Hi-Fi	160 kbps	Stereo	J-stereo
CD	192 kbps	Stereo	Stereo
Studio	256 kbps	Stereo	Stereo

The default is CD quality (192 kbps, Stereo).

* Note that the FM Radio bit rate is higher than the next preset quality, which is Voice. Voice is @ 64 kbps and FM Radio is @ 96 kbps. The reason that there is a discrepancy is because Voice is mono and FM Radio is stereo. If you were to divide the FM Radio bit rate in half, you would have 48 kbps per channel, which is lower than the Voice preset.

WMA Quality:

To get to the WMA File Options, go into the Settings and click the Output format - WMA tab. WMA is a second generation compressed audio format.

The main option on this screen is the quality format, which is chosen by moving the slider. The possible settings are:

Preset	Bit rate	Mode	Frequency

1	8kbps	Mono	8,000 Hz
2	32 kbps	Stereo	22,050 Hz
3	48 kbps	Mono	44,100 Hz
4	64 kbps	Stereo	44,100 Hz
5	96 kbps	Stereo	44,100 Hz
6	128 kbps	Stereo	44,100 Hz (default)
7	192 kbps	Stereo	44,100 Hz

WAV Quality:

CD Quality: This is the most popular setting.

CD Quality
This is the most popular setting. If you are creating WAV files so that you can burn them on custom CD compilations, choose this option. It will ensure that any non-CD quality format WAVs and MP3 files are converted to CD quality compatible WAV files. CD quality format WAV files are 44,100 Hz, 16 bits, stereo.

Specific Settings:

Specific Settings
The last alternative can be used for special situations where you need all your resultant WAV files to be in a specific format. For example, you can use this if you need all your WAV files to be mono. **Sample Rate** The sample rate is the number of sound points per second. For example, 22,050 Hz means that there will be 22,050 samples of sound data every second. You can also type in the sample rate if its not in the list. **Mode** Stereo means that the resultant files will have a left and right channel. Mono means that there is only one channel. **Bits Per Sample** Each sample is stored in a finite number of bits. The more bits you use, the less noise is introduced to the sound. However, the file size will also get bigger as higher bits per sample are used. The default is 16 bit. 8 bit sounds may sound noisy especially for subtle dynamic sounds.

Please note that the higher you select in sound quality, the more disk space is required to store the converted files.
If you have little disk space left, then this might be a reason to select a lower sound quality.

MP3 advanced

MP3 Advanced window includes 4 tabs:

- General
- Filtering
- VBR
- Expert

General

In the **General** tab you can select the necessary **Channels**. The suggested choice includes the following variants: **Stereo**, **Joint Stereo**, **Forced Joint Stereo**, **Dual Channels** and **Mono**. Read the description of each mode in the table below:

Channel	Description
Stereo	In this mode, the encoder makes no use of potentially existing correlations between the two input channels. It can, however, negotiate the bit demand between both channels, i.e. give one channel more bits if the other contains silence.
Joint stereo	In this mode, the encoder will make use of a correlation between both channels. The signal will be matrixed into a sum ("mid") and difference ("side") signal. For quasi-mono signals, this will give a significant gain in encoding quality. This mode does not destroy phase information like IS stereo that may be used by other encoders. This setting can be used to encode DOLBY ProLogic surround signals.
Forced Joint Stereo	This mode will force MS joint stereo on all frames. It's faster and it uses some special mid and side masking threshold.
Dual Channels	In this mode, the 2 channels will be totally independently encoded. Each channel will have exactly half of the bitrate. This mode is designed for applications like dual languages encoding (for example: English in one channel and French in the other). Using this encoding mode for regular stereo files will result in a lower quality encoding.
Mono	This option will generate a mono file, if the input file is a stereo file, the input stream will be downsampled to a mono file by averaging the left and right channel.

In the right part you will find the opportunity to set **Encoder Quality**. You can specify the output quality; thus you can trade off encoding time against sound quality. The default (normal) is recommended for the lower bit rates (<160 kbps), high quality for bit rates >160 kbps. The voice quality is more or less optimized to generate the best quality for voice

There are some built-in presets you can use. They have for the most part been subject to and tuned via rigorous double blind listening tests to verify and achieve this objective. These are continually updated to coincide with the latest developments that occur and as a result should provide you with nearly the best quality currently possible. You can find them in **Preset** and **Alt-Preset** lists.

Presets available:

Phone, **SW**, **AM**, **FM**, **Voice**, **Radio**, **Tape**, **HiFi**, **CD**, **Studio**, **R3Mix**.

Alt-Presets available:

Preset	**Description**
None	Do not use any alt-presets
Fast Standard	VBR mode preset. It should generally be understandable to most people with most music and is already quite high in quality. The resulting bit rate should be within the 170-210 kbps range, according to music complexity. Enables the new fast VBR method for a Standard preset. Its disadvantage is that often the bitrate will be slightly higher than with the normal mode and quality may be slightly lower also.
Standard	VBR mode preset. It should generally be understandable to most people with most music and is already quite high in quality. The resulting bit rate should be within the 170-210kbps range, according to music complexity.
Fast Extreme	VBR mode preset. If you have extremely good hearing and similar equipment, this preset will provide slightly higher quality than the "standard" mode. The resulting bit rate should be within the 200-240 kbps range, according to music complexity. The "fast"-option increases speed significantly but may give a tiny bit lower quality.
Extreme	VBR mode preset. If you have extremely good hearing and similar equipment, this preset will provide slightly higher quality than the "standard" mode. The resulting bit rate should be within the 200-240 kbps range, according to music complexity.
Insane	CBR mode preset. The option gives you the current theoretical maximum quality possible. The output files are flat 320 kbps. Using this may be a little insane, since the difference in quality in between extreme and insane is minimal. However, if you simply don't care about file size, want maximum quality, or you have hardware that can't handle VBR files, then you could use this option.

You can also select **Mpeg Tag Version** out of **ID3 Ver.1** or **ID3 Ver.2**.

ID3 Ver.1

The audio format MPEG layer I, layer II and layer III (MP3) has no native way of saving information about the contents, except for some simple yes/no parameters like "private", "copyrighted" and "original home" (meaning this is the original file and not a copy). A solution to this problem was introduced with the program "Studio3" by Eric Kemp alias NamkraD in 1996. By adding a small chunk of extra data in the end of the file one could get the MP3 file to carry information about the audio and not just the audio itself.

The placement of the tag, as the data was called, was probably chosen as there were little chance that it should disturb decoders. In order to make it easy to detect a fixed size of 128 bytes was chosen. The tag has the following layout (as hinted by the scheme to the right):

Song title	30 characters
Artist	30 characters
Album	30 characters
Year	4 characters
Comment	30 characters
Genre	1 byte

If one sums the size of all these fields we see that 30+30+30+4+30+1 equals 125 bytes and not 128 bytes. The missing three bytes can be found at the very beginning of the tag, before the song title. These three bytes are always "TAG" and is the identification that this is indeed a ID3 tag. The easiest way to find a ID3v1/1.1 tag is to look for the word "TAG" 128 bytes from the end of a file.

As all artists doesn't have a 30 character name it is said that if there is some bytes left after the information is entered in the field, those bytes should be file with the binary value 0. You might also think that you cannot write that much in the genre field, being one byte big, but it is more clever than that. The byte value you enter in the genre field corresponds to a value in a predefined list. The list that Eric Kemp created had 80 entries, ranging from 0 to 79.

ID3 Ver.2

ID3v2 is a new tagging system that lets you put enriching and relevant information about your audio files within them. In more down to earth terms, ID3v2 is a chunk of data prepended to the binary audio data. Each ID3v2 tag holds one or more smaller chunks of information, called frames. These frames can contain any kind of information and data you could think of such as title, album, performer, website, lyrics, equalizer presets, pictures etc. The block scheme to the right is an example of how the layout of a typical ID3v2 tagged audio file may look like.

One of the design goals were that the ID3v2 should be very flexible and expandable. It is very easy to add new functions to the ID3v2 tag, because, just like in HTML, all parsers will ignore any information they don't recognize. Since each frame can be 16MB and the entire tag can be 256MB you'll probably never again be in the same situation as when you tried to write a useful comment in the old ID3 being limited to 30 characters.

Speaking of characters, the ID3v2 supports Unicode so even if you use the Bopomofo character set you'll be able to write in your native language. You can also include in which language you're writing so that one file might contain e.g. the same lyrics but in different languages.

Even though the tag supports a lot of byte consuming capabilities like inline pictures and even the possibility to include any other file, ID3v2 still tries to use the bytes as efficient as possibly. If you convert an ID3v1 tag to an ID3v2 tag it is even likely that the new tag will be smaller. If you convert an ID3v1 tag where all fields are full (that is, all 30 characters are used in every field) to an ID3v2 tag it will be 56 bytes bigger. This is the worst case scenario for ID3v1 to ID3v2 conversion.

Since it's so easy to implement new functionality into ID3v2, one can hope that we'll see a lot of creative uses for ID3v2 in the future. E.g. there is a built-in system for rating the music and counting how often you listen to a file, just to mention some brainstorm results that are included. This feature can be used to build play lists that play your favorite songs more often than others.

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Filtering

High pass Filter cuts the lowest frequencies and passes the highest. The **Low pass Filter** allows only the lower frequencies to be present into the output signal; it will cut the beautiful crystal sound of a violin (frequencies over 10 KHz), but if it could amplify rather than just pass the low frequencies, than it would enhance your favorite disco music with lots of percussions and bass.

High pass filtering frequency (in kHz): Frequencies below the specified one will be cut off.

Width of High pass filter (in kHz): The width of the high pass filter. The default is 15% of the high pass frequency.

Low pass filtering frequency (in kHz): Frequencies above the specified one will be cut off.

Width of Low pass filter (in kHz): The width of the low pass filter. The default is 15% of the low pass frequency.

Regarding to the Nyquist Sampling Theorem the sample rate have to be at least two times higher than the highest frequency of analog audio signal. For example, the human ear can detect sound across the frequency range of 20 Hz to 20 kHz. According to the sampling theorem, one should sample sound signals at least at 40 kHz in order for the reconstructed sound signal to be acceptable to the human ear.

So applying the low-pass analog filter to a file with sample rate of 8kHz you should take into consideration that the frequency parameter of this method have to be at least two times less (< 4 kHz) than the file's sample rate.

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VBR

At **VBR** tab you should select Coding Method first. The description of each method you can find in the table below:

Coding Method	Description
None	Don't use VBR, instead it is encoding with a Constant Bit Rate (CBR)
Default	Use the default VBR method (currently set to VBR-MTRH)
Old	The functional approach, based on masking, bisection in the bit domain
New	The approach, based on masking and direct noise allocation
MTRH	A merger of old and new (VBR) routine
ABR	The Average Bit Rate (ABR) setting, the encoding principle is based on perceptual entropy, but more like CBR than VBR

Maximum VBR Bitrate: Allows to specify maximum bit rate when using VBR (Variable Bit Rate), this selecting depends on what base bit rate you have chosen in the main encoder tab. It's recommended to set 320 kbit/s unless you want low quality VBR files.

VBR Quality: In VBR mode, you are able to specify a quality setting which will affect encoding bit rate allocation. If you use quality 0, the max bit rate will be reached easily, while using quality 9 the bit rate usually will be around the base bit rate. The lower the VBR quality value, the better the audio quality, but also the bigger the output file. Recommended setting for high quality VBR encoding is 1 or 0.

Write VBR Header: This tag is embedded in frame 0 of the MP3 file. It lets VBR aware players correctly seek and calculate playing times of VBR files.

Target bit rate for ABR: The allowed range of the ABR bit rate is 4 - 310 kbit/s, you can use any integer value within that range.

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Expert

At Expert tab you can find the opportunity to set the following options:

- **Include CRC-Checksum**
  When enabled, the encoder will calculate the cyclic redundancy check (CRC) for the MP3 frames, and will add the CRC value to the MP3 stream.
- **Comply as much as possible to ISO MPEG spec**
  With this option, the encoder will enforce the 7680 bit limitation on total frame size. This results in many wasted bits for high bit rate encodings.
- **Allow block types to differ between channels**
  Allows the left and right channels to use different block types. Normally this is not allowed, only because the FhG encoder does not seem to allow it either.
- **Short blocks**
  Encode all frames using short blocks.
- **Bit reservoir**
  Enable bit reservoir.

You can also set **ATH Control** function, useful for low volume. ATH is used to approximate an equal loudness curve. Select one of the suggested modes:

ATH Control Mode	Description
Default	
Only	This option ignores the output of the psy-model and only use masking from the ATH. Might be useful at very high bitrates or for testing the ATH.
Disabled	Disable any use of the ATH (absolute threshold of hearing) for masking. Normally, humans are unable to hear any sound below this threshold.
Only for short blocks	Ignore psychoacoustic model for short blocks, use ATH only.

There are also three **De-emphasis** options: **default**, **0 / 15 microseconds** and **citt j.17.**

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Bit rate

Bit rate is defined as the number of data points used to approximate the true wave form. Obviously, the higher you set, the more accurately the wave form is approximated.

The CODEC takes the bit rate into consideration as it writes each frame to the bit stream. If the bit rate is low, the irrelevancy and redundancy criteria will be measured harshly, and more subtlety will be stripped out, which results in poorer sound quality. If the bit rate is high, the codec will be applied with leniency, and the sound quality will be better, which will result in larger file size.

Constant Bit Rate (CBR) encoding maintains the same bit rate throughout an encoded file. If your audio has moments of silence, it is captured and encoded at that rate, as are areas of very diverse frequencies (that might sound better if encoded to a higher bit rate).

Variable Bit Rate (VBR) is an MP3 encoding method. As its name implies, the bit rate is varied throughout the file. The codec guesses which parts could benefit from more bits per second, and which can use less. The result is a much higher quality file.

Frequency

Sampling frequency also impacts fidelity. The sampling frequency is essentially the number of times the sound event is quantized within a given time period. Sampling frequencies are specified in KiloHertz (KHz), a term meaning samples per second. The key in understanding how sampling frequency affects fidelity is the Nyquist sampling theorem. Basically, when applied to audio signals the Nyquist theorem states that the highest possible pitch in the sound is one-half that of the sampling frequency.

For example, "CD-quality" sound requires 16-bit words sampled at 44.1 KHz. Essentially this means 44,100 16-bit words (705,600 bits) are used to digitally describe each second of sound on a compact disc. The highest pitch possible is 22.05 KHz (approximately the top of human hearing range), which is half of 44.1 KHz.

Audio file formats

Audio Converter supports the following audio formats:

Codec	Description
ADPCM	Compressed WAV format. ADPCM (Adaptive Differential Pulse Code Modulation) is an audio compression scheme which compresses from 16-bit to 4-bit for a 4:1 compression ratio.
ALAW	Compressed WAV format. 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 is rather fast and generally, widely supported.
DSP	Compressed WAV format. DSP Group True Speech (TM) format.
GSM	Compressed WAV format. Good for keeping human speech.
G.726	Used for computer telephony. Good for keeping human speech.
MP3	MPEG Layer-3 format. Very popular format for keeping music.
PCM	Standard Windows WAV format for non-compressed audio files. Pulse Code Modulation (PCM) is the standard method of digitally encoding audio. It is the basic uncompressed data format used in file types such as Windows .wav.
ULAW	Compressed WAV format. u-Law (or CCITT standard G.711) is an audio compression scheme and international standard in telephony applications. u-Law is very similar to A-Law, a variation of u-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, u-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.
VOX	Dialogic ADPCM format. 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.
RAW	Raw format of audio files. Doesn't contain header of an audio file.
WMA	Windows Media Audio format. A special type of advanced streaming format file for use with audio content encoded with the Windows Media Audio codec. The .wma extension indicates a file format and how the content is encoded.
CCIT U-Law	Compressed WAV format.
Ogg Vorbis	Ogg Vorbis format.

MP3 format

MP3 files contain perceptually encoded sound data.

The frequencies that humans cannot perceive are removed, although some audio purists say they can tell the difference between a high bit-rate MP3 and a Wave file.

A MP3 file is 10 times smaller than an equivalent WAV file.

MP3 files usually end with .mp3, .mp1 or .mp2 file extensions.

.MP3
.MP2
.MP1

WMA format

WMA is acronym for Windows Media Audio. WMA files contain perceptually encoded sound data.

The frequencies that humans cannot perceive are removed, although some audio purists say they can tell the difference between a high bit-rate WMA and a Wave file.

A WMA file can be as much as 20 times smaller than an equivalent WAV file.

.WMA

Wave format

Wave files usually contain uncompressed PCM audio data.

Sometimes, PCM data may be compressed as ADPCM, GSM or True-Speech.

Wave files end with the .wav extension.

.WAV

Available mp3 parameters combinations

You can find available MP3 parameters combinations in the table below:

Frequency	MONO		Stereo	
	Minimum	Maximum	Minimum	Maximum
48 KHz	64 Kbps	320 Kbps	128 Kbps	320 Kbps
44 KHz	56 Kbps	320 Kbps	112 Kbps	320 Kbps
32 KHz	40 Kbps	320 Kbps	80 Kbps	320 Kbps
24 KHz	32 Kbps	160 Kbps	64 Kbps	160 Kbps
22 KHz	32 Kbps	160 Kbps	56 Kbps	160 Kbps
16 KHz	32 Kbps	160 Kbps	40 Kbps	160 Kbps
12 KHz	24 Kbps	160 Kbps	32 Kbps	160 Kbps
11 KHz	16 Kbps	160 Kbps	32 Kbps	160 Kbps
8 KHz	8 Kbps	160 Kbps	8 Kbps	160 Kbps

Frequency	8 KHz		11.025 KHz		16 KHz		22.05 KHz		24 KHz		32 KHz		44.1 KHz		48 KHz	
Bitrate	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo
8	+	+	+													
16	+	+	+	+	+	+	+	+	+							
24	+	+	+	+	+	+	+	+	+	+						
32	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+
40	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+
48	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+
56	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+
64	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+	+
80					+	+	+	+	+	+	+	+	+	+	+	+
96					+	+	+	+	+	+	+	+	+	+	+	+
112					+	+	+	+	+	+	+	+	+	+	+	+
128					+	+	+	+	+	+	+	+	+	+	+	+
144					+	+	+	+	+	+	+	+	+	+	+	+
160					+	+	+	+	+	+	+	+	+	+	+	+
192											+	+	+	+	+	+
224											+	+	+	+	+	+
256											+	+	+	+	+	+
320											+	+	+	+	+	+
384																

MP3 supports all the combinations of the frequency and bit rate if the bit rate varies from Minimum to Maximum values for the current supported frequency.

Available mp2 parameters combinations

You can find available MP2 parameters combinations in the table below:

Frequency	16 KHz		22.05 KHz		24 KHz		32 KHz		44.1 KHz		48 KHz	
Bitrate	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo	mono	stereo
8	+		+		+							
16	+	+	+	+	+	+						
24	+	+	+	+	+	+						
32	+	+	+	+	+	+	+		+		+	
40	+	+	+	+	+	+	+		+		+	
48	+	+	+	+	+	+	+	+	+	+	+	+
56	+	+	+	+	+	+	+	+	+	+	+	+
64	+	+	+	+	+	+	+	+	+	+	+	+
80	+	+	+	+	+	+	+	+	+	+	+	+
96	+	+	+	+	+	+	+	+	+	+	+	+
112	+	+	+	+	+	+	+	+	+	+	+	+
128	+	+	+	+	+	+	+	+	+	+	+	+
144	+	+	+	+	+	+	+	+	+	+	+	+
160	+	+	+	+	+	+	+	+	+	+	+	+
192							+	+	+	+	+	+
224							+	+	+	+	+	+
256							+	+	+	+	+	+
320							+	+	+	+	+	+
384							+	+	+	+	+	+

MP2 supports all the combinations of the frequency and bit rate if the frequency varies from Minimum to Maximum values for the current supported bit rate.

Available ogg parameters combinations

You can find the available ogg parameters combinations in the table below:

Frequency	Mono		Stereo	
	Minimum	**Maximum**	**Minimum**	**Maximum**
48 KHz	48 Kbps	192 Kbps	48 Kbps	320 Kbps
44 KHz	48 Kbps	128 Kbps	48 Kbps	256 Kbps
32 KHz	48 Kbps	112 Kbps	48 Kbps	192 Kbps
24 KHz	32 Kbps	56 Kbps	48 Kbps	160 Kbps
22 KHz	32 Kbps	56 Kbps	48 Kbps	160 Kbps
16 KHz	48 Kbps	96 Kbps	48 Kbps	192 Kbps
12 KHz	32 Kbps	48 Kbps	48 Kbps	96 Kbps
11 KHz	48 Kbps	48 Kbps	48 Kbps	64 Kbps
8 KHz	8 Kbps	24 Kbps	16 Kbps	48 Kbps

Ogg Vorbis supports all the combinations of the frequency and bit rate if the bit rate varies from Minimum to Maximum values for the current supported frequency.

Audio converter keystrokes

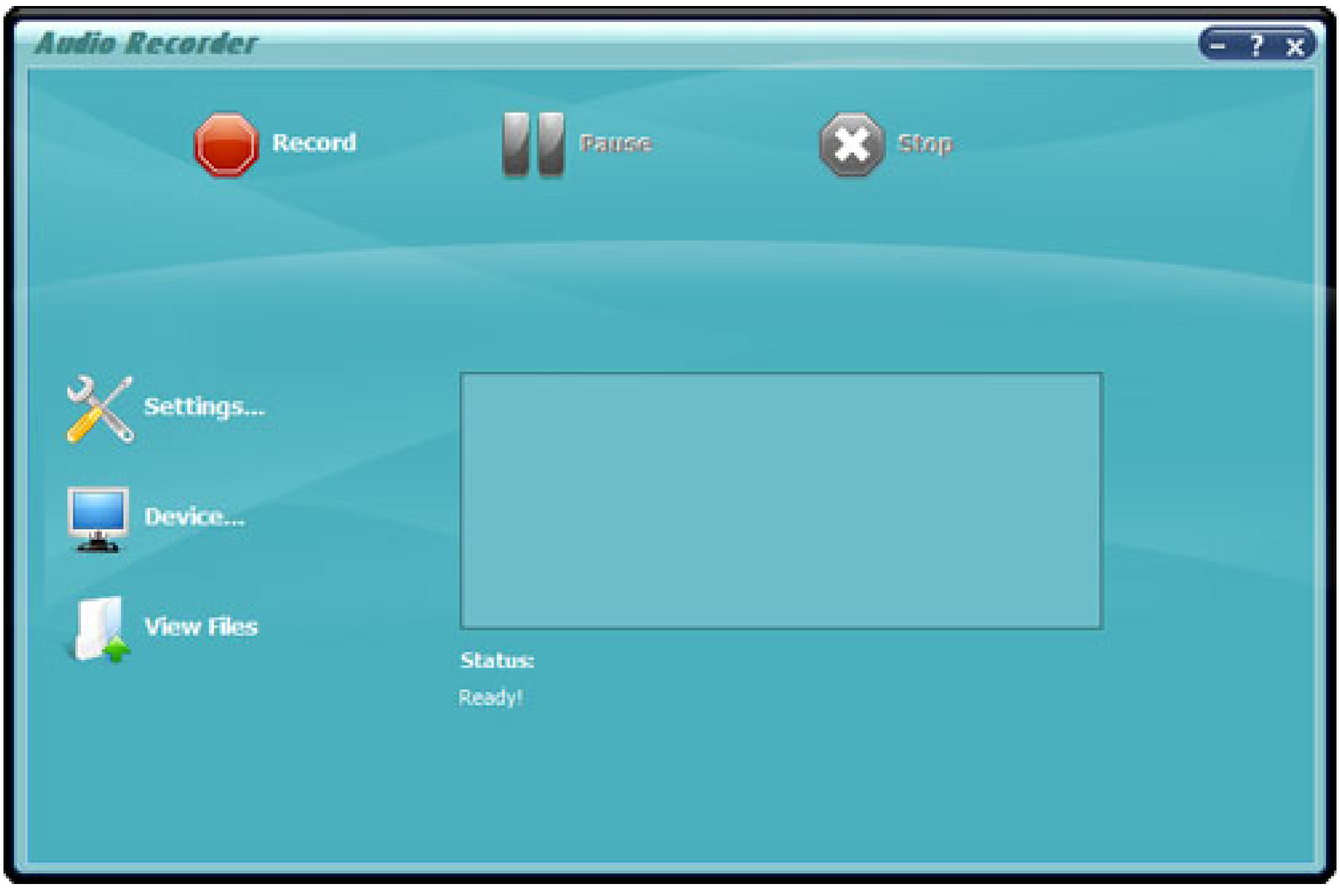
This is a list of the functions in the Audio Converter, which can be operated by pressing certain keys on the keyboard:

Key:	Function:
F9	Start converting
Ctrl+P	Play the selected file
Alt+F4	Exit
Ctrl+A	Select all files
Ctrl+N	Select none
Del	Delete select files from list
Ctrl+U	Move up
Ctrl+D	Move down
F1	Help

Overview

With **Audio Recorder** you can record voice from microphone, internet streaming audio, or music played by Winamp, Windows Media Player, Quick Time, Real Player, etc.

Audio Recorder is able to automatically detect the recording formats your sound card supports and then set the application's parameters for the best possible performance. The recordings can be saved as .mp3, .wav, .wma, .vqf and .ogg files.



See also:

- Making an audio CD of a video, music cassette or record

Making an audio CD

The **Audio Recorder** provides you with an easy way to digitize audio recordings from videos, music cassettes or LPs.

Read the following information and tips to help you when making audio CDs:

- Digital recordings
- Sound quality
- Copyrights
- CD tracks
- System requirements

Digital recordings

Audio CD contains digital information to reproduce sound. The digital information consists of strings of numbers, which encode the frequency and volume of sound.

A music cassette or LP contains analog information to reproduce sound.

Before you can make an audio CD from a music cassette or LP, the analog recording must first be digitized. The digitizing is what the Audio Recorder does, in cooperation with your sound card. The digital recording is saved to the hard drive as a WAV,MP3 or WMA file.

Sound quality

Audio CDs are known to have a perfect sound quality, which means that a CD recording almost perfectly reproduces the original sound.

When you digitize an LP or music cassette, the clicks and noises are digitized as well, for they are part of the sound that is produced. Using the filters in the Audio Recorder you can enhance the sound quality and make the presence of noise or clicks sound less disturbing.

Because you will listen to a CD recording of your music cassettes and LPs for a long time, you have to pay special attention to the following issues when digitizing:

- Always make use of the best playback equipment you have.
  It is highly recommended to tear apart your living room audio set and to drag that perfect cassette player to your computer.

- If you are going to digitize recordings from a music cassette, then set your cassette player tape switch to the right type (Normal, Chrome, Metal, etc).
  Also verify that you playback with the same noise reduction system (Dolby B or C for example), that was used to make the cassette recording.

- LPs must be cleaned and dusted well and preferably you must use a new pick-up needle.
  This reduces the chance for clicks and noise that will be difficult to filter out later.

- While digitizing, pay attention to the recording volume level in the Audio Recorder.
  A recording volume that is too high will distort the sound. See also: Setting the volume.

- Make sure you select the Compact Disc stereo CD quality (Mp3 format & Wav format) or 128kbps, 44100hz, Stereo quality (Wma format) in the Audio Recorder "Settings-Output Format Settings" window if you want to create standard audio Compact Discs.
  It is recommended to save recordings on disk as uncompressed WAV files, especially when you are going to burn them to audio CDs that will be played back on a regular audio CD player.

CD tracks

Audio CDs contain more than one track, which make it easy to search for a particular fragment by choosing a certain track number on your CD player.

When you are going to make an audio CD of digital recordings, you need a separate digital sound file for each individual track.

Quick tour

With the Sound Recorder you can digitize audio recordings from music cassettes, LPs, videos or any other sound source.

A new recording will be stored in a temporary file on the hard drive. When you click the Stop button, the new recording will automatically be created.

Making a recording is very easy. Follow these 4 steps for a successful recording:

- Connecting the source
- Selecting the desired recording quality
- Setting the volume
- Record

See also:

- Did you know... (tips & tricks)
- Fast keys

Connecting the source

To be able to make a digital recording of sound from music cassettes, LP's or videos, you must have the video recorder, cassette player or record player connected with the sound card in your computer.

Here are some general guidelines:

- A cassette player or video recorder can usually be connected to the sound card directly through the **Line In** or **Auxiliary** input connector.
  On a cassette player you must connect the **Out** or **Play** output connectors to the sound card.
  On a video recorder you must connect the (stereo) audio output connector to the sound card. The audio output signal is usually available together with the video output signal on a so-called SCART output connector on the video recorder. Special cables are available at your retailer.

- The signal of a record player is usually too weak to be connected to the sound card directly and must be connected to the **Line In** or **Auxiliary** input connector of the sound card through an amplifier.
  You connect the record player to the amplifier and the amplifier to the sound card. Usually you can use the **Rec Out** output connector on the amplifier for the connection to the sound card.

  TIPS:

  Do not connect a record player to the microphone input of the sound card! A microphone connection of a sound card is usually not stereo but mono and also the signal might be distorted because the microphone channel the sound card is not designed for other sources but microphones.

- At Sound source in the **Audio Recorder** window you choose which input of the sound card must be recorded. Usually this is **Auxiliary** or **Line In** for recording the connected video recorder, cassette player or record player. You can of course also record from any of the other sources, like the microphone. Your sound card provides the kinds and naming of available sound sources and this will differ from one sound card to another.

You can hear what you're recording through the computer speakers connected to the sound card.

Selecting the desired recording quality

Before you start recording, you must select the desired sound quality of the recording in the Audio Recorder Settings window.

MP3 Preset Quality:

The easiest way of choosing a level of compression for your MP3 files is to use the Preset Quality slider.

The Preset Quality slider allows you to choose from several levels of quality and resultant file sizes. As you choose higher quality, the file size goes up as well. On the other hand, lower quality presets will be smaller and easier to send over the Internet.

Preset	Bit rate	Mode	LAME
Phone	16 kbps	Mono	X
Shortwave	24 kbps	Mono	X
AM Radio	32 kbps	Mono	X
FM Radio*	96 kbps	Stereo	X (J-stereo)
Voice	64 kbps	Mono	X
Radio	112 kbps	Stereo	X (Stereo)
Tape	128 kbps	Stereo	X (Stereo)
Hi-Fi	160 kbps	Stereo	X (Stereo)
CD	192 kbps	Stereo	Stereo
Studio	256 kbps	Stereo	Stereo

The default is CD quality (192 kbps, Stereo).

* Note that the FM Radio bit rate is higher than the next preset quality, which is Voice. Voice is @ 64 kbps and FM Radio is @ 96 kbps. The reason that there is a discrepancy is because Voice is mono and FM Radio is stereo. If you were to divide the FM Radio bit rate in half, you would have 48 kbps per channel, which is lower than the Voice preset.

WMA Quality:

To get to the WMA File Options, go into the Settings and click the Output format - WMA tab. WMA is a second generation compressed audio format.

The main option on this screen is the quality format, which is chosen by moving the slider. The possible settings are:

Preset	Bit rate	Mode	Frequency
1	8 kbps	Mono	8,000 Hz
2	32 kbps	Stereo	22,050 Hz
3	48 kbps	Mono	44,100 Hz
4	64 kbps	Stereo	44,100 Hz
5	96 kbps	Stereo	44,100 Hz
6	128 kbps	Stereo	44,100 Hz (default)
7	192 kbps	Stereo	44,100 Hz

WAV Quality:

CD Quality: This is the most popular setting. Use this when creating WAV files for burning on custom audio cds.

CD Quality
This is the most popular setting. If you are creating WAV files so that you can burn them on custom CD compilations, choose this option. It will ensure that any non-CD quality format WAVs and MP3 files are converted to CD quality compatible WAV files. CD quality format WAV files are 44,100 Hz, 16 bits, stereo.

Specific Settings: Choose this if you want the WAV files in a specific format.

Specific Settings
The last alternative can be used for special situations where you need all your resulting WAV files to be in a specific format. For example, you can use this if you need all your WAV files to be mono. **Sample Rate** The sample rate is the number of sound points per second. For example, 22,050 Hz means that there will be 22,050 samples of sound data every second. You can also type in the sample rate if its not in the list. **Mode** Stereo means that the resulting files will have a left and right channel. Mono means that there is only one channel. **Bits Per Sample** Each sample is stored in a finite number of bits. The more bits you use, the less noise is introduced to the sound. However, the file size will also get bigger as higher bits per sample are used. The default is 16 bit. 8 bit sounds may sound noisy especially for subtle dynamic sounds.

TIPS:

1. Note that the sound quality of the recording will never be better than the original quality of the sound source. A rule of thumb is to select the sound quality that most closely compares to the quality of the sound source.

2. If you are not sure what to choose, then always select **Presets - CD Quality**.

Although it is possible, it does not make sense to record a telephone conversation using **Presets - CD Quality** in stereo. That would consume unnecessary disk space to accommodate the kind of detail that is not present in a telephone line.
On the other hand, is also makes no sense to record a stereo compact disc using mono telephone recording quality, since that would discard most of the enhanced details from the music.

Please note that the higher the selected sound quality, the more disk space is required to store the recording.
If you have little disk space left, then this might be a reason to select a lower sound quality.
The maximum possible duration of a recording with the selected sound quality is reported in the Sound Recorder window.

Setting the volume

Like recording on a cassette recorder, you must adjust the recording volume.

1. Check that the sound source is correctly connected to the sound card and that the right source is selected. Read Connecting the source for more information.

2. Check that the speakers are switched on and that the volume is open.

3. Play a loud fragment of the song you want to record on the video recorder, cassette player or record player. Adjust the **Volume** in the Audio Recorder Settings - automatic gain control window.

Some sound sources, like Digital CD sources, don't have a volume control and the volume control will be disabled. The volume will then automatically be at the correct level.

Some sound cards don't support separate control of the left and right sound channel volume which does not mean that the recording will not be in stereo.

Record

You don’t have to pay attention to the exact moment when the sound starts or ends while recording.

1. Set the video recorder, the cassette player or the record player on stand-by at the beginning of the song you want to record.

2. Start recording by clicking the red record button in the Audio Recorder.

3. Start playback of the song on the video recorder, cassette player or record player.

4. While recording, please pay attention that the peak meters will not reach the red zone too often or for too long. See also: Setting the Volume.

5. When the song is finished, click the stop button.

The new recording will be stored in a temporary file on the hard drive. When you click the Stop button, the new recording is automatically created.

Audio Recorder keystrokes

This is a list of the functions in the Audio Recorder, which can be operated by pressing certain keys on the keyboard:

Key:	**Function:**
F9	Start recording
F11	Stop recording
F10	Pause recording
Ctrl+O	View recorded files
F2	Config settings
F1	Help

Did you know

... that the Sound Recorder can record from virtually any sound source that is audible through the computer speakers?

- Recording Internet broadcasts (streaming audio)
- Recording live performances

... and that you can:

- Convert home study courses from tape to CD
- Publish MP3 sound files on the Internet or E-mail them to friends

... and that most functions of the software are also accessible with keyboard keystrokes:

- Audio Recorder keystrokes

... and that you can read more about:

- Making an audio CD
- Connecting cassette, tape or record-players
- Recording sound qualities
- Frequently asked questions

Visit our site on the Internet as well: http://www.audioeditor.us

Recording Internet broadcasts (streaming audio)

With the Audio Recorder you can record from virtually any sound source that is audible through the computer speakers, including Internet radio broadcasts (streaming audio). Just make sure to follow these steps:

1. First start playback of the Internet broadcast in the player you are using, like RealPlayer.
2. Wait for RealPlayer to start playback. First it will download a couple of seconds of sound, before it starts playback.
3. Start the Sound Recorder and select the appropriate sound source. The names of the sound source differs from one system to another, but the sound source for recording RealAudio playback, is usually named **Wave**, **Stereo Mix**, **What You Hear**, or something in a similar wording. The Sound Recorder will automatically select the appropriate recording quality.

4. Start recording in the Sound Recorder. Optionally you can restart playback in RealPlayer, to record the whole sound clip from the beginning.

Note that on certain computers you cannot change the recording quality to anything other than that of the sound quality of the Internet broadcast while recording.

Also on certain computers you will get an error message in RealPlayer about not being able to access the sound card if you started the Sound Recorder before starting play-back in the RealAudio player. In that case the selected recording quality in the Sound Recorder window does not match that of the Internet broadcast and then RealPlayer cannot access the sound card for playback. Just follow the steps explained above, to resolve this conflict.

Recording live performances

With the Audio Recorder you can also record live performances. There are a couple of options here:

1. Connect a microphone to the Mic input of the computer or laptop and record the live performance directly from the **Mic or Microphone** sound source. On many computers the microphone input is mono and not stereo, so if you connect a stereo microphone (or two separate microphones with an adapter plug), the recording will only be in mono. If you see two separate volume sliders in the Sound Recorder window when you select the **Mic or Microphone** sound source, then you can record in stereo.

2. Connect the Line-Out output on an external sound system or PA to the **Line-In** or **Aux** input on the computer and record from the **Line-In**, **Aux** or **Auxiliary** sound source. This is of course most appropriate with complex performances where some sort of sound system is already in place, like in theaters or concert halls.

3. Record the live performance with other traditional equipment, like a cassette-, tape- or minidisk-recorder. Later these recordings can be transferred to your computer by connecting a cassette-, tape- or minidisk-player to the **Line-In** or **Aux** input on the computer and record from the **Line-In**, **Aux** or **Auxiliary** sound source. See Connecting cassette, tape or record-players as well.

Convert home study courses from tape to CD

Many home study courses like language courses come with cassette tapes containing audible lesson material. Constantly having to wind and re-wind those tapes to find the right fragment can become very annoying and might even withhold you from completing the course.

Converting those cassette tapes to CD is the solution! On a CD player you can more easily jump to the fragment you need and also makes it easier to replay the same fragment over and over.

Record the tapes using the Audio Recorder and subsequently use Editor to create separate tracks of each lesson.

Publish MP3 sound files on the Internet or E-mail them to friends

With the Sound Recorder you can record and save compressed MP3 sound files.

Compressed MP3 sound files are usually small enough to publish on the Internet for download by others, or to send them as attachments in an E-mail message.
You'd better not do this with uncompressed WAV sound files because they are very large end are impractical to download or receive by E-mail within a reasonable amount of time.
Besides that, most E-mail providers have a limit on the E-mail attachment size.

Using the filters (AGC)

Sound cards use AGC (Automatic Gain Control) to improve recordings.

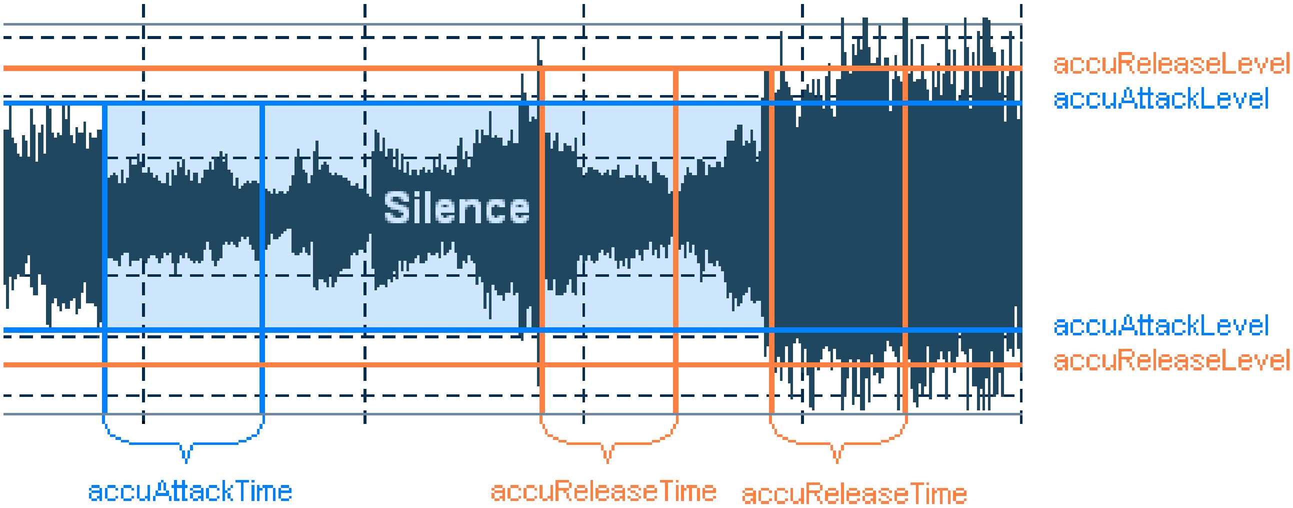
The following properties allow to get or set parameters of AGC: High Level, Low Level, Attack Time.

The volume level will be decreased if it is higher than the High Level and the volume level will be increased if it is lower than the Low Level. The rate of the volume changing is equal to (20db / Attack Time).

AGC Properties	Details
High Level	The level is in db. Varies from -92 to 0. This is the maximal allowed volume level.
Low Level	The level is in db. Varies from -92 to 0. This is the minimal allowed volume level.
Attack Time	The time for level change by 20 db, in ms.

Using voice active system

The following picture shows how the accu algorithm works:



If signal is below the Attack Level for more than Attack Time, that spot in the audio will be considered the beginning of a pause in audio.

If signal is above the Release Level for more than Release Time, that spot in the audio will be considered end of the pause in audio.

Silence Definition

For very quiet high quality audio, the Attack Level and Release Level value will be lower (like -60dB). For noisier audio, the value may be much higher (like -30dB).

If audio is above this given threshold for more than the number of milliseconds given, audio will be considered valid, and not silence. Use higher values for Release Time to ignore short periods of audio (like clicks, static, or other noise). If this value is too high, short words may be skipped.

Aucc Properties	Details
Attack Level	The level is in db. Varies from -92 to 0. the level of loudness for the starting point of recording. This is a level for non-silence detection.
Attack Time	Time is in ms. The sounding time for the starting point of recording. This is time for non-silence detection.
Release Level	The level is in db. Varies from -92 to 0. The level of loudness for the ending point of recording. This is a level for silence detection.
Release Time	Time is in ms. The silence time for the ending point of recording. This is time for silence detection.

Wave format

Wave files usually contain uncompressed PCM audio data.

In some cases, it may be compressed PCM data in a format such as ADPCM, GSM or True-Speech.

Wave files end with swav extension ..

.WAV

WMA format

WMA is short for Windows Media Audio. WMA files contain perceptually encoded sound data.

The frequencies that humans cannot perceive are removed, although some audio purists say they can tell the difference between a high bit-rate WMA and a Wave file.

A WMA file can be as much as 20 times smaller than an equivalent WAV file.

.WMA

MP3 format

MP3 files contain perceptually encoded sound data.

The frequencies that humans cannot perceive are removed, although some audio purists say they can tell the difference between a high bit-rate MP3 and a Wave file.

A typical MP3 is 10 times smaller than an equivalent WAV file.

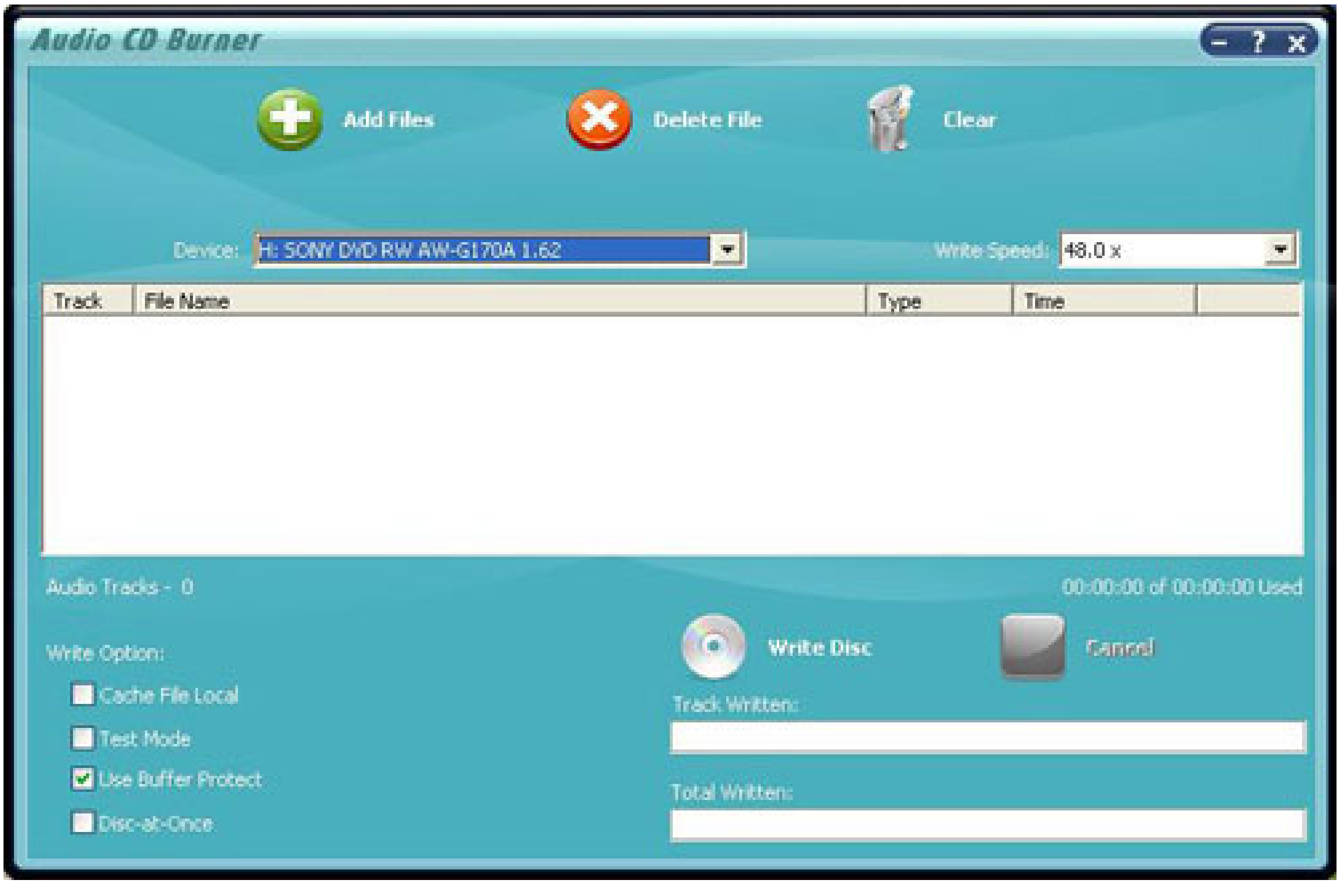
MP3 files usually end with mp3, mp1 or mp2 file extensions.

.MP3
.MP2
.MP1

Introduction

Audio CD Burner is able to burn audio CD from MP3, WAV, WMA, and OGG files to an audio CD playable in any standard CD player with ease.

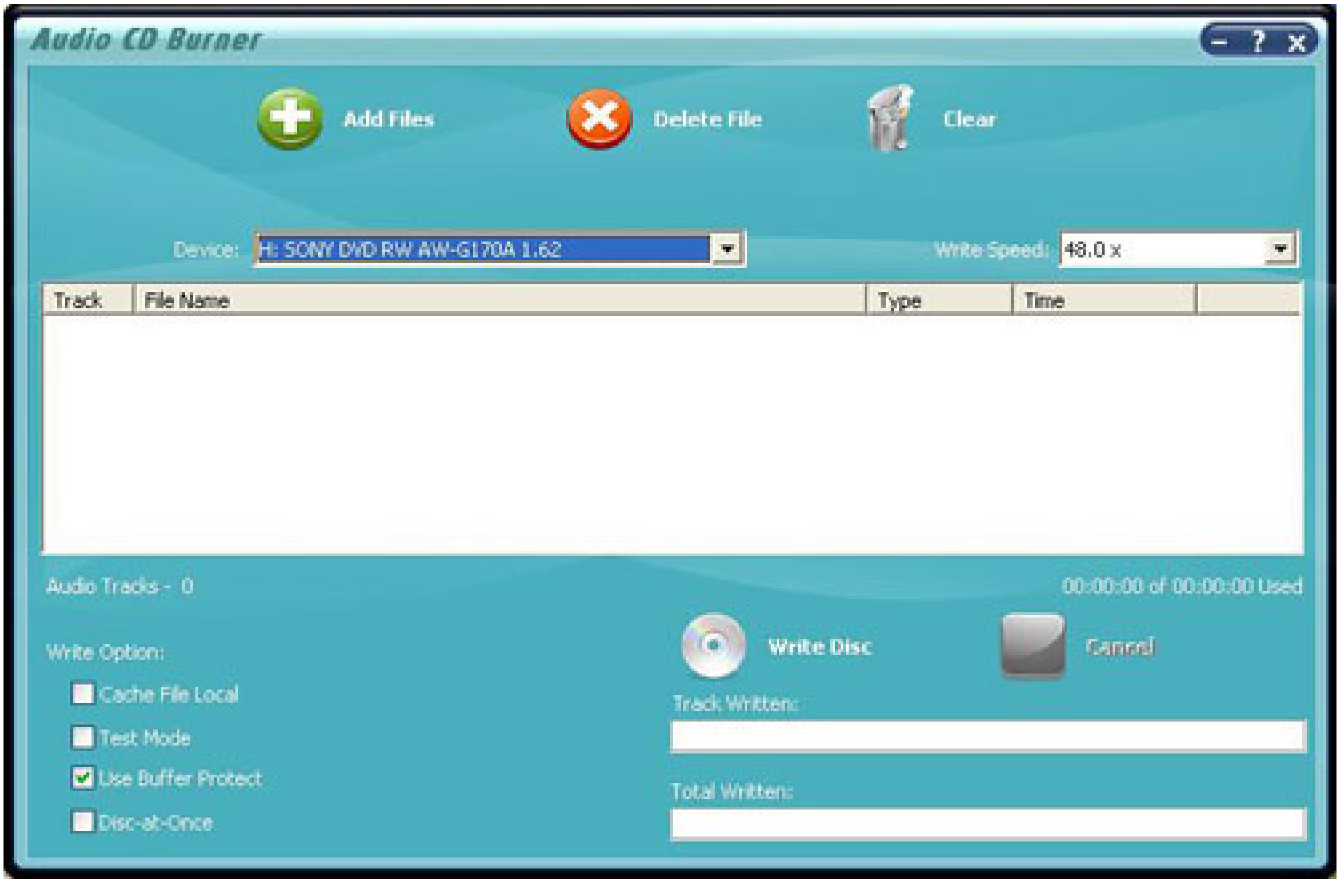
The great feature makes Audio CD Burner the perfect audio burner software for users of any experience level!



See also:

- How to proceed
- How to burn

How to proceed



- **Add Files**: Add files to burn them on CD later.
- **Delete File**: Delete unwanted files before burning.
- **Clear**: Delete all listed files.

Right-click a listed file to perform more operations:

- **Add Files (Ctrl +O)**: Add files to burn them on CD later.
- **Up (Ctrl +U)**: Change the order of the tracks.
- **Down (Ctrl +D)**: Change the order of the tracks.
- **Remove Track (Del)**: Delete selected files.
- **Clear Tracks (Shift +Del)**: Delete all listed files.
- **Eject (Alt +E)**: Eject disk.
- **Close Tray (Alt +C)**: Closes the device tray if applicable.

How to burn?

Let's start burning files step by step:

> **Add files**

Click "**Add Files**" button to add audio files one by one.

> **Burning Setting**

Device: Select desired writing device.

Write Speed: Select writing speed.

Write Option: Four options available, just check the boxes to perform corresponding operations when writing.

• **Cache File Local** -- Check the box to cache your files prior to writing.

• **Test mode** -- Check the box to make the the writing process in Test mode.

• **Use buffer protect** --Check the box to set the buffer protection.

• **Disc-at-once**: Check the box to perform DAO mode when burning; uncheck the box to perform TAO mode when burning.

awTAO: Disc is written in Track-at-Once mode and disc is Closed

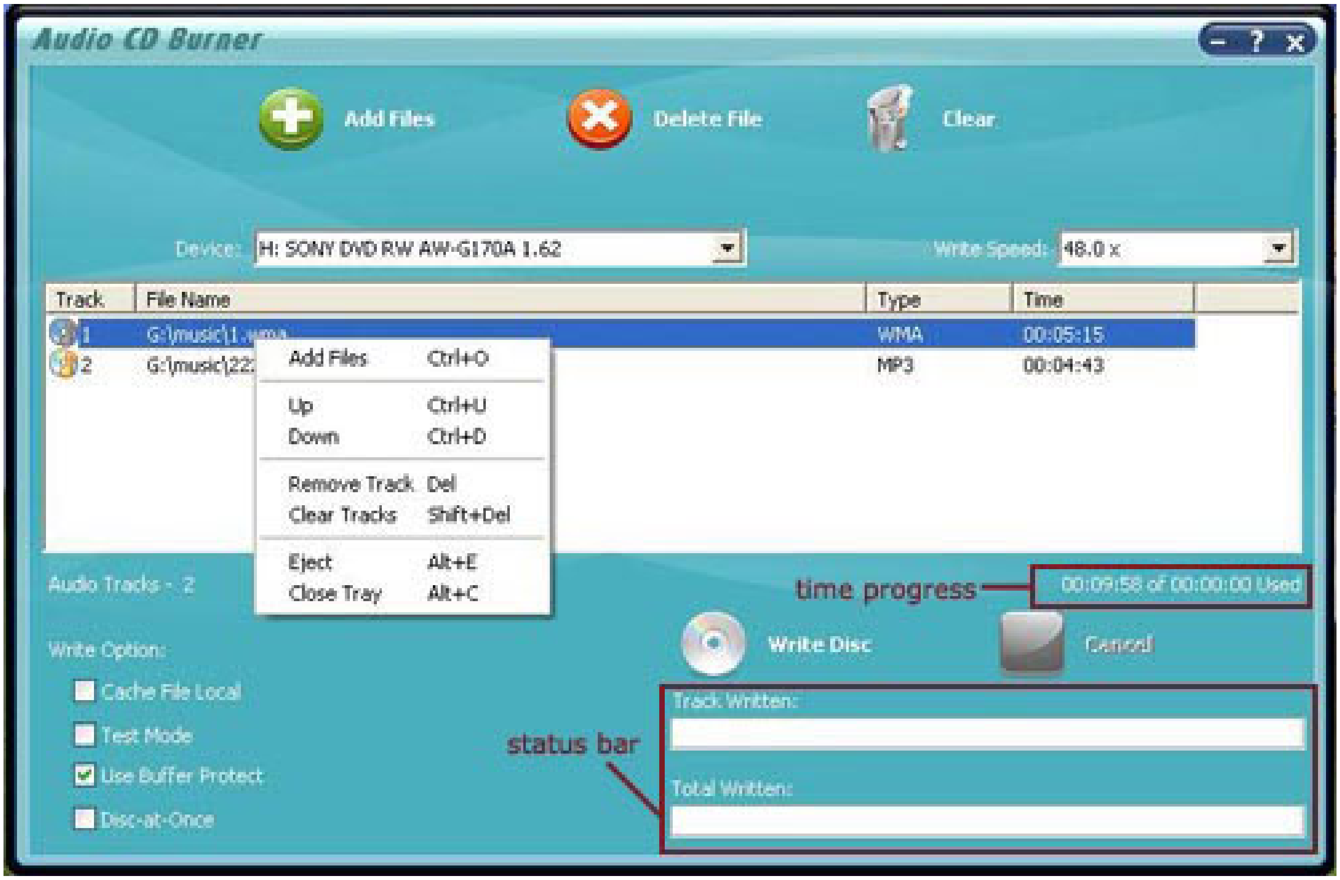
awDAO: Disc is written in any valid Disc-at-Once mode and disc is finalized (closed).

> **Burning**

Click "**Write Disc**" to start burning task.

The total duration of all listed files can not exceed **80min**.

Two status bars display detailed burning status.



Track Written: Display the burning progress of current file.

Total Written: Display the burning progress of all files.

MP3 format

MP3 files contain perceptually encoded sound data.

The frequencies that humans cannot perceive are removed, although some audio purists say they can tell the difference between a high bit-rate MP3 and a Wave file.

A typical MP3 is 10 times smaller than an equivalent WAV file.

MP3 files usually end with mp3, mp1 or mp2 file extensions.

.MP3
.MP2
.MP1

Wave format

Wave files usually contain uncompressed PCM audio data.

In some cases, it may be compressed PCM data in a format such as ADPCM, GSM or True-Speech.

Wave files end with swav extension .

.WAV

WMA format

WMA is acronym for Windows Media Audio. WMA files contain perceptually encoded sound data.

The frequencies that humans cannot perceive are removed, although some audio purists say they can tell the difference between a high bit-rate WMA and a Wave file.

A WMA file can be as much as 20 times smaller than an equivalent WAV file.

.WMA

Introduction

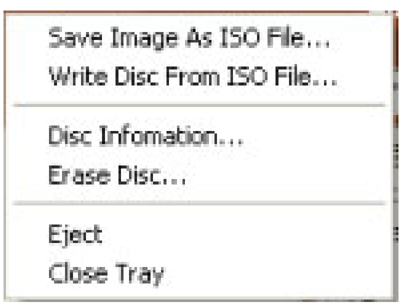
Burn your favorite songs to CDs so you can listen them in your home or car stereo MP3 CD player. **MP3 to CD Burner** is such a CD-burning software that can convert MP3 files to conventional MP3 CD or Audio CD with easy steps. It supports MP3 to CD on the fly with high speed and excellent quality.

See also:

- How to proceed
- How to burn

How to proceed

Move Down Icon: Select operation of burning process in the pop up dialog:

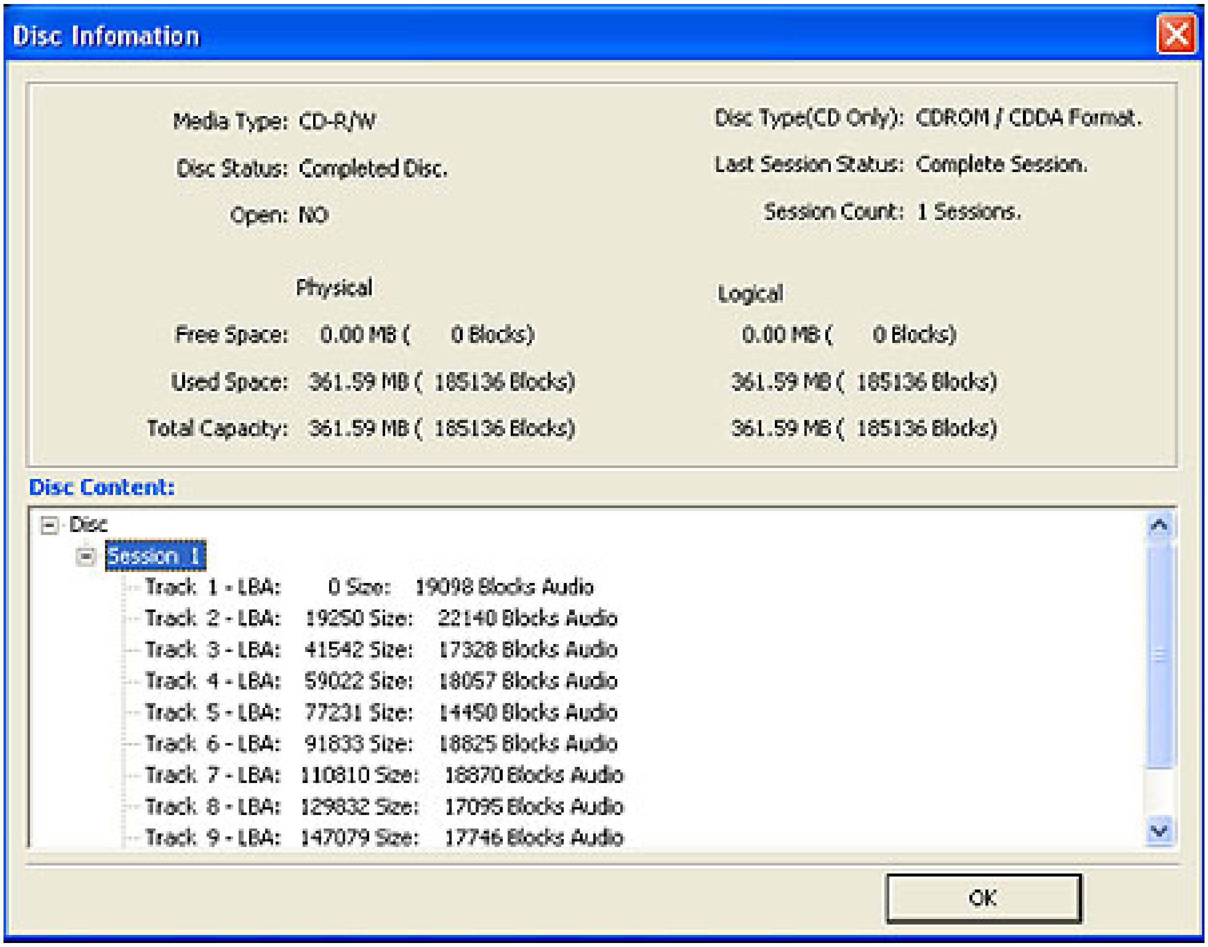


Save Image As ISO File: Save current listed files as ISO file.

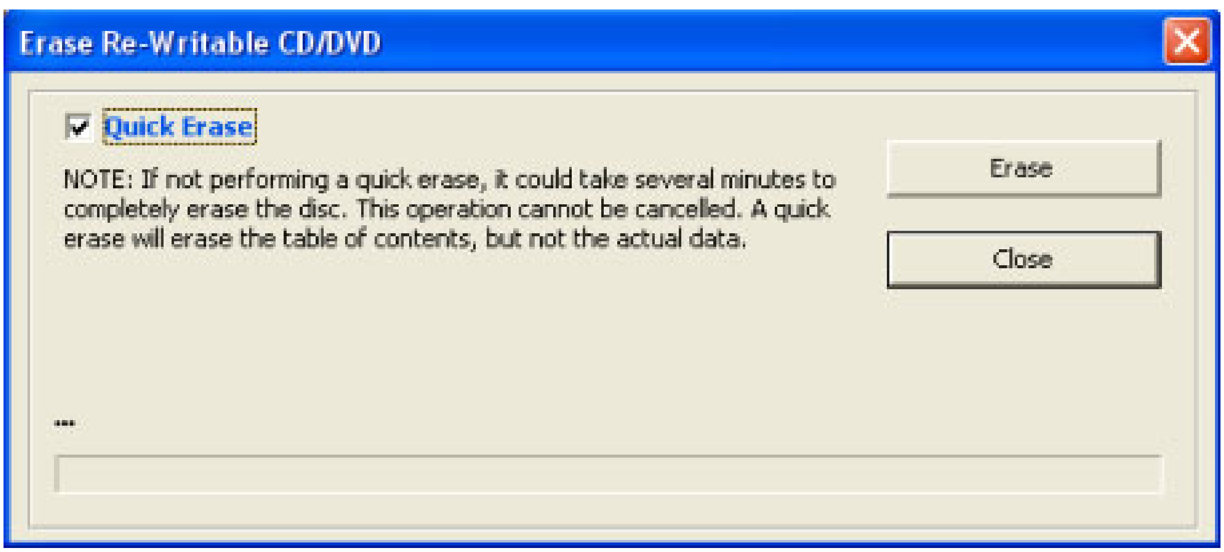
Hypothetical CD-ROM is required if you are able to open all ISO files.

Write Disc From ISO File: Write ISO files.

Disc Information: Display detailed information of the disc:



Erase Disc: Erase the hard disc quickly.



Eject: Eject disk.

Add Files: Add files to burn them on CD later.

Delete File: Delete unwanted files before burning

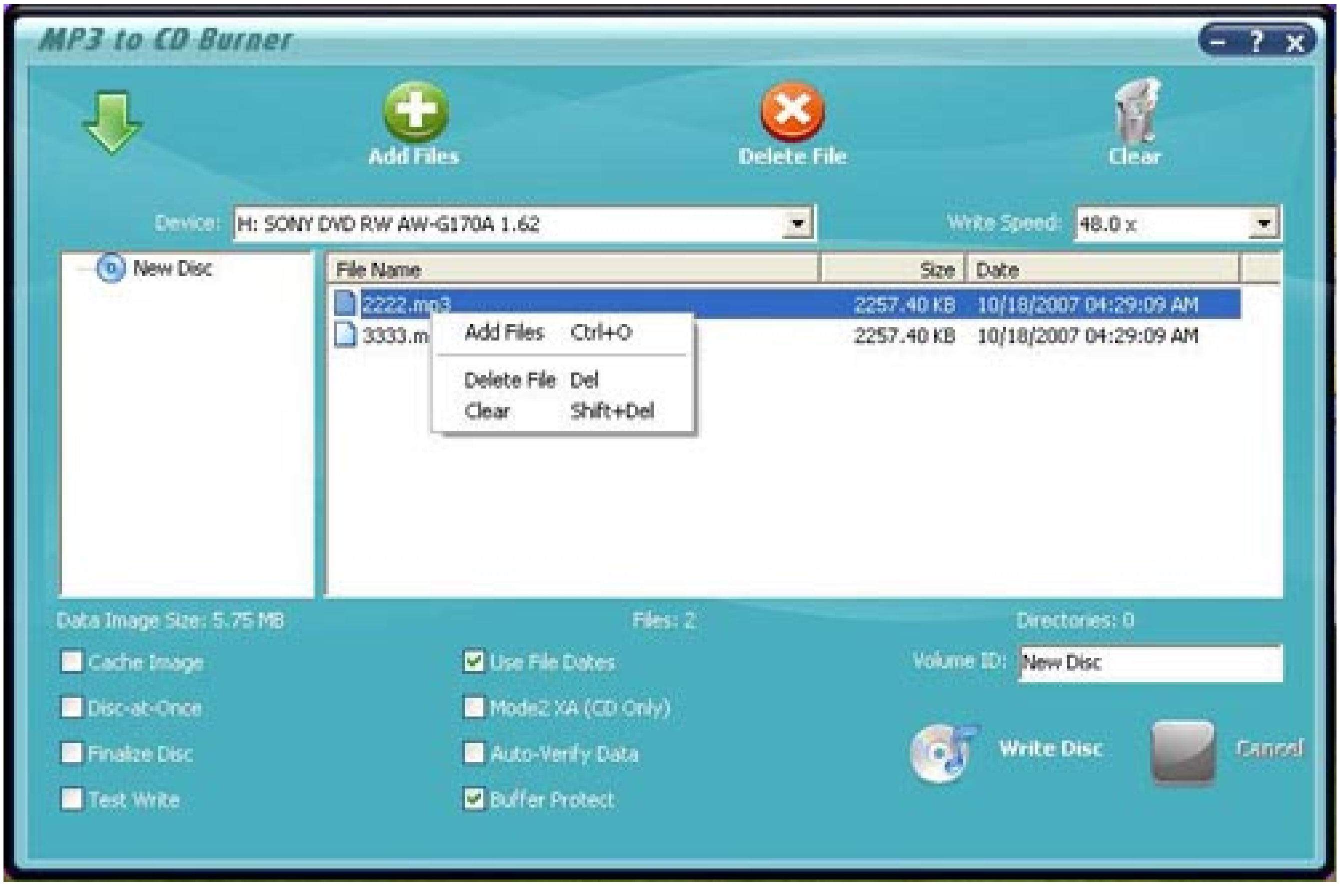
Clear: Delete all listed files.

Check box : Check a box to perform corresponding options.

- **Volume ID**: New Disc (Default). It enables you to rename the disc.

How to burn?

Let's start burning files step by step:



Add / Delete Files

Click "**Add Files**" button to add audio files.

Select listed files, click **Delete File (Del)** to delete selected file(s), and click **Clear** to delete all listed files.

Burning Setting

Device: Select desired writing device.

Write Speed: Select writing speed.

Write Option: Eight options available. Just check the boxes to perform corresponding operations when writing.

• **Cache File Local** -- Check the box to cache your files prior to writing.

• **Disc-at-once**-- Check the box to perform DAO mode when burning; uncheck the box to perform TAO mode when burning.

awTAO: Disc is written in Track-at-Once mode and disc is Closed

awDAO: Disc is written in any valid Disc-at-Once mode and disc is finalized (closed).

• **Test mode** -- Check the box to make the writing process in Test mode.

• **Finalize Disc**-- Check the box to close/finalize the disc. (Ignored for Disc-at-Once)

• **Use File Dates**-- Check the box to use today's date or the last modification. A Data item's date can also be altered directly.

• **Mode2 XA (CD Only)**-- Check the box to enable/disable Mode2XA. (Advanced)

• **Auto Verify Data**-- Check the box to verify files before burning.

• **Buffer protect** --Check the box to set the buffer protection.

Burning

Click "**Write Disc**" to start burning task.

MP3 format

MP3 files contain perceptually encoded sound data.

The frequencies that humans cannot perceive are removed, although some audio purists say they can tell the difference between a high bit-rate MP3 and a Wave file.

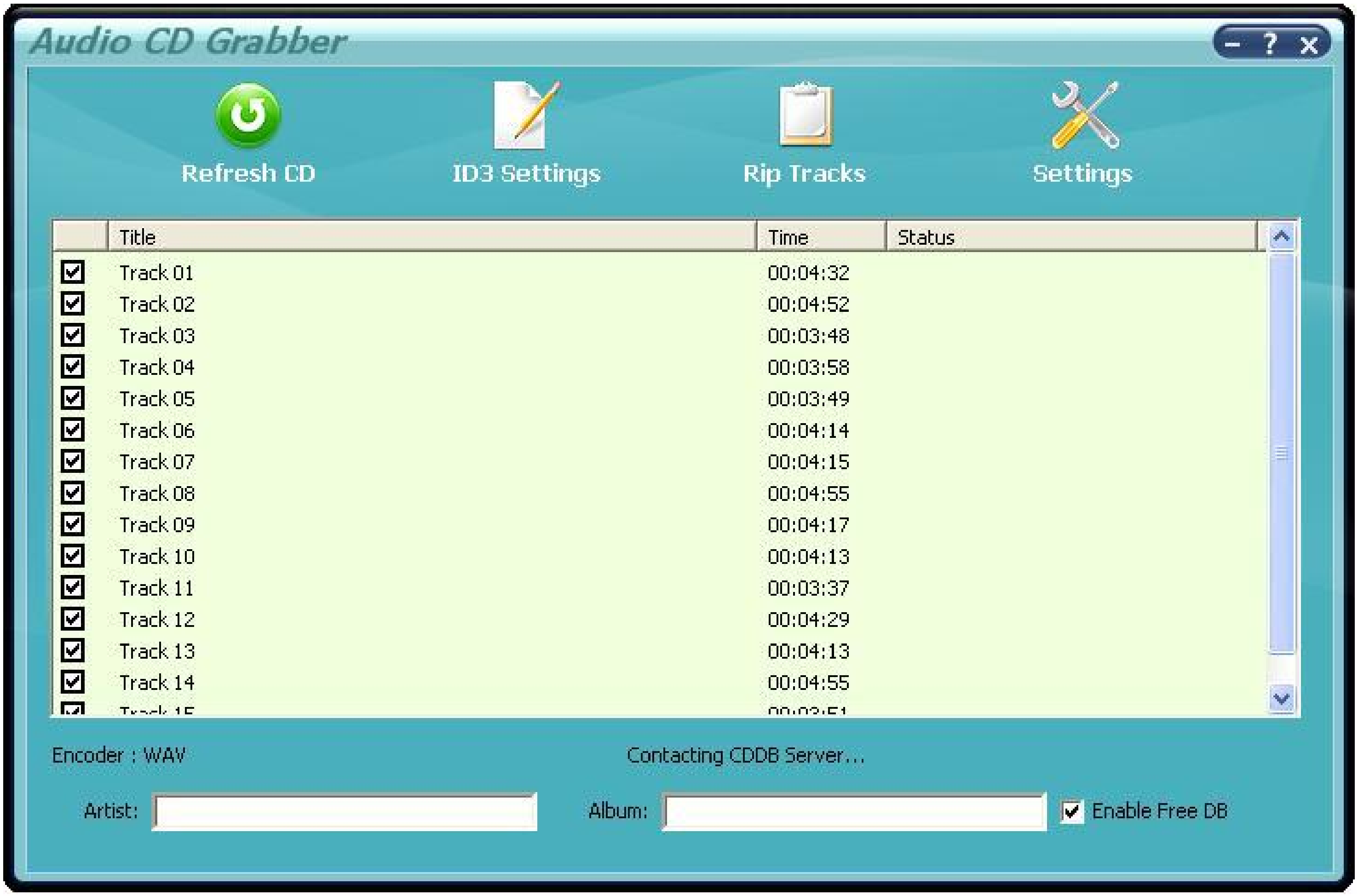
A typical MP3 is 10 times smaller than an equivalent WAV file.

MP3 files usually end with mp3, mp1 or mp2 file extensions.

.MP3
.MP2
.MP1

Introduction

Grab music with **Audio CD Grabber**. It is an easy-to-use audio application that enables you to record music, voice or other sound you hear such as voice from microphone, webcasts from the Internet, music played by Winamp, Windows Media Player, Quick Time, Real Player, Flash, games, etc. Besides you can also make your own WAV/MP3/WMA/OGG files, rip audio tracks from audio CD/DVD, save them to your hard disk and extract audio stream from Internet Streaming Media and save them as WAV, MP3, RAW or OGG files.



Refresh CD: Rescans the TOC of the currently loaded CD.

ID3 Settings: Edit ID3 tag information.

Rip Tracks: Begins ripping all tracks marked with a check mark.

Settings: Brings up the options dialog.

Artist: Edit the Artist's name. Included in the ID3 tag.

Album: Edit the Album title. Included in the ID3 tag.

Enable Free DB: Tick for querying the Freedb database.

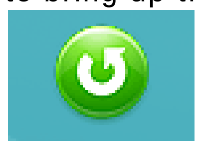
Features list:

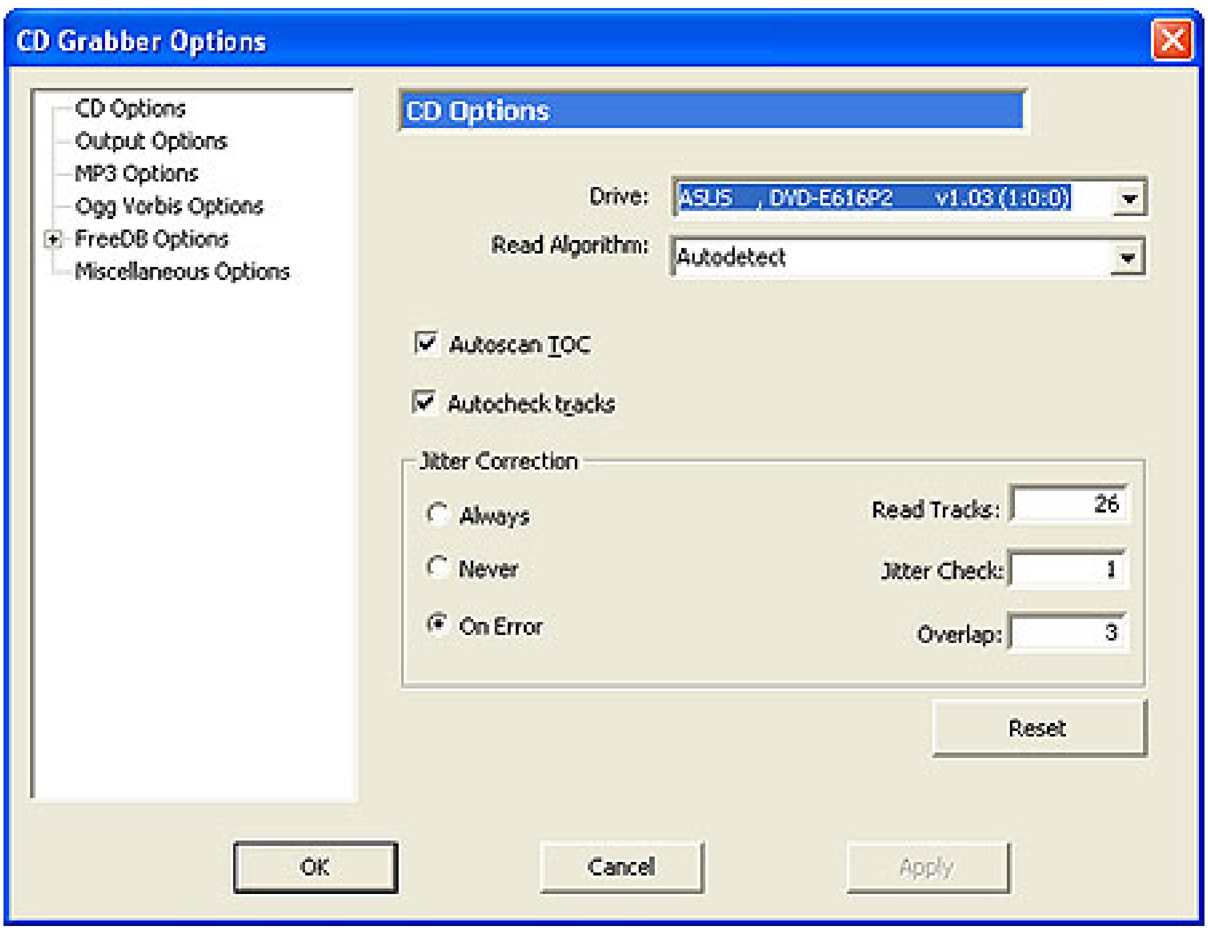
- Easy to use and user-friendly interface.
- Name the files which you rip using information received from the CDDB source
- Rip audio tracks from a personal CD and save them to your hard drive in various digital audio formats, like RAW, OGG, MP3 and WAV music files
- When ripping change the information obtained from the audio source (Name of Artist, Album and tracks)
- Select details you want to include in the file name for the audio tracks you rip (Track Number, Song Title, Artist, Album)
- Add the information to your output audio file such as Title, Artist, Album, Comment
- Visually judge the progress of the ripping process.

Quick Start

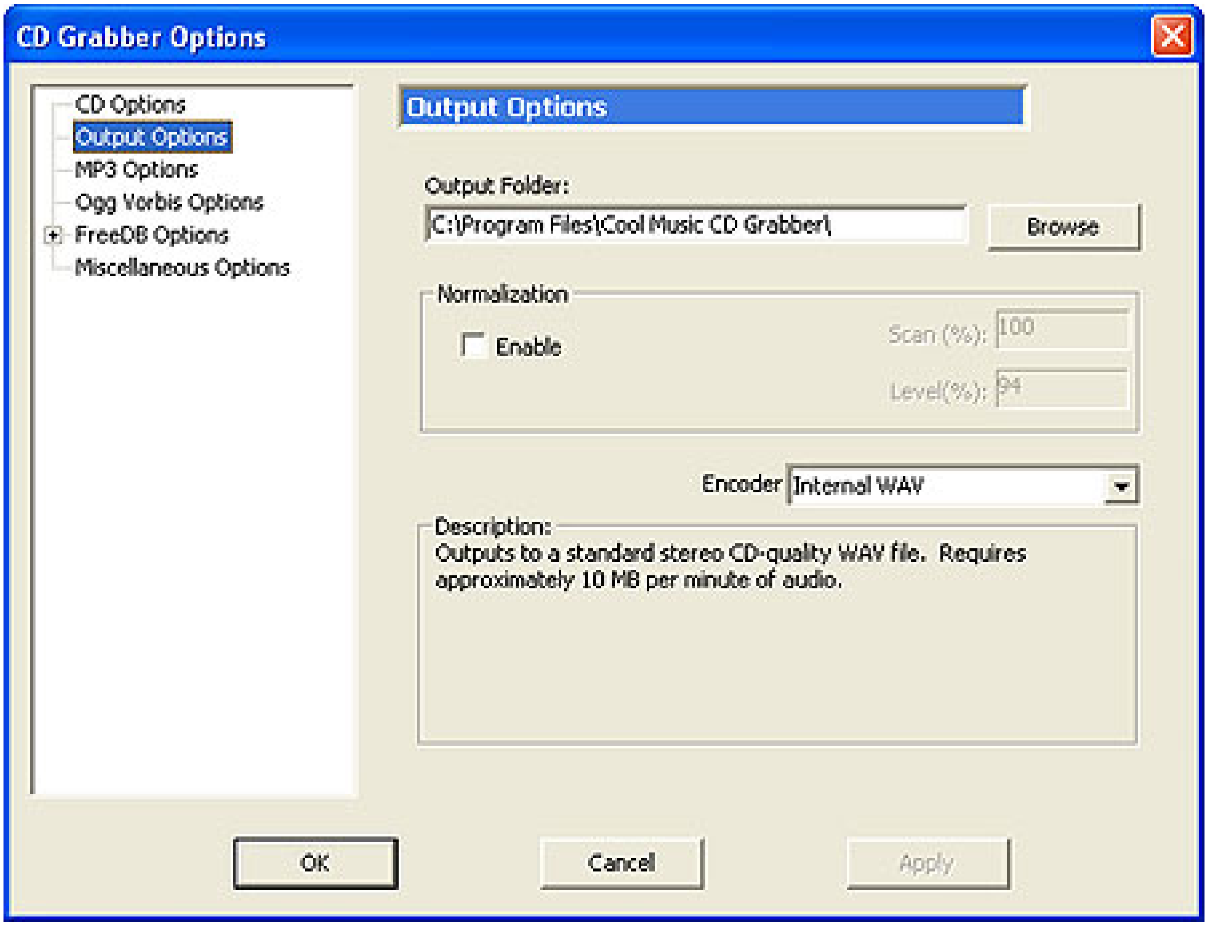
The following instructions will get you ripping tracks in no time!

1. Start the program. The default location is Start->Audio CD Grabber.

2. Select a CD unit by clicking the  button to bring up the CD options dialog. Place a CD into the CD drive on your computer and hit the  button. You can cause Audio CD Grabber to automatically scan the TOC on startup by clicking on the Settings Button on the main window and checking the Auto scan TOC checkbox.

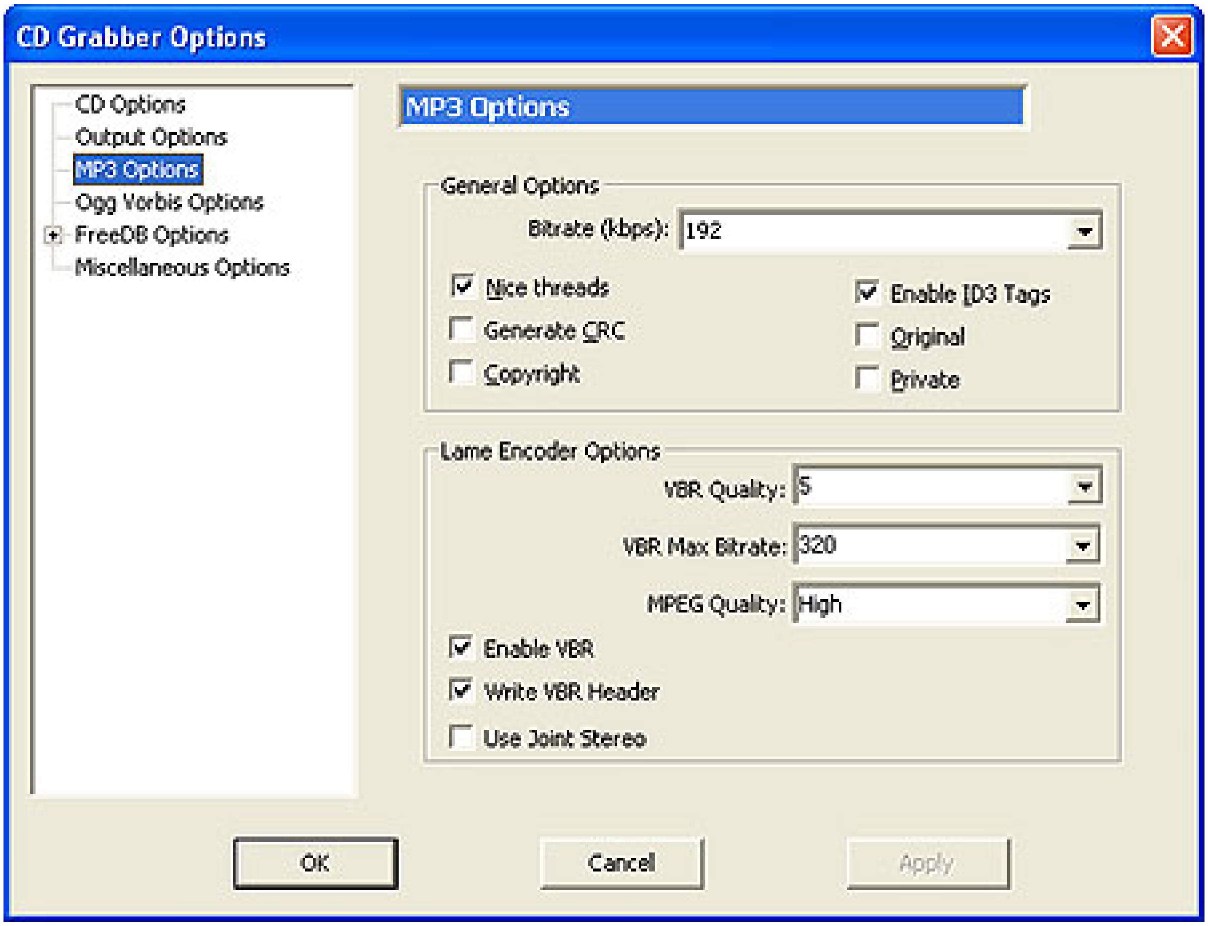


3. Check your output directory and format. The default "output" directory is the directory where you installed Audio CD Grabber. Select desired output directory by clicking the Setting->Output Options-> Browse. Desired output encoder is also selected in this window.



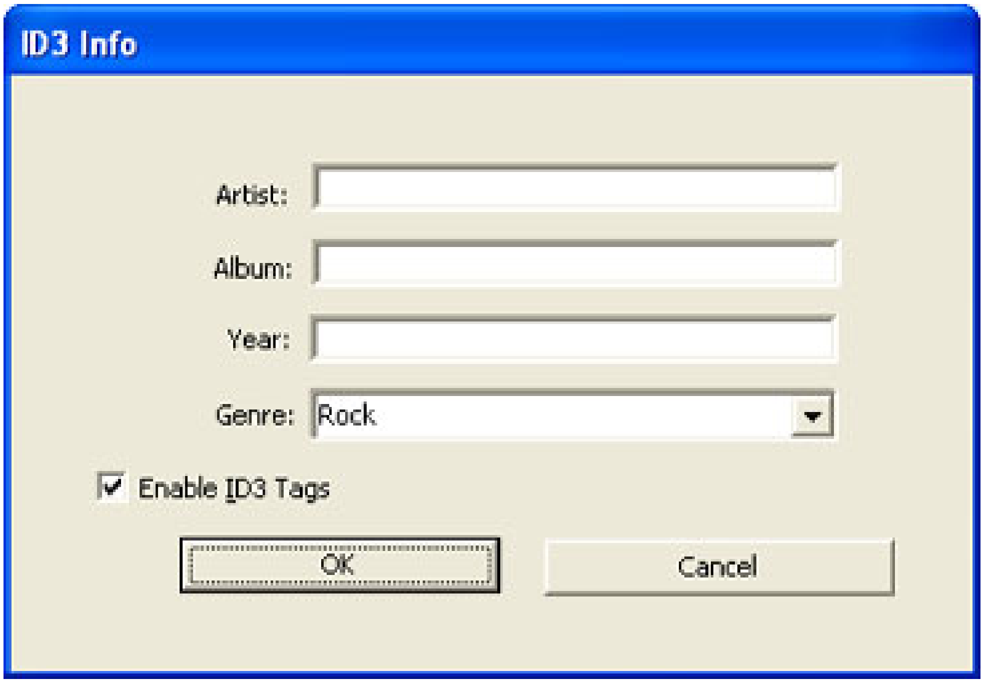
4. Check the track(s) you want to rip. Click the Select All item to select all tracks. Check the "Autocheck track" checkbox in the CD Options to automatically check all tracks.

5. Click MP3 Options to set the MP3 file properties.



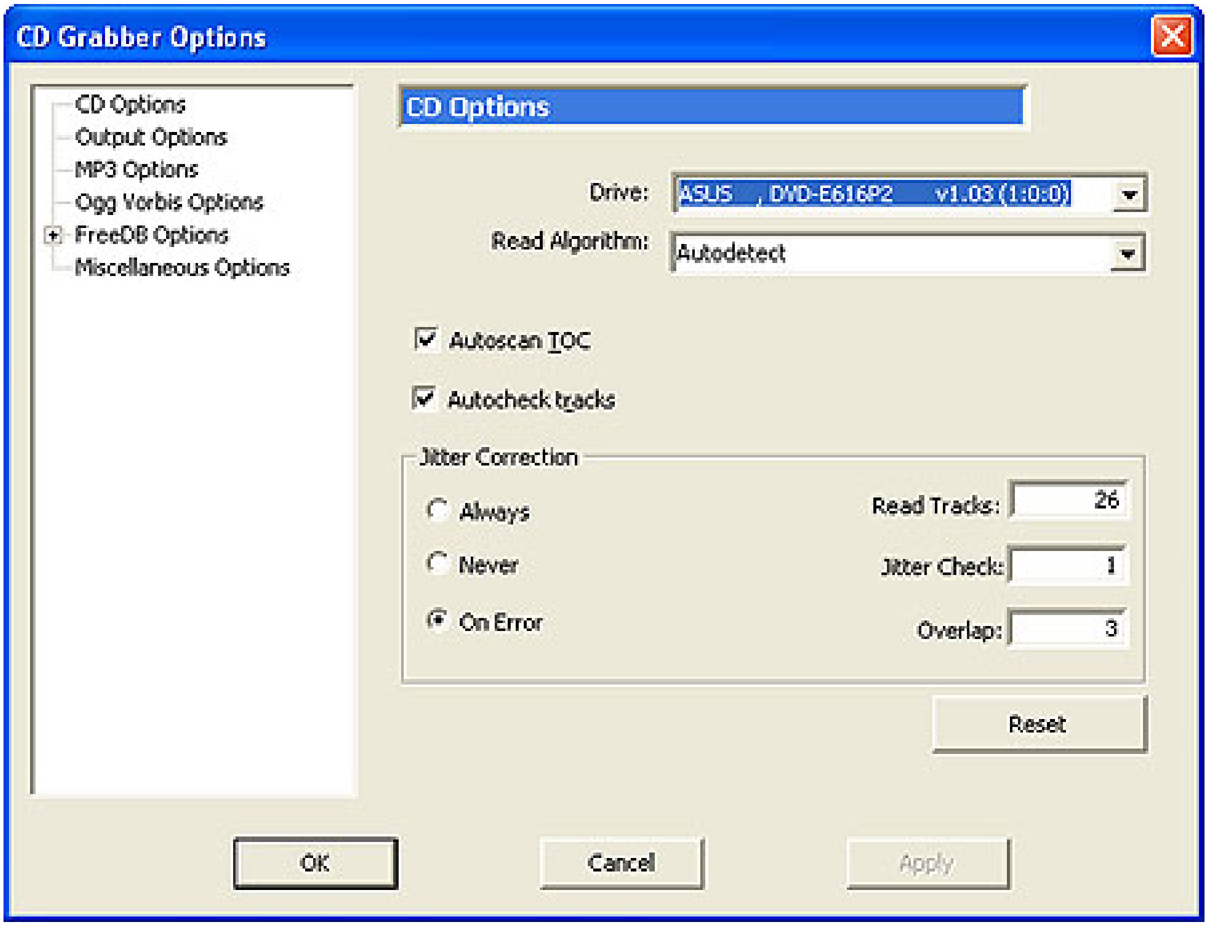
6. To edit the names of the tracks, right-click the track name and then select Rename from the popup menu.

7. If you have enabled ID3 tags in the MP3 options, you need to set the ID3 info for the CD. Just click the  button to set.



9. Click the  button to start extracting all checked tracks.

CD Options Dialog



Drive: Select desired CD-rom unit.

Read Algorithm: Selects the read algorithm used by the program. (Autodetect is mostly used). If you are unable to rip tracks using autodetect or if the sound is garbled, try the other settings.

Autoscan TOC: When checked, the program will automatically scan the CD in default drive when the program is started.

Autocheck tracks: When checked, the program will automatically check all tracks after an automatic scan of the TOC or after you press the "Refresh" button.

Jitter Correction: When selecting "Always", the program will always attempt to align the output file by using overlapped reads. For "On Error", it will attempt jitter correction only after an error is reported by the CD. "Never" disables jitter correction. "On Error" gives almost as good results as "Always", and is much faster, since the drive has to seek after every read when in "Always" mode.

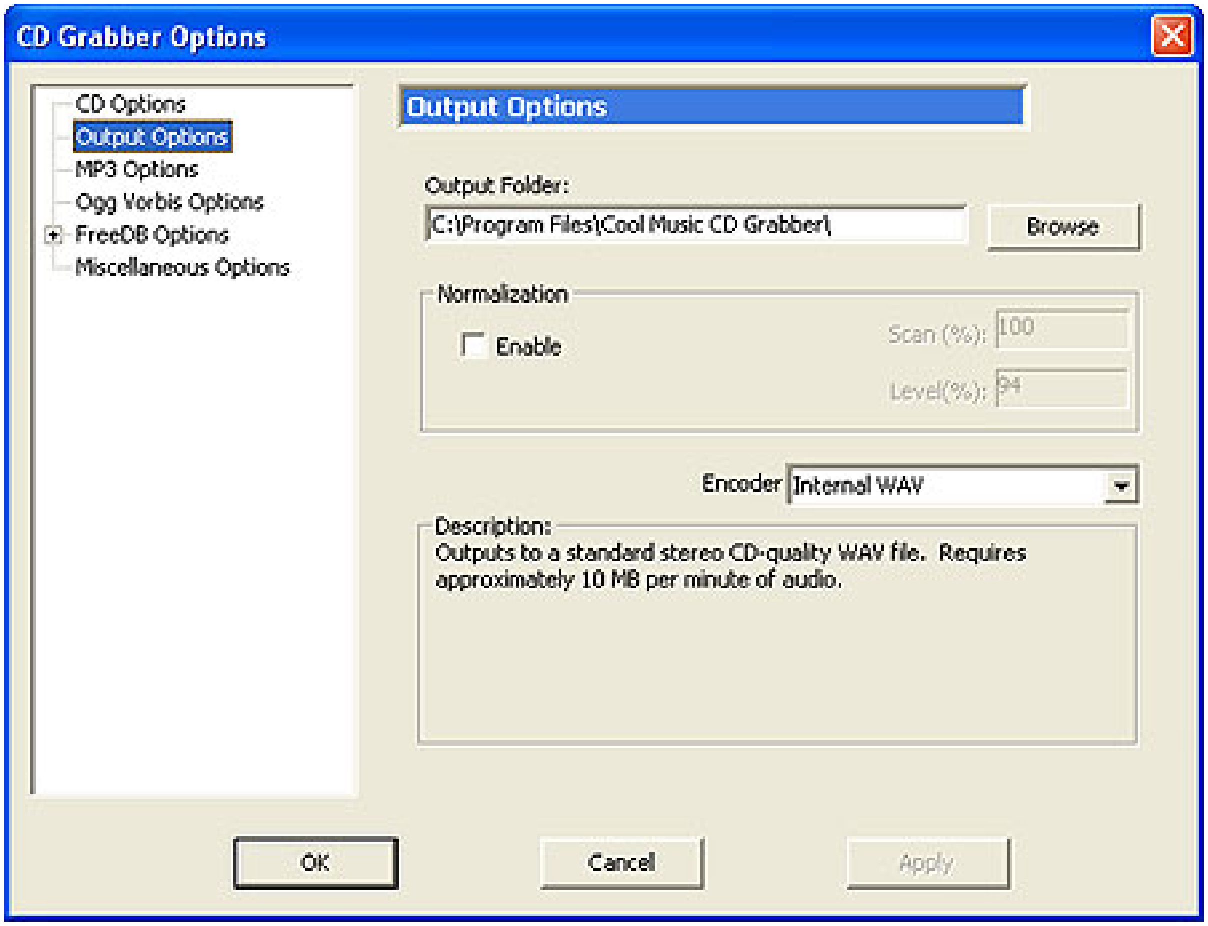
Read Tracks: Maximum number of frames that will be read at once. Some drives can handle larger values, but some can't handle more than 27.

Jitter Check: When using overlapped reads (for Jitter correction modes "always" and "on error"), this is the number of frames that the program will attempt to match.

Overlap: When performing jitter correction, this is the number of frames that will be overlapped.

Reset: Resets all the CD Options to default setting.

Output Options Dialog



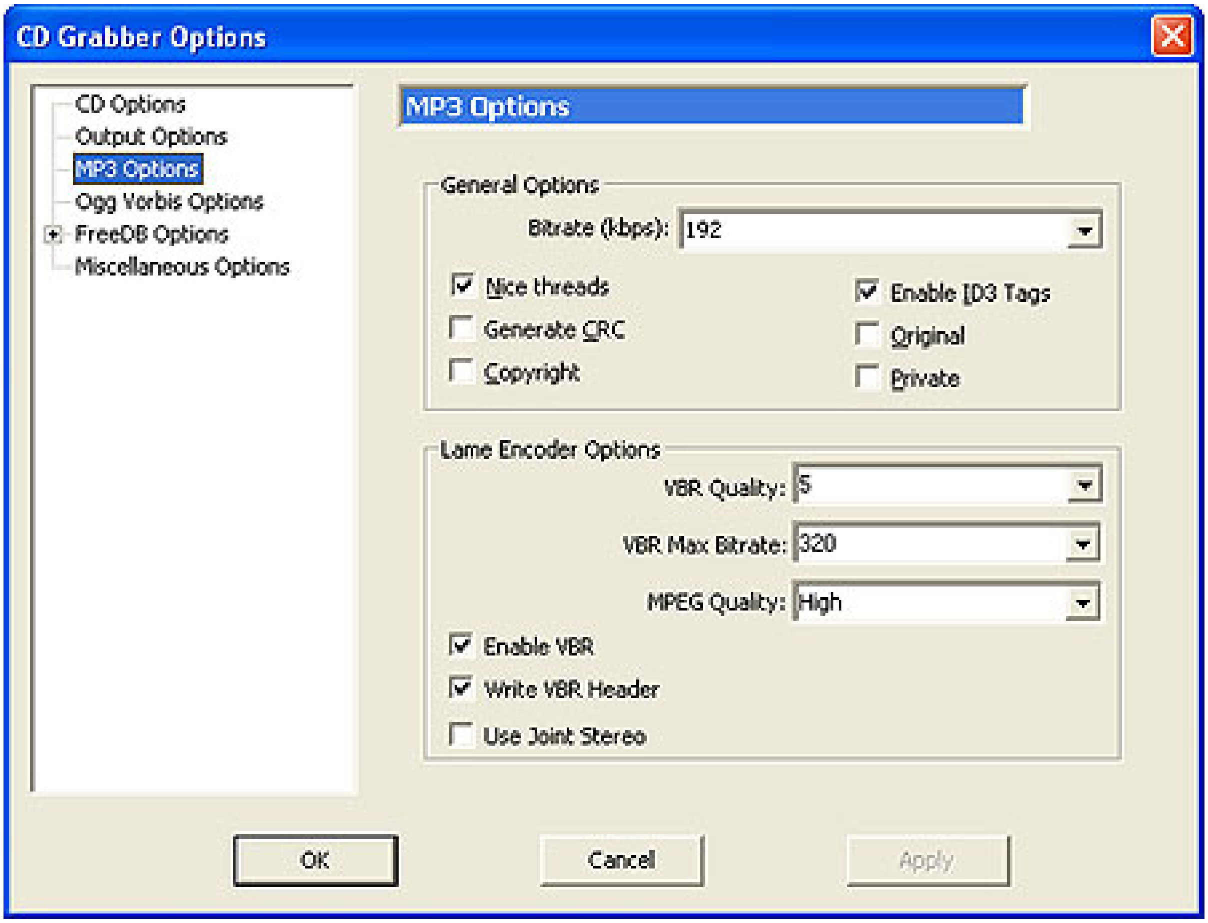
Output Folder: The directory in which output files will be saved.

Browse: Click the Browse button to select desired output directory.

Normalization: Check the box to adjust the parameter of Level and Scan.

Encoder: Select desired encoder and output format from the drop down list. Currently the Lame_Enc.dll, BladeEnc.dll and Vorb_Enc.dll encoders are supported. You may also select WAV files and raw data.

MP3 Options Dialog



General Options:

Bit rate: Select the constant bit rate for the MP3 file. Note: when using VBR(Variable Bit Rate) under LAME_ENC, this represents the minimum bit rate.

Enable ID3 Tags: When checked, an ID3v1 tag will be added to the MP3 file. If no ID3 information was input for the current CD in the drive, no tag will be written, even if the item is checked.

Generate CRC: When checked, a CRC will be generated for each MP3 frame.

Original: Set the "Original" flag.

Copyright: Set the Copyright flag.

Private: Set the Private flag.

Lame Encoder Options:

VBR Quality: A value from 0 to 9. 0 represents the highest quality, it increases the encoding time. According to the LAME documentation, in order to attain high quality files, values for VBR Quality should be 4 or less.

VBR Max. Bit rate: The maximum bit rate used when VBR is enabled.

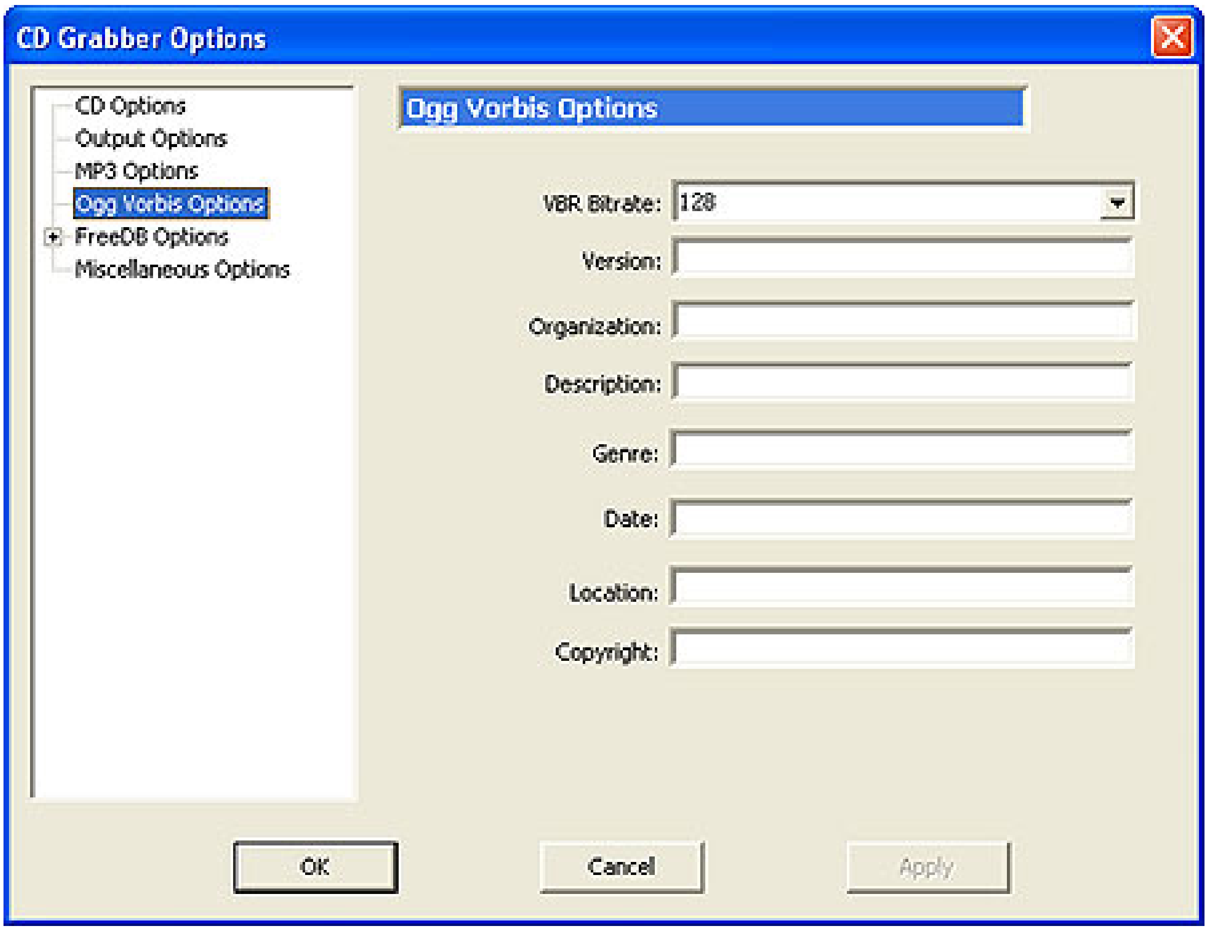
MPEG Quality: The quality of the MP3 file: Normal, Low, High, Voice.

Enable VBR: When checked, the VBR (Variable Bit Rate) option is enabled.

Write VBR Header: Causes the VBR header to be written to the MP3 file. When using VBR, this header will allow MP3 players to determine the proper length of the file.

Vorbis Options Dialog

Ogg Vorbis is a new, open-source audio codec similar to MP3 in quality and compression. However, it is not encumbered by patent issues. This means that it can be included in a freeware product without having to obtain a costly license. Also, unlike MP3, it is inherently VBR, so the encoder only uses the amount of bits it needs, resulting in smaller files of the same quality. Vorb_enc.dll, which is included in this distribution, is based on the libraries from xiph.org.



VBR Bit rate: Vorbis is inherently VBR (variable bit rate), and the bit rate that you select will be an ideal average.

Version: Used to designate multiple versions of same track and stored in the Vorbis comment header.

Organization: Publisher of disc.

Description: Short description of track contents.

Genre: The overall genre (if any) that the music belongs to.

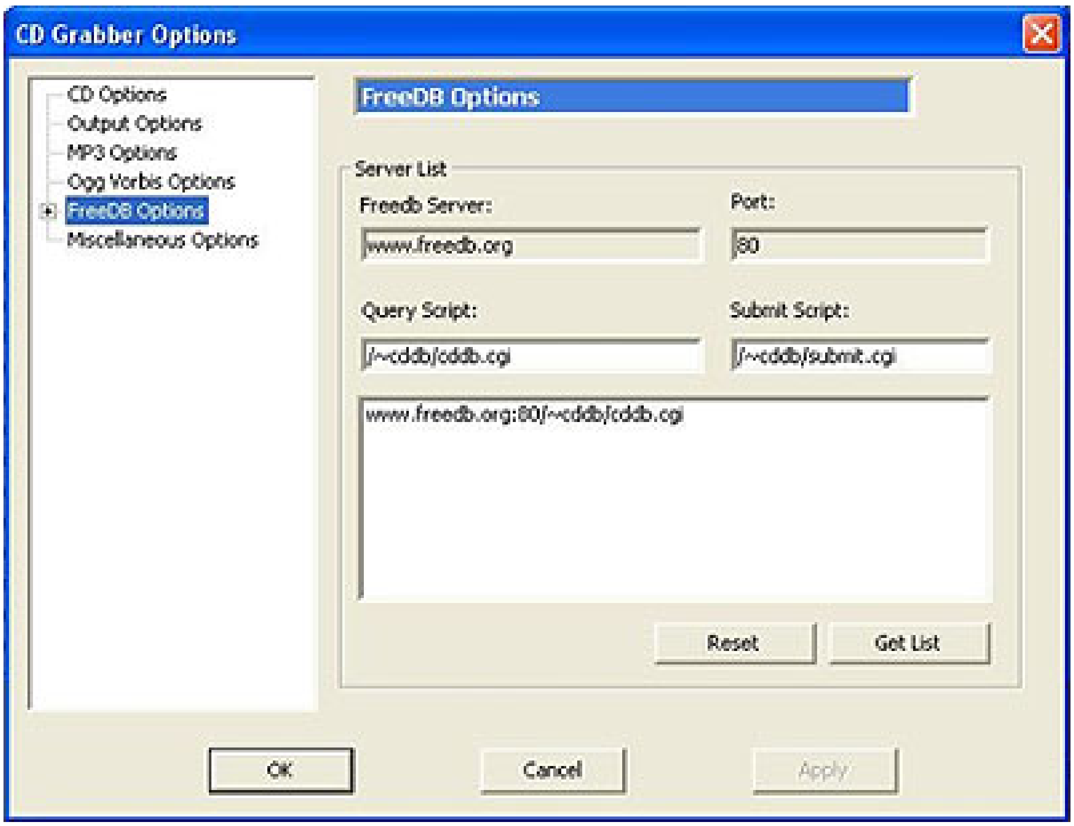
Date: Date when the track was recorded.

Location: The place to which the track was recorded.

Copyright: Copyright info for the track.

Freedb Options

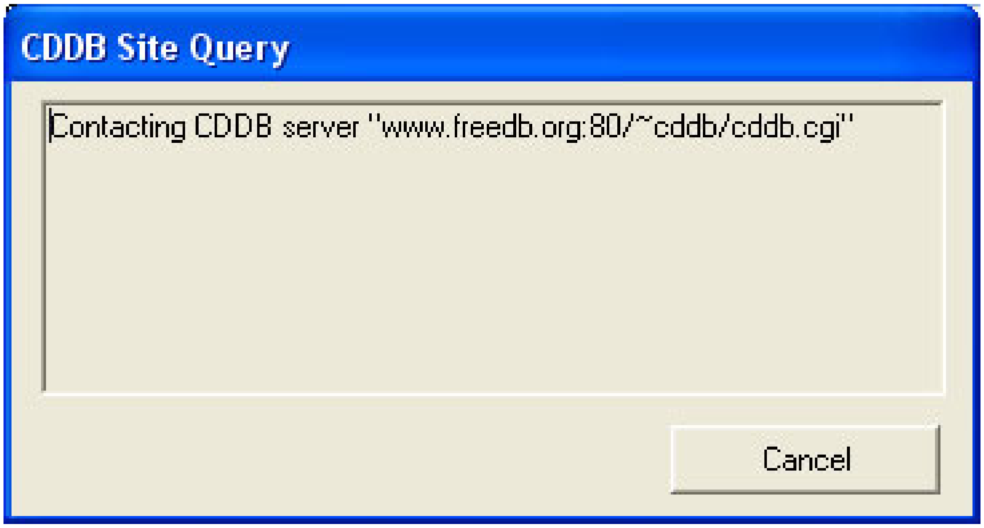
Audio CD Grabber is capable of querying the Freedb database and other databases based on the original protocol on the Internet to automatically fill in the artist name, album title and the titles of all of the songs. Check "**Enable Free DB**" on the main window to perform this feature.



Freedb Server: Address of the Freedb server to use.

CGI Script: This should contain the relative address of the CGI used for submitting queries on the server.

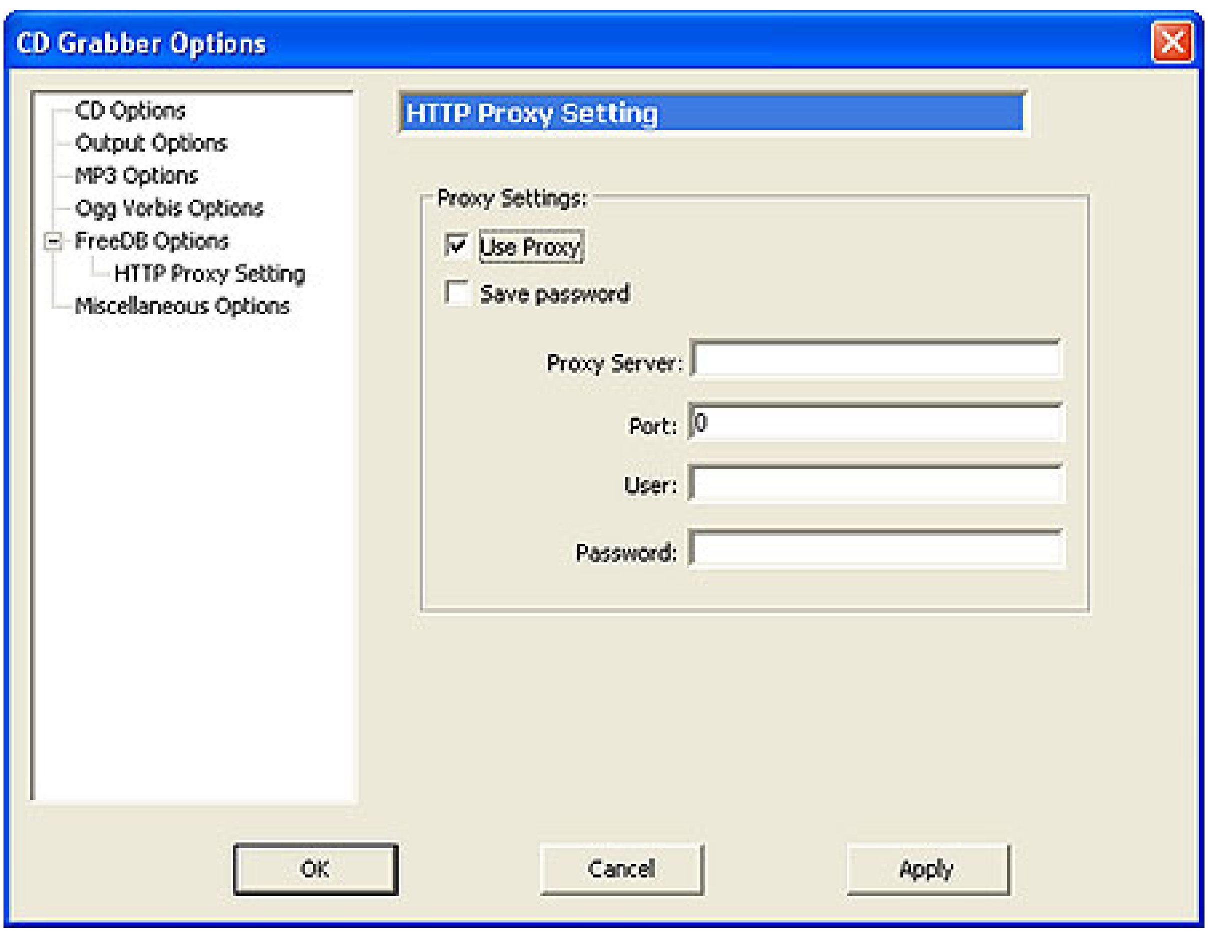
Get List: Gets the available server list from the currently configured server by clicking "**Get List**".



If you change the Freedb server, you must click "Apply" before you can use the "Get List" button.

Reset List: Reset the list of available servers to "freedb.org". Do not reset the configured server.

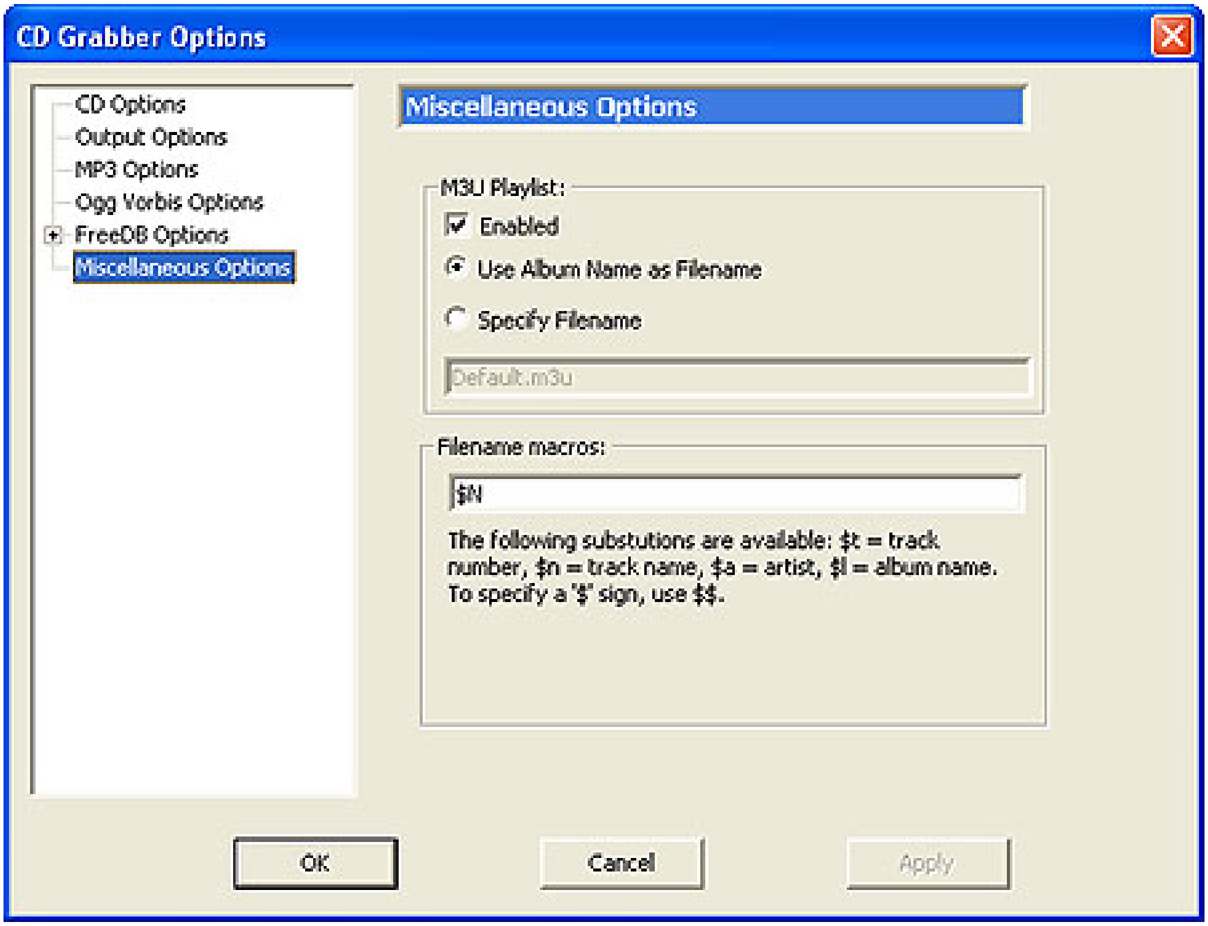
Use Proxy: If you are located behind a firewall and need to connect through an HTTP proxy, check this box.



Proxy Server: Address of the proxy server. Contact your system administrator for this information.

Proxy Port: Port on which to contact the proxy server. Contact your system administrator for this information.

Misc Options Dialog



M3U Playlist

Enabled: When checked, an M3U playlist file (used by WinAmp) will be generated.

Use Album Name As Filename: When selected, the generated M3U file will be named with the Album Name.

Specify Filename: Specify desired filename for M3U files.

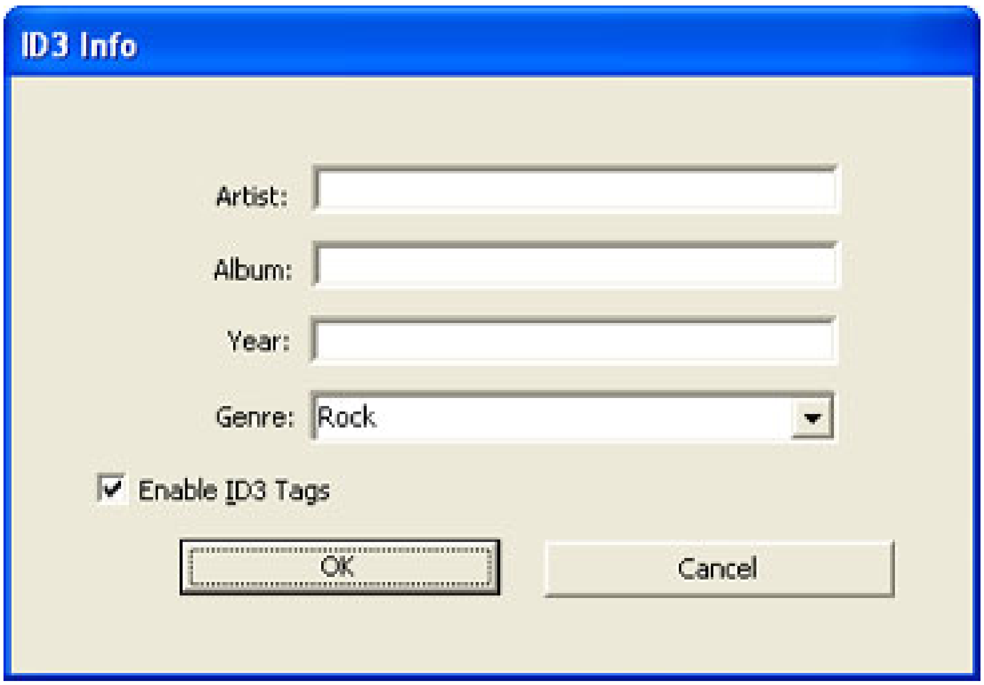
Filename Macros

Specify how the output files will be named. The mechanism works thusly:
1. If a macro is entered in the provided edit box, it will be used. "$T" will be replaced with the track number, "$N" with the track name, "$A" with the artist, and "$L" with the album title. Note: if none item is supplied, it will be simply skipped, so make sure all items are supplied.
2. If no macro is supplied, but the file has been renamed (by or manually), that name will be used

3. The track will be otherwise named as "Track x.ext" (x is the track number and .ext is the extension for the configured output format).

ID3 Tags

ID3 tagging is a method of recording information about the file such as the artist's name, album title, song title, year and genre directly in an MP3 file. Currently, only ID3v1 is supported, since most MP3 players support it. ID3v2 is still under development and will be enabled in future releases after the format is finalized. For more information on ID3, you can visit www.id3.org. The "ID3 Info" dialog is available by clicking the "**ID3 Settings**" button on the main window.



Artist: Enter the name of the artist here.

Album: Enter the name of the album here.

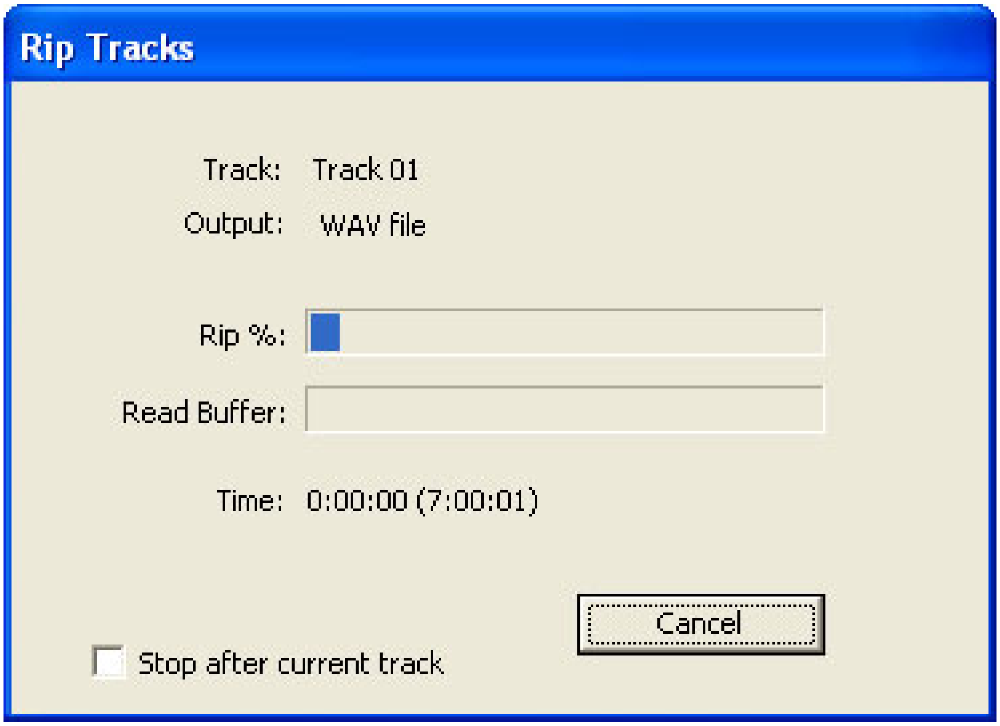
Year: The year when the track was recorded.

Genre: The genre that the track fits best.

Enable ID3 Tags: When checked, an ID3 tag will be generated for MP3 files. If the artist and album name are not filled in, no MP3 tag will be generated, regardless of the state of this checkbox.

Rip Tracks Dialog

If you need to rip just one or more tracks, click on  to bring up the Rip Tracks dialog.



The top status bar displays the current progress, and the bottom status bar displays the buffer progress. You are able to see the total time and used time of ripping current track. Check the box to stop the ripping progress after finishing the current track .

Frequently asked questions

Click on the question in the list below, to jump directly to the answer:

- I have attempted to download the software many times, but each time the download is more than the file size listed on the web site. Why am I having this problem?
- Can I record an Internet broadcast through RealAudio?
- My recordings contain pops, skips or other distortions, while the original sounds fine. What is wrong?
- I have no sound output, except from your Audio Recorder. Why is this?
- Why is the disk full so quickly when I use Audio Recorder?
- Why is the amount of available disk space not decreasing after I recorded something?
- What Can Audio CD Burner do for me?
- What is Disc-At-Once in CD Burner?
- I am using CD Burner. Has it limit the quantity of files when adding files ?
- What are ID3 Tags in CD Grabber?
- I can't rip any tracks at all. Help!
- I can rip, but it seems to rip very slowly?
- How can I tell if my CD supports digital audio extraction when using CD Grabber?
- I'm and pops in the ripped tracks. What can I do?
- Is the online order secure?
- I purchased a copy of Digital Music Record Convert Burn Station, and now I want to use it on another computer. What can I do?
- I have purchased a previous version of Digital Music Record Convert Burn Station in the past. How do I upgrade to the new version?
- Do you produce a version of Digital Music Record Convert Burn Station for the Mac?
- Can I purchase the software in a local store near where I live?
- What is the update/upgrade policy for Digital Music Record Convert Burn Station?
- What is Digital Music Record Convert Burn Station's refund policy?
- What are the recommended Operating System configurations for Digital Music Record Convert Burn Station?
- What are the hardware Requirements for Digital Music Record Convert Burn Station?
- How do I uninstall Digital Music Record Convert Burn Station?

Q: **I have attempted to download the software many times, but each time the download is more than the file size listed on the web site. Why am I having this problem?**

A: This problem seems to occur only when using a download manager with specific network conditions. If at all possible, you will have better results by downloading the software normally through your browser.

TOP

Q: **Can I record an Internet broadcast through RealAudio?**

A: Yes.
You can record from virtually any sound source that is audible through the computer speakers, including Internet broadcasts with RealAudio for example.
Just make sure to follow these steps:

1. First start play-back of the Internet broadcast in the player you are using, like RealPlayer.
2. Wait for RealPlayer to start play-back. First it will download a couple of seconds of sound, before it starts play-back.
3. Then start the Sound Recorder and select the appropriate sound source. The names of the sound sources differ from one system to another, but the sound source for recording RealAudio play-back, is usually named **Wave**, **Stereo Mix**, **What You Hear**, or something in similar wording. The Sound Recorder will automatically select the appropriate recording quality if needed.
4. Start recording in the Sound Recorder. Optionally you can restart play-back in RealPlayer, to record the whole sound clip from the beginning.

Note that on certain computers you cannot change the recording quality to anything other than that of the sound quality of the Internet broadcast.
Also on certain computers you will get an error message in RealPlayer, about not being able to access the sound card, if you started the Sound Recorder before starting play-back in the RealAudio player. In that case the selected recording quality in the Sound Recorder window does not match that of the Internet broadcast and then RealPlayer cannot access the sound card for play-back. Just follow the steps explained above, to resolve this conflict.

TOP

Q: **My recordings contain pops, skips or other distortions, while the original sounds fine. What is wrong?**

A: If your computer is sufficiently equipped (see System requirements), this could be caused by a defect in the sound card hardware, but most likely it is caused by a problem in the software driver of your sound card.
The Sound Recorder is totally depending on the quality of the sound card hardware and software driver, for the digitizing process.
Here are some recommendations and things to check, in order of likeliness that they cause the problem:

1. Check if there is an update available for the software driver of your sound card.
   Most manufacturers of sound cards have an Internet site where free updates can be downloaded.
2. Make sure that no other tasks are active on your computer, while you are recording.
   The activity of other tasks, could have a negative impact on the response time of other processes like the recording process, causing short skips and pops.
   Do not forget about the tasks sitting in the "system tray" of Windows. That is where the clock is displayed, at the bottom right hand of the screen. Usually this is where programs like virus scanners are active.
   Also, if you have a continuous Internet connection (through cable or (A)DSL for example), it is best to disconnect it while you are recording.
3. Check the hard disk light on your computer.
   While recording, it should blink just shortly every 2 or 3 seconds. If the light blinks constantly and if you constantly hear hard disk activity while you are recording, then the speed of your hard disk could be the problem.
   What might help then, is to de-fragment your hard disk. This can be done with the hard disk de-fragmentation tool that comes with Windows and it can be found in the Start menu under Programs / Accessories / System tools / Defragmentation.

TOP

Q: **I have no sound output, except from your Audio Recorder. Why is this?**

A: The Sound Recorder automatically make sure that the volume is set correctly when it starts, but other programs most times don't do this automatically.
If you don't hear any sound when the Recorder is not running, then you can manually control the sound channels on your system with the Windows volume controls.
To show the Windows volume controls, double click on the little speaker icon in the task bar of Windows (near the clock) and make sure that the volume slider for the Wave channel is not set too low or even muted.

TOP

Q: **Why is the disk full so quickly when I use Audio Recorder?**

A: Sound files are very large files. Specially sound files that are suitable for making an audio CD take a lot of disk space, because they contain the information for two-channel (stereo) sound with a frequency of 44.1 kHz.
For each second of digital sound, you need approximately 172 Kb of free disk space. That is approximately 10 megabyte for each minute.
Please note that the actual amount of required disk space depends on the selected recording quality.
As soon as you have recorded the sound files on an audio CD, you can of course remove the files from disk again.

TOP

Q: **Why is the amount of available disk space not decreasing after I recorded something?**

A: A new recording is always stored on the hard disk as a temporary file.
While recording, the amount of available disk space on the disk that is used to store the temporary file will decrease.
This is indicated in the Sound Recorder window, as the time that the new recording can last before the disk will be full.
After recording, when you save the new recording to *another* disk with the Sound Editor, the used space will be released on the disk that was used to store the temporary recording file and this space will then be reused for the next recording.

TOP

Q: **What Can Audio CD Burner do for me?**

A: Audio CD Burner is a MP3/WMA/WAV to Audio CD converting software. It burns your MP3/WMA/WAV collections to make a normal Audio CD which can be played on your home, car stereo or portable CD player.

TOP

Q: **Is the online order secure?**

A: Yes, we use the online ordering services provided by Element5, which is a famous leading shareware ordering service company. You can click here to get more information.

TOP

Q: **What is Disc-At-Once in CD Burner?**

A: DAO is a method of writing CDs in which one or more tracks are written in a single operation, and the CD is closed, without ever turning off the writing laser. Audio CD burner writes Audio CD in DAO mode. Audio CD burner automatically supports DAO mode.

TOP

Q: **I am using CD Burner. Has it limit the quantity of files when adding files?**

A: This software hasn't limited the quantity, but the total duration of all listed files can not exceed 80min. Therefore, you should pay attention to this point when adding files.

TOP

Q: **What are ID3 Tags in CD Grabber?**

A: A: ID3 tags are information about an MP3 file, such as the Artist, Album, Title, Year, Track, Genre, Comments, Lyrics and even custom comments. ID3 tags are optional information that can be added when the audio file is created (encoded). The following Tag fields are available for you to track important music information. Each music file can contain info in every available field.

Artist - The name of the Artist who performed the song.
Album - The name of the CD (not always the name of the artist).
Genre - The style of music this song is i.e.....Rock, Rap, Classical etc.
Year- Year of publication of the music.

TOP

Q: **I can't rip any tracks at all. Help!**

A: Try resetting the jitter correction values to their defaults in the CD Options dialog (click on the "Reset" button). If that doesn't work, In the CD Options dialog, try using the "Auto detect" read algorithm. If it still doesn't work, try the others individually. If none of this helps, send an email to support@audioeditor.us with the make and model of your CD rom, operating system, whether it's a SCSI, ATAPI, CD-R, etc. and a description of the problem. It's also possible that your drive simply doesn't support digital audio extraction -- while most SCSI drives support digital audio extraction, not all IDE drives do.

TOP

Q: **I can rip, but it seems to rip very slowly.**

A: Try checking the "On Error" jitter correction in the "CD Options" page. This mode is faster than the "Always" setting, and should produce results comparable to "Always".

TOP

Q: **How can I tell if my CD supports digital audio extraction when using CD Grabber?**

A: If Audio CD Grabber doesn't seem to work with your CD, try a different program -- you can find many shareware and freeware programs at http://www.audioeditor.us.

TOP

Q: **I'm still getting clicks and pops in the ripped tracks. What can I do?**

A: Go to the CD Options page by selecting the Settings->CD Options item. Try using the "Always" setting for jitter correction -- this setting will attempt to correct inaccuracies in the positioning of the read regardless of whether the CD reported an error or not. If you receive errors, try increasing the overlap a little; this will increase the size of the buffer that is used to align the read, and so increases the chances for successful alignment.

TOP

Q: **I purchased a copy of Digital Music Record Convert Burn Station, and now I want to use it on another computer. What can I do?**

A: The software's license is for a single computer. If you'd like to use the software on an additional computer, you have to purchase another copy from http://www.audioeditor.us/purchase.htm

TOP

Q: **I have purchased a previous version of Digital Music Record Convert Burn Station in the past. How do I upgrade to the new version?**
Why is this?

A: You may upgrade to the new version of Digital Music Record Convert Burn Station for free from the following URL: http://www.audioeditor.us/download.htm

TOP

Q: **Do you produce a version of Digital Music Record Convert Burn Station for the Mac?**

A: No. Digital Smart Software does not develop Macintosh software at this time. Unfortunately, as we are not familiar with the Macintosh software market, we do not know of a solution to recommend for similar operations on the Mac platform.

TOP

Q: **Can I purchase the software in a local store near where I live?**

A: Digital Music Record Convert Burn Station is not currently sold through retail distributors. We are using the try before you buy method of distribution at present which allows users to install the software and ensure it's what they're looking for before having to spend any money whatsoever. The software can be purchased from anywhere in the world, though, directly from the Digital Music Record Convert Burn Station web site or via postal mail, fax, phone, or wire transfer.

TOP

Q: **What is the update/upgrade policy for Digital Music Record Convert Burn Station?**

A: You may upgrade to the new version of Digital Music Record Convert Burn Station for free from the following URL: http://www.audioeditor.us/download.htm

TOP

Q: **What is Digital Music Record Convert Burn Station's refund policy?**

A: Once we confirm the problem comes from our products and our staff could not fix it, the refund will be made within 1 business day.

TOP

Q: **What are the recommended Operating System configurations for Digital Music Record Convert Burn Station?**

A: Windows 2000, or Windows XP, Windows 2003 and Windows Vista.

TOP

Q: **What are the hardware Requirements for Digital Music Record Convert Burn Station?**

A: Minimum:

Intel® Pentium ® class 400 MHz processor or better; 128 MB RAM; 30 MB Hard Drive space; 12x CD-ROM drive or better (optional); Sound Blaster-compatible sound card and speakers/headphones; SVGA or higher color video display card (minimum resolution 800x600); Internet connection for ordering, and support.

Recommended:

Intel® Pentium® III class 800 MHz MMX or better; 256 MB RAM or more; 50 MB Hard Drive space; 48x CD-ROM drive; Sound Blaster-compatible sound card and speakers/headphones; 16-bit color video card; Internet connection for ordering, and support.

TOP

Q: **How do I uninstall Digital Music Record Convert Burn Station?**

A: To uninstall Digital Music Record Convert Burn Station, click Start from the Windows Taskbar, go into Settings/Control Panels/Add/Remove Programs and select Digital Music Record Convert Burn Station from the Install/Uninstall tab.

TOP

System requirements

To be able to utilize Digital Music Record Convert Burn Station, your computer system must meet the following specifications:

1, Microsoft Windows 2000/XP/2003/Vista (all of the last updates installed are recommended)

2, Intel or AMD or compatible processors at 1000 MHz minimum

3, RAM 128 megabytes (MB)

4, Minimum 50 MB of free hard disk space (When you create an MP3 CD with folders and subfolders, it is required to create a disc image on your hard drive first. For this purpose you need to have approximately 700MB of free space on your hard drive for proper program performance.)

5,Processor, your system must be equipped with at least a 300 MHz processor and 64-Mb memory, to be able to process the enormous amounts of data without problems.

6, Administrative permissions are required for installation

7,CD burner with software

8,Free drive space

9, Sound card

10, CD-RW drive

About Digital Smart Software

Take the lead in multimedia;
Create a Colorful Life!

Digital Smart is a software company founded in 2001. We create Windows based applications. Our goal is to make powerful and easy-to-use applications for home users, professionals and companies.

All our products are Shareware, which is a means of distributing software on a "try before you buy"basis. With this method we are sure that the program you buy meets your needs. If you're interested with a program of our catalog, you can buy it directly from Element5® - using your credit card.

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Digital Music Record Convert Burn Station
Product ID: 300183448

We always do our best to answer your question!

For Partners
support@audioeditor.us

'We benefit together with our partners in a win-win model - the Digital Smart Software Business Model, and we love to talk about creating with you.

Digital Smart Software

AT DIGITAL SMART, THE CUSTOMER COMES FIRST!

http://www.audioeditor.us

Appendix

An important stage of mp3 authoring is specifying the MP3 setting. This involves multiplexing the project’s channel, encoding selecting, VBR and etc into MP3-compliant form. This section describes the various output options in MP3 setting.

The following table documents contain options for specifying Channel Mode and Encode Quality. These options should be configured according to personal preference or hardware requirement.

Channel

Channel	Description
Mono	In this mode, the encoder makes no use of potentially existing correlations between the two input channels. It can, however, negotiate the bit demand between both channels, i.e. give one channel more bits if the other contains silence.
Joint stereo	In this mode, the encoder will make use of a correlation between both channels. The signal will be matrixes into a sum ("mid") and difference ("side") signal. For quasi-mono signals, this will give a significant gain in encoding quality. This mode does not destroy phase information like IS stereo that may be used by other encoders. This setting can be used to encode DOLBY ProLogic surround signals.
Forced Joint Stereo	This mode will force MS joint stereo on all frames. It's faster and it uses some special mid and side masking threshold.
Dual Channels	In this mode, the 2 channels will be totally independently encoded. Each channel will have exactly half of the bit rate. This mode is designed for applications like dual languages encoding (for example: English in one channel and French in the other). Using this encoding mode for regular stereo files will result in a lower quality encoding.
Stereo	This option will generate a mono file, if the input file is a stereo file, the input stream will be down sampled to a mono file by averaging the left and right channel.

VBR Coding Method

Coding Method	Description
None	Do not use VBR, it is encoding with a Constant Bit Rate (CBR).
Default	Use the default VBR method (currently set to VBR-MTRH).
Old	The functional approach, based on masking, bisection in the bit domain .
New	The approach, based on masking and direct noise allocation .
MTRH	A merger of old and new (VBR) routine.
ABR	The Average Bit Rate (ABR) setting, the encoding principle is based on perceptual entropy, but more like CBR than VBR.

Maximum VBR Bit rate: Allows to specify an maximum bit rate when using VBR (Variable Bit Rate), this selecting depends on what base bit rate you have chosen in the main encoder tab. It's recommended to leave this set to 320 kbit/s unless you want low quality VBR files. Please do not worry if you haven't received the registration information right away. Usually it happens due to too secure settings of spam filters used by our clients. And it may happen so that our message is rejected as a spam message by the mail service you use.

VBR Quality: In VBR mode, you are able to specify a quality setting which will affect encoding bit rate allocation. If you use quality 0, the max bit rate will be reached easily, while using quality 9 the bit rate usually will be around the base bit rate. The lower the VBR quality value, the better the audio quality, but also the bigger the output file. Recommended setting for high quality VBR encoding is 1 or 0.

Write VBR Header: This tag is embedded in frame 0 of the MP3 file. It lets VBR aware players correctly seek and calculate playing times of VBR files.

Target bit rate for ABR: The allowed range of the ABR bit rate is 4 - 310 kbit/s, you can use any integer value within that range.