

Audiograbber

What is Audiograbber and what can it do?

Audiograbber is a program that copies music and stores it on your computers harddisk. Its main purpose is to copy music from CD's. This process copies the music digitally and thus you will get perfect copies. It includes functions to check that the copies made are true to the original. Not all CD-ROM drives can however read digital audio so there is also an option to copy the music through the soundcard with a slight sound degradation. It is also possible to copy sound via the soundcard from an external source, like a turntable or a radio.

Audiograbber can connect to a database (freedb) on internet and download and upload disc information like tracknames etc. It has a normalize function to make tracks from different cd's sound equally loud. Tracks can be saved as wav files or converted to MP3 or WMA files with external programs like [Fraunhofers L3enc](#) or internal codecs like [Fraunhofers acm CODEC](#), [BladeEnc's freeware MP3 DLL](#), the [LAME freeware MP3 DLL](#), or Microsofts [Windows Media Audio](#) codec.

System specifications:

- Windows 95 / 98 / ME / NT / 2000
- CD-ROM connected to IDE or SCSI.
- A CD-ROM reader that can actually read digital music (Not all of them can) is useful but analog sampling is also possible. A soundcard is however needed for analog sampling.

Choose [Contents](#) in this help file for further information on Audiograbber.

[What's new? See version information](#)

[The difference between audio CD's and normal data CD's.](#)

The latest version of this program can be found at:

<http://www.audiograbber.com-us.net> (primary site)

<http://www.dezines.com/audio/> (mirror site)

I can be reached on the following e-mail address.

Please read the [hints](#) section before sending me any e-mails asking for support.

jackie@audiograbber.com-us.net

The Problem with reading digital audio under Windows 95 OSR2 release and Windows 98

The latest version of Windows 95 (known as the OSR2 release or windows 4.00 build 1111) and Windows 98 normally can't read digital audio through MSCDEX calls (There is no problem with the ASPI calls however.) The problem has been located to the scsi1hlp.vxd file. Even though the name suggests that the file should only be used for SCSI devices, it actually handles IDE CD-ROM's as well.

Either way, in most cases it helps if the file Scsi1hlp.vxd is replaced with the old version.

To check which version of Windows 95 you are running, do one of the following:

- **Settings, control panel, system.** If it says Microsoft Windows 95 4.00.950 B it means you have the OSR2 release. Note the "B" after the version digits. There is also a version 4.00.950a that doesn't work either.
- **Audiograbber, help, System information.** If it says build number 1111 it means you have the OSR2 release.

To check which scsi1hlp.vxd version your computer is using, do one of the following:

- **Explorer**, right click on the file C:\windows\system\iosubsys\scsi1hlp.vxd. Click on properties and then version.
- **Audiograbber, help, system information.** The version number is written to the left of Scsi1hlp.vxd version.

It doesn't matter which Windows version you are running as long as the right scsi1hlp.vxd file is used. If the CD vendor has provided its own drivers for the CD, it may happen that it works with the wrong scsi1hlp.vxd version. If you are using another version than 4.00.950 and it is not working, I recommend that you replace it with the 4.00.950 version. The file is located in the C:\windows\system\iosubsys directory. I suppose that Microsoft doesn't want me to distribute one of their files so I can only give you a link to Sony's site with the file:

<ftp://ftp.sony.com/ccpg/pc/scsi1hlp.exe>

If it is not found on that site, try to find it on the Internet. Search for scsi1hlp.exe or scsi1hlp.vxd. A good search engine is

<http://ftpsearch.ntnu.no/ftpsearch>

Make a backup copy of your scsi1hlp.vxd before you replace it. Beware of renaming it to scsi1hlp.vxd1111 or something like that. When Windows uses this file, it seems to search for a file that loosely matches scsi1hlp.vxd. Rename it to osr2vxd.bak or something similar.

You have to restart Windows after you have replaced the file!

The difference between audio CD's and normal data CD's.

(Or why is it so hard to copy a track from a music CD?)

A CD-ROM is divided into sectors of 2352 bytes. Data CD's use 2048 of these bytes to store the data file. The other bytes are start / stop information and information of the sector's number. This makes it easy for a computer's CD-ROM device to find the right sector. An audio CD on the other hand uses all of the 2352 bytes to store audio information. This makes it hard for the CD-ROM reader to find exactly where the track starts. All bytes are coming in one long sequence. When the CD drive starts reading it usually has no problem delivering the right data but most CD drives have problems with starting on the correct byte (sample).

A sector (it is called a frame on audio CD's) on an audio CD, as stated earlier, is 2352 bytes. There are 75 frames per second which gives $2352 * 75$ bytes per second of music. (This can also be calculated by 2 (stereo) * 2 (16 bits) * 44100 Hz per second). Philips "Red Book" standard specifies that a CD player should be able to position its head on the right frame but not where on the frame.

Since a computer program has to read a little piece of the track, write it to the hard disk, read another piece etc. there will be problems all the time. The program solves this problem by reading a little bit more data than is written and then compares the end of the previous reading with the beginning of the present reading and in that way can synchronize the readings. (This is called overlapping, synchronization or jitter correction).

Audiograbber can use two different ways of reading the audio, [MSCDEX](#) or [ASPI](#), of which ASPI is the preferred one. ASPI can be used in a kind of multi threaded mode which means that the program can ask for data and get a notification back when the data has arrived. This way the program can do other things while waiting for data and it is often possible to not use any synchronization at all, the data can be copied correctly in a burst copy mode. If the drive spins faster then the computer can handle the data there will be "[possible speed problems](#)".

In MSCDEX mode you can select how many frames of audio should be read in every chunk and with how many frames overlapping. The program does not always get enough memory for all the frames it wants which means that it sometimes has to use a smaller audio buffer. (Under MSCDEX the program uses the low DOS memory to buffer data from the CD drive and depending on what other device drivers and programs want memory from that area, it can differ a bit from time to time or computer to computer) The more sectors that are used for overlapping, the slower the CD driver will read. [When windows 95's protected mode CD-ROM driver](#) is used, the data from the CD drive is always cached and it seems to make the synchronization unnecessary. If a [real mode driver](#) is used for the CD drive and if the data is not cached the synchronization is necessary.

Whether the CD drive reads the audio perfectly can be tested by:

- [Compare two files](#)
- [Calculate checksum](#)

Windows 95 protected mode driver and real mode driver

Before Windows 95, all CD's read data through 16 bits real mode drivers and mscdex.exe. Those drivers were delivered with the CD player and were installed in config.sys and autoexec.bat. Windows 95 has built in support for CD-ROM and the real mode drivers are not normally needed. Anyhow, there are some cases when digital audio can not be read, or is not read correctly with Windows 95's internal driver. If so, you can try to use a real mode driver instead. Under the OSR2 release of Windows 95 the file Scsi1hlp.vxd often causes digital audio to be not read at all. You should then replace it with an older version.

- [How to install a real mode driver](#)
- [How to replace the Scsi1hlp.vxd file](#)

Note that this is old stuff only valid for MSCDEX as read method. Newer CD-ROM drives and computers usually work better with ASPI as read method.

Real mode driver installation

This is a bit complicated and if you don't know what you are doing pretty well, you'd better avoid doing this.

- Install the CD-ROM drivers in config.sys and autoexec.bat. Make sure that you can read an ordinary data CD when you are booting into DOS mode.
(config.sys looks like this
DEVICEHIGH /L:1,15488 =C:\MTM\MTMCDAI.SYS /D:MTMIDE01
and autoexec.bat like this
LH /L:1,46672 C:\WIN95\COMMAND\MSCDEX.EXE /D:MTMIDE01 /M:10)
- Connect the CD-ROM unit to secondary IDE and not together with any hard disk.
- Open Control Panel, choose System and the tab Device Manager.
- Click "View devices by connection"
- Locate Secondary IDE controller. (not the CD itself but the closest higher connection).
- Select the checkbox "Disable in this hardware configuration".
- Click OK. It sometimes happens that the computer crashes at this moment so make sure you are not running any other programs.
- Reboot the computer.

To restore Windows 95's protected mode driver just unselect the checkbox instead.

How to buy the full version of Audiograbber?

The price is \$25. Go to <http://www.audiograbber.com-us.net/buy.html> for information on how to buy Audiograbber online.

If you like the program and can afford the \$25 please buy it. I have spent a lot of time on this program and more money makes it easier to continue the development of Audiograbber!

If you are young and can't afford the \$25, go on and use it anyhow. You are still allowed to use the free version as much as you want!

The difference between the full version and the freeware version.

There is only one difference between the free and the full version. The free version has been limited to only handle half of the tracks and those tracks are selected randomly at program startup (with a slight advantage for odd tracks). By restarting the program the checkboxes will be in front of other tracks. The full version can of course handle them all and has checkboxes in front of every track.

[Here is how to buy the full version](#)

Version information

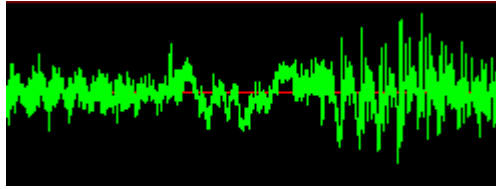
Version history:

- **Version 1.80** 17 March 2001
[ID3v2](#) support.
Switched from CDDb to [freedb](#).
- **Version 1.70** 27 September 2000
Rip to RAM option added, makes ripping smoother.
Better support for compilation discs.
Native disc database and CDDb batch query function.
Better MP3 encoder settings and support for OGG vorbis.
- **Version 1.62** 24 February 2000
Support for native ripping under Win NT/2000. An ASPI manager is no longer needed.
- **Version 1.61** 16 January 2000
[Localization](#) with support for external language files.
Improved normalization, can now normalize files together as one file.
- **Version 1.60** 12 October 1999
[Line in sampling](#) function.
Can now finally rip the last audio track on multimedia discs without problems.
Possibility to rip all tracks before encoding.
Can append id3 tags to wav files. Good to use if the files are going to be encoded at a later time.
Can rip parts of tracks much easier.
- **Version 1.50** 29 May 1999
Freedb [Submit](#) function.
[Windows Media Audio](#) (wma) support.
[Analog sampling](#) from CD-ROM drives.
Remembers the previous window size and position.
Can run [multiple](#) instances.
Can now rip multimedia discs under MSCDEX.
- **Version 1.41** 17 March 1999
Direct rip and encode options.
Better detection of drives that uses inversed byte order.
Integration with MP3 players.
An option to auto-query freedb if a disc is not recognized.
Speed selection for Plextor drives.
- **Version 1.40** 10 December 1998
Support for [BladeEnc's](#) MP3 DLL.
[Enhanced normalizing and sound compression](#) options.
Better naming options.
Better support for external programs.
- **Version 1.31** 25 August 1998
Support for Philips, HP and new Toshiba SCSI drives.
Improved speed selection for IDE drives.
Added a spin up disc option.
Improved the volume meter.

- **Version 1.30** 17 July 1998
 Added freedb retrieve support.
 Better naming options which give more flexible control over the filenames.
 Added a progress bar for the played track on the main window. A volume meter is also added.
- **Version 1.21** 10 June 1998
 Added support for Panasonic (Matsushita) SCSI drives.
 Implemented a new rip method (Unbuffered Burst) and renamed the old Burst Copy method to Buffered Burst.
 Implemented an unlock method for the full version to prevent software piracy.
 Fixed a few minor bugs.
- **Version 1.20** 22 April 1998
 Added ASPI as read method. This enabled Audiograbber to work with SCSI cd's and Windows NT.
 Support for setting a rip offset has been added. This helps CD-ROM's that don't position their head exactly correct.
 Added a tip of the day function.
 An option to let Audiograbber shut down the computer when finished has been implemented.

 A way to easily see how fast a track was ripped.
- **Version 1.10** 15 February 1998
 Added support for Fraunhofers internal MP3 [CODEC](#).
 Better ID3 tag options.
 Some bugs fixed, (and undoubtedly some new created !)
- **Version 1.01** 6 January 1998
 Selects which tracks that can be copied randomly instead of only the odd ones.
 Supports Wav2mp3 for MP3 creation.
 Two new check boxes so that one can easily see if normalization / MP3 creation are selected.
- **Version 1.00** 24 December 1997
 The first version was released.

Normalizing



This function is found under the file menu.

Normalizing means how loud a track is recorded. Different CD's, just like LP's from the old days, are usually recorded at different volumes. All tracks on the same CD are individually, recorded at about the same level and as long as one listens to only one CD at a time there is usually no problem with the loudness of the CD.

The problem starts when you mix tracks from different CD's because they occasionally have an output level that differs from one CD to another. One CD can play quite loud and another noticeably quieter. The normalize function comes in handy in these cases and you can select the same level for all your CD's.

Most of the CD's are properly recorded and the normal level is usually between 95-99 %. By choosing to normalize only the tracks normalized to less than about 91 %, your result will be that all the songs will have approximately the same volume and most tracks will be left in their original condition. These percentage values detect when the sound is loudest on the track. Normalize to 0% means absolute silence.

It is also possible to normalize all files together as if they were only one file. That can be useful if you rip a full disc which hasn't been mastered in a good way by the record company. Normally you shall normalize the files without checking this option though. This option is found on the open dialog when you browse for the files and it can't be used for advanced normalizing or when you drag and drop files on Audiograbber.

From version 1.40 of Audiograbber you can also use a much more [Advanced Normalizing](#) routine which also includes sound compression options to ensure that all your songs will sound equally loud.

It is possible to normalize the tracks right after they have been grabbed from the CD or normalize already stored .wav files from the hard disk.

If you want to normalize wav files you can then choose files from the file menu, directly from the normalize dialog box or by dragging and dropping them from Windows explorer.

Make MP3.

This function is found under the file menu.

An audio file on a CD takes quite a bit of space to store (exactly 1411200 bps), and that's the reason that only a few tracks can be stored on a CD. MP3 is a standard that compresses the music to make it take about 1/11 of the size and still sound as good as the original. The disadvantage with this file format is that it has to be decompressed when playing. This means that you must play it on your computer. It can normally not be played on an ordinary home stereo cd player.

Many major electronic companies are now making devices for playing MP3's on ordinary stereos etc. The most successful company this far is Diamond Multimedia and their RIO MP3 player (A Walkman type MP3 player that can hold up to 20 songs of decent quality).

MP3 stands for mpeg audio layer 3 and should not be written mpeg3 as some people do by mistake. Mpeg is lossy compression, and that means that the file has lost information and can no longer be restored to it's original content. Audiograbber can use a variety of external programs like [Fraunhofers L3enc](#) to create such MP3's.

If your external command line program does not automatically close when finished you can fix that this way: Start Windows Explorer and right click the command line program (l3enc.exe for example). Choose properties -> program and tick the box "close on exit".

Audiograbber also supports [LAME's and BladeEnc's dll's](#) and [Fraunhofers CODEC](#) to create MP3's. This means that it looks as if Audiograbber is doing the compression itself with a progress bar and time estimating. It is still Fraunhofer technology that compresses the wav though. This CODEC is a lot faster than the DOS program L3enc. Other codecs that are supported internally in Audiograbber are QDesigns MP2 codec, Microsoft WMA and [OGG Vorbis](#).

Audiograbber can automatically send the tracks to an external MP3 program right after they have been grabbed. First the track is copied to the hard disk; then it is sent to the external MP3 program and Audiograbber waits for it to finish. Audiograbber can then delete the wave file from the hard disk to release valuable disk space and continue with the next track. One of the main purposes with Audiograbber is to create MP3's when you are doing something different (like having a cup of coffee).

Many people find L3enc and other command line based MP3 encoders a bit complicated to use since it is not a Windows GUI program. For that reason, some front-end programs have been created to use L3enc easily under Windows. Audiograbber can also be used as such a front-end program for external MP3 programs. This means that you can send .wav files directly from Audiograbber to your MP3 program without grabbing them from a CD first. They can be sent from the file menu, from the MP3 settings dialog box or by dragging and dropping them from Windows Explorer.

To play these MP3 files on your computer, I recommend [Winamp](#). Your soundcard can not play compressed files and you need a program to continually decompress them while playing. The audio data is usually sent to the soundcard in 16-bit stereo, 44.1 kHz. (this is called PCM, pulse code modulation and is the same format that is stored on an audio disc.)

MP3 settings

These settings are found under Settings, MP3 settings.

Audiograbber can make MP3's (or other compressed audio formats like WMA's) in two different ways. Either by using an internal encoder which gets integrated in Audiograbber and works as if it was a part of Audiograbber itself or by using an external encoder program. Audiograbber detects which internal encoders it finds when it starts and lists them in the internal encoder box. It looks for the following internal encoders:

LAME and BladeEnc's MP3 dll's. I recommend that you use LAME's dll at 128 Kbit/s joint stereo if you have no other ideas of your own. For archival purposes you can use an even higher bitrate like 192 Kbit/s. Settings for LAME and BladeEnc.

Fraunhofers acm codecs. This encoder comes in a few different flavors. The most common one is the advanced one which gets installed by some Microsoft programs so often the computer already has this encoder when Audiograbber starts. The advanced codec (l3codeca.acm) only supports MP3 encoding up to 56 Kbit/s, 22050 hz. 56 Kbit/s MP3's may sound fine on your cheap computer speakers but you will probably hear encoder artifacts if you listen to the MP3 at a more professional equipment. Another version of this encoder is the professional one which allows MP3 encoding up to 128 Kbit/s. This professional Fraunhofer encoder (l3codecp.acm) has also been hacked by the hacker group Radium and there are at least two different hacked professional acm codecs, one that goes to 256 Kbit/s and one that goes all the way to 320 Kbit/s.

Microsoft WMA.

QDesigns MP2 acm codec. This one is not so commonly used but some hardware equipment at radio stations can only use MP2, not MP3. It can be purchased from <http://www.qdesign.com>

OGG vorbis. This is a new very interesting audio format, it is completely free and open source. It also gives slightly better sounding files than MP3 from what I have heard.

Microsofts PCM converter. This one comes with Windows so it should be found on all computers. This differs from the other internal encoders in the way that it is not really an encoder. All it does is to change bit per sample, frequency, or stereo mode on wave files. Some programs like Adobe Premiere want the wav file in 48000 hz instead of the common 44100 so then the PCM converter comes in handy. Set to Microsoft PCM converter under Internal encoder and to direct rip and encode if you want your wav files in another format than the standard 44100 hz, 16 bit stereo.

You can also use an external encoder. An external encoder is usually a command line .exe program without a graphical user interface. Audiograbber will start the program and pass the proper arguments to it. Different programs use different arguments and in different order so it is important that the "Arguments" box has been setup correct. You can use one of the predefined arguments and modify it to fit your needs. Example (for X3enc.exe): If the predefined argument is "%s %d -b 128000" and you want 192 Kbit/s MP3's instead of 128 Kbit/s then change the 128000 part to 192000. Use %s for source file and %d for destination file in the argument string.

You must also type the correct file extension when you use an external encoder. That is because Audiograbber does not know what kind of files it produces.

Stdin and Stdout

Stdin and Stdout is short for standard input device and standard output device. Most encoder programs can only encode from a file and write the result to another file. A few encoders can however use stdin/stdout as data stream. That means that it can encode data sent to its stdin device and write the

result to the stdout device. Audiograbber can modify external programs stdin/stdout devices so it is used as a pipe between Audiograbber and the external program. That gives the advantage of using an external encoder as internal encoder in Audiograbber.

Launch encoder

This tells how the encoder will be started. Normal means that it will start on the top left position of the screen and minimized means that it will not disturb on the screen, it will only show up on the program bar. As Audiograbber currently is means that it will start in normal mode if Audiograbber is shown on the screen and minimized if Audiograbber is minimized.

Use ID3v1 tag

An **ID3 tag**[2400] is a 128 bytes tag appended to the end of an MP3 file. Its basic purpose is to hold the MP3's artist and track name. MP3 players will look for the artist and track name and display that while the MP3 plays. If no ID3 tag is found at the end of the MP3 then the player will have to display the file name instead.

Append ID3 info to wav file

Select this option to make Audiograbber store ID3 info and information about the final filename and destination of the track at the end of the wav file. When the wav file is later used for MP3 encoding the correct filename and ID3 tag is taken directly from the source file.

Some not so well behaved applications may treat the ID3 info at the end of the wav file as audio data so do not use this option in case you will use the wav file in some other program. This extra ID3 info does however not violate the wav standard and well written programs will just ignore it. Both ID3v1 and ID3v2 info is added to the file.

Rip all tracks before encoding

Normally Audiograbber rips and encodes tracks one by one. Select this option to rip all tracks first, before encoding. This is useful if you only have access to the disc for a short while. The downside is that more temporary harddisk space is required.

This page has [additional](#) info on MP3 settings and send to MP3 program.

LAME and BladeEnc settings

VBR / CBR

Only LAME dll's from ver 3.55 supports both CBR and VBR encoding. Earlier versions of LAME and BladeEnc supports only CBR.

CBR is the regular MP3 encoding mode, meaning Constant Bit rate. The MP3 file will get a constant bitrate all over with this settings. VBR means Variable Bit Rate is a newer encoding option. It means that the bitrate will vary all over the file depending how difficult and complex the audio is. VBR is good in the way that it gives better sounding files for the same file size. The downside is that not all hardware MP3 players can play VBR files.

Bitrate slider

Drag the slider to select the bitrate. 128 Kbit/s is the most common bitrate. You can see the actual bitrate or VBR mode on a label right above the slider. If you click that bitrate label it will expand and show the frequency for CBR modes and the average bitrate for VBR modes.

Quality

Here is a [page](#) that describes the different stereo modes.

Set to Voice if you record voice like a radio show or to low, normal or high quality. Low quality is the same as fast mode in previous versions of Audiograbber. Low quality is about twice as fast as Normal quality while high quality is around 30% slower. High quality is of course supposed to give the maximum sound quality for the bitrate.

Bitstream flags

These flags are encoded into the MP3 file and is not part of the ID3 tag. I don't know what they are good for and I haven't found any MP3 player that really cares about them.

Compare two files

This function is found under the file menu.

The purpose of this function is to test if your CD player reads digital audio perfectly.

When [digital audio](#) is read from the CD there is a problem with the beginning of the frames. That is because there is no proper beginning and end of the frames, rather a long consecutive sentence of bytes. This means that if a track is read twice from the CD, the files should most likely be almost exactly the same. Usually, it is only some 1/1000 of a second difference, so it does not matter much.

The disadvantage is that the DOS command `fc /b track1.wav track2.wav` can not be used to test that the files really are equal. The Compare Two Files function does a comparison in the same way but takes into consideration that the tracks may start a little bit differently. This function also skips the first 44 bytes because they are not part of the track, they contain file size and file format information.

The offset tells how many bytes (not samples, a sample is 4 bytes) the beginnings of the tracks differ from each other. The comparison is halted as soon as a difference is detected. If the track differs, it tells at which position. You can open the file in a program like Cool Edit and see how they differ. Remember that the hexadecimal value Audiograbber gives you should be subtracted by 44 and then divided by 4 to get the exact sample.

Another test function is the [checksum](#) function of Audiograbber.

pruefsummen@audiograbber.de

<http://www.audiograbber.de/pruefsummen.shtml>

Calculate checksum

This function is found under the file menu.

Some CD players that actually can read digital audio do not necessarily read 100% correctly. The resultant files are then not exact copies of the originals. This can also happen if the CD is scratched, has fingerprints etc. It is hard to tell if a track is read perfectly only by listening to it. Checksums are therefore a good way to test if the CD player read 100% correctly. When a track is copied from a CD disc, Audiograbber automatically calculates a checksum. By reading the same track on two different CD players the checksums can be compared. If they are the same then the tracks are also the same and the reading is perfect.

But hey, readings do not usually start on exactly the same byte, so how can a checksum be calculated then? Yeah, a presumption for this is that the track begins and ends with silence. The checksum will then be calculated on only that part of the track that is over a value of 127.

(A sample from the CD can be between -32768 to +32767 (16 bits). -32768 means that the speaker's membrane is pulled back as much as possible, +32767 that it is pushed forward to maximum position and 0 that it is in its middle position. The more the membrane is moving back and forth the louder the speaker is playing. Samples in the interval -127 to +127 are interpreted as silence in Audiograbber. When a checksum is calculated, Audiograbber finds the first sample outside this interval. Then it finds where the file goes outside the interval for the last time and after that it starts to calculate the checksum. It does not matter if there is a bit of silence in the middle of the track, that part will also be included in the checksum.)

NOTE! *If there is not enough silence (one frame = 2352 bytes) in either the beginning or the end of the track an X is added to the checksum. This indicates that other readings from the CD may give other checksums even if they are read in the same way. The checksum can not be used if it contains an X.*

By comparing checksums from tracks you have grabbed with your own CD with already known and confirmed correct checksums you can test if the CD reads the way it should. There are some correct checksums on <http://www.audiograbber.com-us.net/checksums.html> that you can compare with if you are lucky to have some of the CD's listed. Understand that you must have a copy of the exact same CD, it is not enough to have the same song.

You can report your own checksums on <http://www.audiograbber.com-us.net/checksums.html>. Those tracks should then have been read at least twice on a CD player that you are sure reads digital audio correctly. Report checksums from track 1,3 and 5. When you only want the checksum and not the entire track the function test comes in handy and you don't have to save the track on the hard disk.

Compare two files is another function that controls the CD's audio reading. If two checksums are not the same the compare two files function can see where the files are different. Checksums can also be calculated on tracks that have been read by other programs.

Store disc in database

This function is found under the CD menu (Store disc in database now).

Audiograbber has a plain text file for storage of disc info named discs.txt and it is found in the same directory as Audiograbber. This text file is fully compatible with Windows cdplayer.ini file. Disc info such as track names and artist names will be stored in this file when you use this function. You need to manually click this function if you have modified the track names for the disc yourself (by pressing F2 on a track) to have the new track names saved in discs.txt.

If you have checked the option "Use cdplayer.ini" under General Settings, Misc tab, then Audiograbber will store the disc info in both its own discs.txt and Windows cdplayer.ini. Discs.txt is proprietary for Audiograbber and no other program uses that file. Cdplayer.ini on the other hand is a more general way of storing disc info and is used by many programs so it is a good idea to store the disc info there too so other programs find it.

When Audiograbber refreshes the tracklist it will first look for the track names in its own discs.txt. If not found it will look for it in cdplayer.ini (if "Use cdplayer.ini" is checked) instead.

Cdplayer.ini has one major drawback and that is that it can only be up to 64 Kbyte, it will not store any more discs when it has reached that size (around 100 discs). Audiograbber's own discs.txt does not have that limitation and that is also the main reason for Audiograbber to have its own database file.

Lyrics are not saved to discs.txt in order to not make that file too large, instead they are saved to separate plain text files based on the freedb disc id with the extension .lyr.

Copy tracklist to clipboard

This function is found under the file menu.

This function copies the information about the tracks to Windows' clipboard. You can then paste it into another program like notepad. It can be good to use this function when testing [checksums](#).

Refresh Tracklist

This function is found under the CD menu.

The track list is automatically filled when Audiograbber starts. When you change CD you should use this function to fill the track list with information about the new CD. If the CD is recognized either in Audiograbbers own [database](#) or in Windows cdplayer.ini the track list will be filled with correct track names. If not, the track titles will be Track xx where xx indicates the track's ordinal number on the CD. Pressing F5 can also refresh the track list.

Audiograbber will automatically refresh the tracklist if your computer has been setup to notify programs of disc changes. That option is in Windows 95/98 found in Control panel -> System -> Device Manager -> CDROM -> (unit name) -> Properties -> Settings and it is called Auto insert notification.

Grab

This function is found under the CD menu.

This is the program's main function and it copies the chosen tracks from the CD to the hard disk. You choose tracks to copy by checking the appropriate checkboxes.

NOTE! *If you get pops and click in your files then do not run other programs when Audiograbber is copying a track. It sometimes happens that the reading will not be correct if Audiograbber is not given full access to the processor.*

When a track has been copied it will optionally be normalized and processed by an MP3 encoder.

Test

This function is found under the CD menu.

The purpose of this function is to calculate checksums without saving the files to the hard disk. Test is basically the same as Grab but nothing gets saved to the hard disk; neither normalization nor MP3 creation.

Shuffle

This function is found under the CD menu.

If you want to use Audiograbber as an ordinary CD player this function can be nice to use. By using shuffle, the tracks in the track list will be rearranged in a randomized order. You can also click the cd player button with a question mark to play tracks in random order. If cd player buttons are not shown in the status bar then click Settings, Show play buttons.

Select all

This function is reached by right clicking a track in the track list.

This function selects all audio tracks. You can also press Ctrl + A or the checked checkbox in the status bar to select all tracks. You can select a range of tracks by holding down shift when clicking on a checkbox.

Select none

This function is reached by right clicking a track in the track list.

This function unchecks all tracks in the track list so that they are no longer selected. You can also press Ctrl + N or click the unchecked checkbox in the status bar.

Track properties

This function is reached by double clicking or right clicking a track in the track list.

By double clicking a track you will get a dialog box in which you can adjust the track's properties like start and stop of the track etc.

Rename

You can rename a track by clicking an already selected track in the track list. Another way to rename a track is to right click it and choose Rename or by selecting the track and pressing F2.

Swap artist and trackname

This function is reached by right clicking a track in the track list. It is only available for compilation discs. Normally tracks on compilation discs has the artist name first followed by the track name. Some discs on freedb has however been stored with the track name first. If you get disc info from freedb with the artist and track names swapped then you can use this option to swap them back to normal.

Possible speed problems

A "possible speed problem" occurs when the computer/harddisk can't keep up with the flow of information coming from the CD-ROM. This forces the CD-ROM to take breaks from reading the CD. Because the CD disc is always spinning, difficulties can occur in finding the correct start position when reading recommences. Quality CD-ROM drives can handle "possible speed problems", but cheaper drives sometimes generate a pop or a click.

You can use the [checksum](#) function to find out if there are problems with this. Rip a track twice and see if you get the same checksum. If you do, then the files have been ripped exactly the same and "possible speed problems" can be ignored. If the checksums don't match then you must listen to the file and see if the speed problems have affected the song. Sometimes it is only a minor error and not audible. If you get annoying pops and clicks in your final file then change to a synchronized read method. Another way to decrease possible speed problems is to use the rip to RAM option under General settings, ASPI. Then Audiograbber stores as much as possible of the ripped data in internal RAM so harddisk speed and caching is not a problem.

Note that for Analog sampling the term is changed from "possible speed problems" to "speed problems". If you get a speed problem when sampling analog then there is definitely a piece of audio missing in the final file. Normally it should be no problem at all for the computer to handle incoming data when the analog mode is used. It is only 1x speed and even a 486 computer should be able to handle this. However, If you are sampling directly to an MP3 file then you may get speed problems whereby you should create a wav file and then encode it.

Submit to freedb

This function is found under the CD menu.

Freedb is a huge database available on the internet where individuals have submitted their disc data. Audiograbber can also submit (upload) disc info to this database. Before you submit a new disc you shall first check if the disc is already present in the database. Press the penguin button to see if it is already in the database. If it is not found in the database or if you want to correct some spelling errors then go to the CD menu and press "Submit to freedb".

The first page that welcomes you is where you enter artist and album name. If you are submitting a compilation disc with many artist then type Various in the artist field. The categories available here are the categories used in the database. You can not make up your own categories and they are not compatible with the MP3-ID3v1 tag categories. Use "Misc" in case you don't find an appropriate category.

Here is also a field with additional info for the disc. You can enter which artists that performs on the disc or any other interesting data. It is optional to use this field and it is not used by Audiograbber. Other programs may however use this information.

You can also notice that after a disc retrieve from freedb then additional fields with data that is not normally used by Audiograbber may have been filled in.

Go through each track and correct spelling mistakes and optionally fill in additional info about the tracks. The additional field for each track is normally left empty but if there is something special to note about the track then enter that info here. This info is not used by Audiograbber but may be used by other programs that downloads the disc info.

If you are submitting a totally new disc then it is a good idea to press the erase pen in the upper right corner. The track name list will then become empty and it is faster to fill in the tracknames. When you are done then you can press the "Export to tracklist" button and send the info back to Audiograbber.

When you have filled in all the track names and checked the spellings carefully then press the "Next" button. On the next page you will only have to check that your e-mail address is correct. If your disc submission should fail for some reason then you will get an e-mail telling you why. If the disc submission is accepted then no e-mail will be sent. Press the Submit button and the disc info will be uploaded to the database. It should normally only take around 10 seconds to upload the disc info. If you have decided to also use a local freedb disc database you can also press a button to store the disc info in your local database.

It is recommended that you use ASPI (under General settings) to fill the tracklist of the disc if it is a multimedia disc with a data track at the end. MSCDEX and Analog will not see the data track at the end and it is good to have that included in the database. For normal discs it doesn't matter if you use ASPI, MSCDEX or Analog.

There is a small risk that two different discs has so similar TOC (Table Of Contents, ie the discs has equally many tracks and the tracks are also equally long) that they generate the same freedb disc ID. It is still possible to have both discs in the database but the only thing that separates them from each other will be the category. This means that if you want to submit a disc that is already in the database but is a completely different disc then you must specify another category for your disc.

Refresh

Press this button when a new CD is inserted or if you want to refresh the track list. The track names will be retrieved from the disc database or from cdplayer.ini if you have checked "use CD player.ini" under "General settings". If the disc is not recognized then tracks will get default names with ordinal numbers appended.

Grab

Press this button to copy the selected tracks. You select tracks by checking the box in front of the track names.

Settings

Press this button to adjust the program's general settings.

Normalize

Different CD's are recorded at different volumes. Press this button in order to select that copied tracks automatically are adjusted to the same recording level.

MP3

MP3 is a standard for digital audio compression. Audiograbber can send the tracks to an external program or internal dll for automatic creation of MP3's. Here is where you select your MP3 or other compressed audio format options.

Freedb retrieve

Press this button to automatically retrieve the track names from an internet database.

MP3 Player

Press this button to launch your preferred MP3 player with a playlist (an .m3u list) based on the checked tracks. If you right click this button then the MP3 Player will be launched without any playlist.

Exit

Press here to exit the program. Your settings will be saved in audiograbber.ini.

Artist

The name of the artist who made the disc. If this is a compilation disc then check the "Compilation disc" box and enter "Various artists" here.

Compilation disc

Check this box if the disc has more than one artist. With this box checked the tracklist will have an extra column for artist names.

Album

The album name.

Track name

Double click to go to the properties menu for each track. Right click to get a pop-up menu. You can also change the order of the tracks by dragging them to new positions. Tracks with a yellow checkbox has at least one individual ID3v2 property.

Artist name

Double click to go to the properties menu for each track. Right click to get a pop-up menu. You can also change the order of the tracks by dragging them to new positions.

Play time

Describes the length of the track in minutes and seconds.

File size

The track will take up this much space on the hard disk. The file format is PCM wav, 16-bit stereo, 44.1 kHz, same as on the CD. Audiograbber can not handle any other file formats. The file size is in direct proportion to the track length.

Information

Here is information about how the copying has gone and which checksum has been calculated. Sync OK is no guarantee that the copying is completely perfect.

CD info

Pretty obvious eh...

Check all

Click here to quickly select all.

Check none

Unselects all tracks.

Show track numbers

Toggles displaying of track numbers in the tracklist.

Total time left

The copying of the tracks is estimated to take this long. Any normalizing and mp3 processing are included. The more often the program is used, the better it gets at estimating the total time left.

Disk space needed

This tells how much hard disk space is needed and how much disk space is available on the hard disk being used to store the tracks. [Click here](#) to quickly switch to the play buttons.

Play time

Information telling which track is currently playing and for how long it has been playing - how much time remains. Click to switch between showing time elapsed and time remaining.

Play

Plays selected track. Keyboard shortcut Ctrl + P. Use the other play button with a check mark to play more tracks than one.

Pause

Use this button to pause and resume the audio. Keyboard shortcut Ctrl + P.

Stop

Stops playing the track(s). Keyboard shortcut Ctrl + S.

Play list

Plays the tracks selected with a check mark in the track list. Keyboard shortcut Ctrl + L. *Note: The playing starts from the track that is selected.*

Fast back

Jumps quickly backward in the track. It jumps more and more quickly the longer the button is held down.

Fast forward

Jumps quickly forward in the track. When the end of the track is reached it will continue with the next one if you are playing tracks from a list.

Track back

Skips to beginning of the track. If it already is in the beginning of the track it then jumps to the beginning of the previous track. Keyboard shortcut Ctrl + B.

Track forward

Skips to the end of the track. If tracks from a list are already playing, the next selected track will start playing. Keyboard shortcut Ctrl + F.

Empty space

Click here or on one of the disabled buttons to quickly hide all buttons.

Random play

Plays the tracks selected with a check mark in random order. Playing does not stop until the stop button is pressed. Keyboard shortcut Ctrl + R.

Eject

Ejects or loads the disc. Keyboard shortcut Ctrl + E.

Speaker

Switch between display of the volume meter and status bar.

Play progress

Indicator that displays how much of a track has been played. You can manually drag this meter to a new position.

Volume meter

Adjusts the volume of your CD-ROM drive.

Show play buttons

This function is found under the Settings menu.

This option toggles between showing CD play buttons or disk info in the statusbar. It can also be changed by clicking on an empty area of the right part of the statusbar.

The following buttons are available:

Play single track. (Plays the highlighted track and only that one).

Pause/Resume.

Stop.

Play selected tracks. (Plays the tracks with ticked checkboxes, starts with the highlighted one).

Fast rewind.

Fast forward.

Skip backwards. (Restarts the track from beginning or when it is already on the beginning of the track, plays the previous track).

Skip forward. (Plays next track).

Play selected tracks in random order.

Eject/insert disc.

These buttons has also [keyboard shortcuts](#).

Track bar

Shows the position of the track. One can manually drag it to set a new position.

Play button

Starts playing from the position of the above meter.

Stop button

Stops playing the track.

Set start

Sets the track start position to the value of the above meter.

Set end

Sets the track end position to the value of the above meter.

Position

Shows from where on the CD the track is playing right now. Position in the track is shown in minutes and seconds while position on the CD is measured in frames. One frame is $\frac{1}{75}$ of a second.

Track name

Simply the name of the track. Can be changed from here or by clicking an already selected track in the track list.

Artist name

The name of the artist who has made the track. This edit field is only shown for compilation discs. On ordinary discs the artist name shall be entered in the artist field on the main form instead.

Track size

This many bytes will be used to store the track on the hard disk. (Frames * 2352 bytes).

Minutes, seconds

The tracks start and stop in absolute position on the CD, reported in minutes and seconds.

Sectors

The tracks start and stop in absolute position on the CD, reported in sectors. Frames and sectors are the same and it is 75 sectors on every second of music.

Apply

Performs all the changes you might have made without leaving this dialog box.

Cancel

Leaves this dialog box without performing the changes you might have made.

OK

Leaves this dialog box and performs the changes you might have made.

Use normalizing

Select this box if you wish to automatically adjust the recording level, leave this checkbox unchecked if you wish to copy the track exactly as it is.

Normalize to

Select the recording level between 0 -100%. 0% is equal to silence and 100% is the highest volume. 98% is recommended.

Conditions

Use this option to adjust just those tracks that were imperfectly recorded at the CD factory (too low output level). It is recommended that this function be used.

Lower than

Those tracks that have a peak level lower than this value will be normalized. Approximately 90 % is recommended.

Higher than

Tracks with a higher peak level than this are normalized. 99% or 100% is recommended.

Browse

Click here to choose one or more wav files from the hard disk to normalize. They will be normalized to the values selected in this dialog box.

Cancel

Leaves this dialog box without performing the changes you might have made.

OK

Leaves this dialog box and performs the changes you might have made.

Direct rip and encode

Select this option to rip directly to an MP3 file without first creating a wav file. This does not work with MSCDEX as rip method and normalization can not be used in combination with this option. This option can not be used with the "rip all tracks before encoding" option and it needs an internal encoder (or an external used as internal) to be enabled.

Encoder priority

You can lower the CPU priority of your encoder here. This only works when encoding is the only task that is going on, ripping is so sensitive that normal priority is always used. Useful if you are doing other things on the computer during encoding.

First wav, then MP3 file

Select this option to have Audiograbber first rip to a wav file and then encode it to MP3. The wav file will be kept.

First wav, then MP3 file

Select this option to have Audiograbber first rip to a wav file and then encode it to MP3. The wav file will be deleted when the MP3 file is finished.

Rip to wav file

This option lets Audiograbber rip to a standard wav file, 44100 hz, 16 bit stereo, a bitwise copy of the audio data on the cd. This is the best option if you plan to burn a new cd from your ripped files.

MP3 program name

The name of the executable MP3 program. Use the browse button to locate your version of the program. Common ones includes L3enc.exe, BladeEnc.exe or X3enc.exe. Go to Audiograbbers web page to find links to external encoders. It does not necessarily have to be an MP3 encoder, it can be some other kind of encoder program too.

Browse

If you have an external MP3 program you can use this button to locate it.

Predefined arguments

This is a list of common encoder arguments. Different encoder programs want the arguments in different ways and this list holds some sample values for common encoders. Select a value in this list and the arguments line will change to the corresponding argument. You can optionally modify the arguments line if you want other special settings for the encoder.

Stdin/Stdout capable

Stdin and Stdout is short for standard input device and standard output device. Most encoder programs can only encode from a file and write the result to another file. A few encoders can however use stdin/stdout as data stream. That means that it can encode data sent to its stdin device and write the result to the stdout device. Audiograbber can modify external programs stdin/stdout devices so it is used as a pipe between Audiograbber and the external program. That gives the advantage of using an external encoder as internal encoder in Audiograbber.

Use encoder as internal encoder

This option becomes available if the encoder can handle stdin/stdout as data stream. With this option checked it is possible to use "Direct rip and encode" even with external programs. Instead of letting the external program encode from a wave file to another file Audiograbber will send PCM audio data to the encoders stdin and receive encoded data back via the encoders stdout.

Launch encoder

This tells how the encoder will be started. Normal means that it will start on the top left position of the screen and minimized means that it will not disturb on the screen, it will only show up on the program bar. As Audiograbber currently is means that it will start in normal mode if Audiograbber is shown on the screen and minimized if Audiograbber is minimized.

Highest quality

Tells Fraunhofers MP3 codec that it should take a longer time to analyze the wav file and create an MP3 file with extra high quality.

Rip all tracks before encoding

Normally Audiograbber rips and encodes tracks one by one. Select this option to rip all tracks first, before encoding. This is useful if you only have access to the disc for a short while. The downside is that more temporary harddisk space is required.

Constant bitrate

This is the standard mode for MP3 files. MP3 files consist of frames and each frame will have the same amount of bits.

Variable bitrate

This is a rather new way of MP3 encoding. Each frame in the MP3 file will have a bitrate that depends on the complexity of the sound. Parts of the song that is easy to encode and needs fewer bits to sound good will get fewer bits and the encoder will allocate more bits to the more complex parts of the song. This gives basically a better sounding file for the same amount of bits. Drawback is that some MP3 players has problems to play VBR files.

Quality

Mono means that the file has only one sound channel. Joint Stereo means that the encoder takes advantage of similarities in the two stereo channels and makes a stereo file. Stereo includes two independent channels. The total bitrate remains constant, but the split between the channels can vary. Dual Stereo could really be named dual mono because it is two completely separated channels, each with half of the total bitrate. Joint stereo is recommended for bitrates below 224 Kbit/s. Not all options may be enabled here depending on which encoder you have. The latest LAME dll's supports all settings though.

Quality type

Use Voice for voice recordings and low, normal or high for music. High will give a slightly better sounding file but it will also take longer time to encode. Low is the fastest encoding mode and it doesn't reduce the sound quality very much.

Bitstream flags

These flags are encoded into the MP3 file but it doesn't matter much how they are set. I haven't seen any player that really cares about these flags. CRC will use two bytes in the MP3 file for a cyclic redundancy checksum which might be useful for streamed MP3 files.

WMA Bitrate

Tells how many bits that shall be used per second of audio. The more bits the better sound quality and larger file. 128 Kbit/s is just a little less than one MB per minute of audio. 64 Kbit/s is a good choice for portable WMA players.

Mono

Mono means that the file has only one sound channel. If the original file is stereo then the sound channels will be joined together. Mono is useful to save bits and file size, especially when the original sound is a voice recording.

Stereo

Stereo is two different sound channels and the most common sound format for music on computers.

Packaged WMA

This setting tells the WMA encoder to make files that can only be played back on the computer they were created on. Only God and Microsoft knows what that should be good for. Microsoft asked me to implement this option anyhow.

Bitrate

Chooses how many bits per second the MP3 file will consist of. The higher the bitrate the better the quality and the larger the file. Most songs sound good with a bitrate of 128000 bps but a very few need 192000 bps or even more to sound equal to the original. It's hard to say in advance which tracks these are. VBR bitrate files will have different bitrates all over the file depending on complexity of the sound.

Bitrate

Chooses how many bits per second the OGG file will consist of. The higher the bitrate the better the quality and the larger the file. Most songs sound good with a bitrate of 128000 bps but a very few need 192000 bps or even higher to sound equal to the original. Click and move the slider to see what average bitrate each setting corresponds to.

Arguments

Type the arguments for the external encoder program here. You can use one of the predefined arguments in the list above or make one on your own. Read the documentation for the encoder program to understand what arguments it can use. Press the info button to get additional information for which parameters Audiograbber can send.

Use ID3v1 tag

ID3v1 tag contains extra information about the song that is added to the end of the MP3 file. An ID3v1 tag will be created if this option is selected. It is Audiograbber that creates this ID3v1 tag when the MP3 program has finished, not the MP3 program itself.

Append ID3 info to wav file

Select this option to make Audiograbber store ID3 info and information about the final filename and destination of the track at the end of the wav file. When the wav file is later used for MP3 encoding the correct filename and ID3 tag is taken directly from the source file. Both ID3v1 and v2 data will be stored in the wav file.

Some not so well behaved applications may treat the ID3 info at the end of the wav file as audio data so do not use this option in case you will use the wav file in some other program. This extra ID3 info does however not violate the wav standard and well written programs will just ignore it.

Browse

Click here to choose one or more wav files from the hard disk to send to your MP3 program with the options specified in this dialog box.

Cancel

Leaves this dialog box without performing the changes you might have made.

OK

Leaves this dialog box and performs the changes you might have made.

Edit ID3v1 tag

Press this button to edit the info that goes into the ID3v1 Tag.

External MP3 program name

Select this option if you use an external program to create your MP3's (or any other compressed audio format).

Internal Encoder

This option becomes available if you have one or more of the following encoders installed: Fraunhofers acm codecs, LAME or BladeEnc's MP3 dll's, QDesigns MP2 acm codec, Microsoft WMA encoder or PCM converter or an OGG Vorbis dll. It will look as if Audiograbber is doing the MP3 creation however this is not the case. Audiograbber is just a frontend to the encoders. Go to Audiograbbbers web page to find links to encoders.

CODEC Name

7 different encoders can show up here: Any of Fraunhofers acm codecs (If you only have 56 Kbit options with Fraunhofer then search the helpfile for l3codecp.acm for more info). Qdesigns MP2 acm codec, BladeEnc's or LAME's freeware MP3 DLLs, Vorbis OGG dll, Microsoft MS Audio codec and Microsoft's PCM Converter are also internally supported if they are found on the computer.

Mode

Select what bitrate should be used for the MP3 or MP2 file. The higher bitrate the larger file and better sound quality. 128 Kbit/s is standard for MP3 files. 128 Kbit/s is a compression factor of 1/11 and each minute of audio takes just a little less than 1 MB.

Own extension

Type the file extension here. The most common one is .mp3 but nowadays there are many different file types besides MP3. The dot between the file name and extension shall not be entered here.

Info button

Press this button to get information about the parameters you can specify on the own arguments line.

Directory

Select a directory to store your wav or mp3 files in. Audiograbber also uses this directory for temporary files.

Browse

Click here to select an appropriate destination directory for your files.

Time estimation

This is used to calculate how much time the program will need to process its tasks. How fast the external MP3 program or internal codec runs can not be changed from here because there are too many bitrates that are involved. Those settings are stored in Audiograbber.ini and can be changed by editing that file manually.

Normalize speed

Detects how fast the adjustment of the recording level is done compared to how fast a track is played. The speed of the hard disk is also a vital part of this.

CD audio read speed

Detects how fast the CD-ROM drive can read digital audio compared to normal playing of the audio. Music is normally read much slower than ordinary data from the CD.

Silence Tab

Silence can be deleted automatically from the start and the end of the tracks. Normally, there is quite a bit of silence between the tracks on the CD, and Audiograbber checks the first and last four seconds for silence.

Leading silence

This option works with wav files only. It works when you rip to a wav file either in RAM or to the harddisk but it does not work if you rip directly to MP3.

Trailing silence

Select this option to delete trailing silence. This option does not work with direct rip and encode, you need to rip to a wav file first.

Keep

It is best to let the tracks start with at least a little silence; this value can be between 0 and 4 seconds.

Keep

This value can be between 0 and 4 seconds. A CD does not have a value for the end of the track; it only has values for the beginnings of the tracks. This means that there is usually quite a bit of silence at the end of the track. The recommended value is about 0.5 seconds.

Audio blocks

Determines how many audio frames should be grabbed each time. One second is 75 frames.

Synch size

Detects how many extra frames should be grabbed each time for synchronization. This is explained in more detail in the help file.

Continue

Tells Audiograbber what to do if synchronization fails. The track will not be a perfect copy if sync is lost. Check this box if you want Audiograbber to continue with the copying even if sync is lost.

CD player.ini

Check this box if you want to store the track names and the order of the tracks in cdplayer.ini. That file is also used by Windows 95/98's default CD player. If this option is used and the CD is recognized the track list will be filled with the track names when you press the refresh button.

Audiograbber will also store the disc info in its internal database file, discs.txt, regardless if cdplayer.ini is used or not..

Autosave

Check this box if you want to have the data retrieved from freedb stored in your discs.txt and cdplayer.ini file automatically.

AutoQuery freedb

Check this box if you want Audiograbber to automatically look up the disc in freedb (internet) if it is not recognized.

CD ROM unit

Tells which CD-ROM Unit that should be used for ripping through MSCDEX calls.

Cancel

Leaves this dialog box without performing the changes you might have made.

OK

Leaves this dialog box and performs the changes you might have made.

Lock values

The time estimation values are recalculated every time a track has been processed by Audiograbber. If you don't want any recalculation of the time estimation variables then check this box.

First file

Enter the name of the first file. The complete path must be used.

Second file

Enter the name of the second file. The complete path must be used.

Browse

Browse to find a file to compare here. You can select two files at once with any of these browse buttons.

Offset

A 0 offset means that the files start at exactly the same positions. A positive offset means that file 1 starts after file 2 and a negative offset means that file 1 starts before file 2. The offset will most likely be a multiple of four because every sample takes 4 bytes. (16 bit * stereo).

Progress bar

Detects what percent of the files have been compared and found equal.

Value list

If the files differ from each other, the first values that do not match are found in here. All values are hexadecimal. It is not possible to continue to search for more differences after a mismatch has been detected.

The first 44 bytes of the files are the wav header and will not be compared.

Start / Stop

Starts and stops file comparing. This button is only enabled when two valid file names have been given.

Close

Closes this dialog box.

Windows version

Detects which version of Windows you are running.

Audiograbber version

This is the version and build number of Audiograbber. For Audiograbber a change of .10 means a large update and a change of .01 means a small update. Build number increases indicates bug fixes and minor updates. Other programs have different ways of using version and build numbers so it can't be compared with other programs. Audiograbber 1.50 is for example equivalent to Audiocatalyst 2.10.

Build number

Detects which minor version of windows you are running.

950 is the first windows 95 version.

1111 is the OSR2 release.

There is also a 953 version that is reported the same way as the 950 version.

Windows 98 is build number 1998.

More versions to come...

Scsi1hlp.vxd version

This file is important when reading digital audio through MSCDEX calls. Usually the only version that works is Windows 95's original version 950. Read more about Scsi1hlp.vxd in the help file.

Frames allocated

This value is only useful when MSCDEX is used as reading method.

Audiograbber will not always get as much memory for the MSCDEX ripping as it wants. This value is the number of allocated frames Audiograbber received this time.

MSCDEX version

Audiograbber can call either MSCDEX or ASPI to read digital audio. MSCDEX is only available under Windows 95/98 and with IDE drives. Version 2.25 or lower means that a real mode driver is used for the CD-ROM drive. Version 2.95 or higher means that Windows 95's protected mode driver is used.

CD number

This is the serial number of the actual CD. This serial number is used by Windows for storage in cdplayer.ini. The number is shown in hexadecimal form. Audiograbber can not detect this number when a real mode driver is used and the serial number will then be "UNKNOWN".

Freedb Disc ID

This is the freedb serial number of the actual CD. This number is based on track information of the disc and is used when the disc is looked up in the freedb database on the Internet.

Close

Close the dialog box.

Filename

Enter the name of the file. Use the complete path.

Browse

Browse for a file to calculate checksum by using this button.

Progress bar

Detects what percent of the file has been scanned and calculated.

Checksum

The checksum that was last calculated in this dialog box. Checksums are also calculated directly when tracks are copied from the CD. This value and the automatically calculated one are naturally the same.

CD number

This is the serial number of the actual CD. The number is given in hexadecimal form.

Start

Entering a valid file name will enable this button. It is then used to start the calculation of a checksum. When the calculation is in progress the button can be used to abort.

Close

Closes this dialog box and aborts eventual checksum calculation.

Cancel

Makes Audiograbber stop waiting for the external MP3 program. The external MP3 program must be aborted manually (usually by pressing Ctrl+C).

ID3v1 tag

This function is found under the settings menu.

ID3v1 tag contains extra information about the song that is added to the end of the MP3 file (128 bytes at the end of the file). You can choose to let Audiograbber append an ID3v1 tag to the MP3 file. It is also possible to let Audiograbber use the newer and more advanced ID3 standard, [ID3v2](#).

The format of the ID3v1 tag is like this:

Track name, 30 characters.

Artist name, 30 characters.

Album name, 30 characters.

Year, 4 characters.

Comment, 30 characters.

Genre, 1 byte.

ID3v1.1 means that the two last characters of the comment field is used to store the track number so with that option turned on the comment field will only have 28 characters. It is possible to enter more than 30 characters in the fields because they can also be used as source for the ID3v2 tag (which has no field length limit) but only the first 30 characters are used in the ID3v1 tag anyhow.

See further down for a list of all defined genres. These are predefined and that is the reason that you can't make up your own.

It is also possible to let Audiograbber use a part of the comment field for storing the checksum of the ripped track. This can be useful if you listen to an MP3 file and gets suspicious if it really was ripped correct. Rip it again and see if you get the same checksum.

Microsofts Windows Media Audio (WMA) can also contain ID3 tag and WMA has the advantage to have some extra fields and are not limited to max 30 characters per field.

(Extra fields used in Audiograbber are: Description, Copyright message, Album cover URL and Promotion URL).

Here's the list of the predefined ID3 genres:

```
char tags[][22] = {
{"Blues"}, // 0
{"Classic Rock"}, // 1
{"Country"}, // 2 etc
{"Dance"},
{"Disco"},
{"Funk"},
{"Grunge"},
{"Hip-Hop"},
{"Jazz"},
{"Metal"},
{"New Age"},
{"Oldies"},
{"Other"},
{"Pop"},
{"R&B"},
{"Rap"},
{"Reggae"},
{"Rock"},
{"Techno"},
```

{ "Industrial"},
{ "Alternative"},
{ "Ska"},
{ "Death Metal"},
{ "Pranks"},
{ "Soundtrack"},
{ "Euro-Techno"},
{ "Ambient"},
{ "Trip-Hop"},
{ "Vocal"},
{ "Jazz+Funk"},
{ "Fusion"},
{ "Trance"},
{ "Classical"},
{ "Instrumental"},
{ "Acid"},
{ "House"},
{ "Game"},
{ "Sound Clip"},
{ "Gospel"},
{ "Noise"},
{ "Alternative Rock"},
{ "Bass"},
{ "Soul"},
{ "Punk"},
{ "Space"},
{ "Meditative"},
{ "Instrumental Pop"},
{ "Instrumental Rock"},
{ "Ethnic"},
{ "Gothic"},
{ "Darkwave"},
{ "Techno-Industrial"},
{ "Electronic"},
{ "Pop-Folk"},
{ "Eurodance"},
{ "Dream"},
{ "Southern Rock"},
{ "Comedy"},
{ "Cult"},
{ "Gangsta"},
{ "Top 40"},
{ "Christian Rap"},
{ "Pop/Funk"},
{ "Jungle"},
{ "Native US"},
{ "Cabaret"},
{ "New Wave"},
{ "Psychadelic"},
{ "Rave"},
{ "Showtunes"},
{ "Trailer"},
{ "Lo-Fi"},
{ "Tribal"},
{ "Acid Punk"},
{ "Acid Jazz"},

{ "Polka" },
{ "Retro" },
{ "Musical" },
{ "Rock & Roll" },
{ "Hard Rock" },
{ "Folk" },
{ "Folk-Rock" },
{ "National Folk" },
{ "Swing" },
{ "Fast Fusion" },
{ "Bebob" },
{ "Latin" },
{ "Revival" },
{ "Celtic" },
{ "Bluegrass" },
{ "Avantgarde" },
{ "Gothic Rock" },
{ "Progressive Rock" },
{ "Psychedelic Rock" },
{ "Symphonic Rock" },
{ "Slow Rock" },
{ "Big Band" },
{ "Chorus" },
{ "Easy Listening" },
{ "Acoustic" },
{ "Humour" },
{ "Speech" },
{ "Chanson" },
{ "Opera" },
{ "Chamber Music" },
{ "Sonata" },
{ "Symphony" },
{ "Booty Bass" },
{ "Primus" },
{ "Porn Groove" },
{ "Satire" },
{ "Slow Jam" },
{ "Club" },
{ "Tango" },
{ "Samba" },
{ "Folklore" },
{ "Ballad" },
{ "Power Ballad" },
{ "Rhythmic Soul" },
{ "Freestyle" },
{ "Duet" },
{ "Punk Rock" },
{ "Drum Solo" },
{ "A Cappella" },
{ "Euro-House" },
{ "Dance Hall" },
{ "Goa" },
{ "Drum & Bass" },
{ "Club-House" },
{ "Hardcore" },
{ "Terror" },

```
{"Indie"},  
{"BritPop"},  
{"Negerpunk"},  
{"Polsk Punk"},  
{"Beat"},  
{"Christian Gangsta Rap"},  
{"Heavy Metal"},  
{"Black Metal"},  
{"Crossover"},  
{"Contemporary Christian"},  
{"Christian Rock"},  
{"Merengue"},  
{"Salsa"},  
{"Thrash Metal"},  
{"Anime"},  
{"JPop"},  
{"Synthpop"}, //147  
{""}; //255 Unknown
```

ID3 Tag title

The ID3 tag title is normally taken from the track name or from the name of the wave files. The title will be taken from this field only when a single file is sent to MP3 creation.

Artist

The name of the artist or the band that has made the song. This can also be set directly from Audiograbbers main window.

Album

The name of the CD. This will also be stored in cdplayer.ini if that file is used for storing info.

Year

The year the song was released.

Genre

The genre of the song. It is only possible to use one of the 147 predefined genres that are specified in the ID3v1 tag specs.

Comment

You can type any comment you like here, for example your nickname.

Append checksum

If this box is checked then Audiograbber will append the checksum of the track if it is ripped through Audiograbber. The comment field, that is 30 characters long, must have 14 characters left for the checksum. This means that you can only use 16 characters for your own comment.

Include tracknumber (ID3v1.1)

Check this box if you want Audiograbber to use the two last characters of the comment field to hold the track number. The size of the comment field will be reduced from 30 to 28 characters. Not all MP3 players supports ID3v1.1 but it is safe to use this option anyhow, it will not cause any problems for none ID3v1.1 enabled players.

Cancel

Leaves this dialog box without performing the changes you might have made.

OK

Leaves this dialog box and performs the changes you might have made.

Description

Description of the song or the album. For example: "An upbeat song that made quite of a rumble when it was released 1987. Listen carefully to the lyrics!"

Copyright message

A copyright message of this song.

Album cover URL

An internet link that points to a picture associated with the disc. Something like
<http://www.audiograbber.com-us.net/jackie.jpg>

Promotion URL

A link to the artists or distributors homepage. Something like <http://www.audiograbber.com-us.net>

Windows Media Audio

Windows Media Audio is Microsoft's answer to MP3. It is basically another file format for compressed audio and the file extensions for such files are .wma. Microsoft itself claims that the sound quality should be the same for a file that is only half the size of an MP3! This may be true or close to true for lower bitrates like 64 Kbit but I am not sure how good it sounds for higher bitrates. I rather stay out of the debates of which formats and which encoders that gives best quality. Other people are more suitable to judge that. Encoding to wma is however very fast, almost as fast as Xing's MP3 encoder and not so much slower than QDesigns MP2 encoder.

It is definitely a file format to check out and with the power of Microsoft promoting it wma has a great chance to be a widely used file format. The encoder and decoder doesn't cost anything.

"Packaged WMA" means that the file can only be played back on the computer it was created on so beware of that option.

You can hopefully find more info at Microsofts own page:

<http://www.microsoft.com/windows/windowsmedia/>

Audiograbber can now internally compress to Windows Media Audio too (It can also use Fraunhofers MP3 acm codec, LAME and BladeEnc's freeware MP3 dll's, QDesigns MP2 acm codec and OGG Vorbis dll) if the computer has the proper codecs installed. You might be able to find the file wmaudioedist.exe on some other major MP3 players homepages. Sonique and Winamp, for example. Both Sonique (ver 1.30) and Winamp (ver 2.61) currently comes as a "full" version which includes Microsoft WMA too. By downloading and installing one of them you will get WMA working in Audiograbber too.

<http://www.sonique.com/>

<http://www.winamp.com/>

Properties

Properties is a dialog box from which you can adjust a track's properties. You go to the Properties dialog box by double clicking a track in the track list. From here you can adjust exactly which sectors the track shall include. The track can also be played in here and you can set the beginning and end position "on the fly". By dragging the meter on top of the dialog box you can easily rewind / forward the track and that is handy if you want to copy just a refrain, for example. You should then click "Set start" and "Set end" at the right time.

By pressing the Apply button your changes will be performed. This button differs from the OK button in that sense that it will apply the changes but not leave the dialog box.

General Settings

The following functions are adjustable from here:

- The program's working directory, where files should be stored. It is also possible to let Audiograbber create sub directories based on artist and album name.
- **CD-ROM access method.** Access method of CD data given here. Audiograbber is able to use either ASPI, MSCDEX or Analog calls. MSCDEX only works whilst running Windows 95/98 and an IDE CD-ROM's. ASPI works with both Windows 95/98/NT and IDE/SCSI readers. It is hard to predict which is best for a particular computer and CD-ROM, the best method (as always) is trial and error! There are more detailed descriptions of [MSCDEX](#) and [ASPI](#) in this help file. [Analog](#) is not as good as the digital rip methods ASPI and MSCDEX but if your drive does not support digital reading of audio then you will have to use analog. The soundcard you specify here is also the one used in the [line in sampling function](#) function.
- **Naming.** This is how your wav and mp3 files will be named. Simply check the boxes from where you want the filename taken. You can change the order by using the arrows at the left side. All fields are separated by a blank, a dash and another blank " - ".
It is also possible to store the files in sub directories based on the artist and album name.
Another option is to only store encoded files in sub directories. This option is primarily intended to be used in conjunction with the option "append ID3 info to wav files" found under MP3 settings. Selecting it lets the optional ID3 info and track names in the wav files get the correct directory structure but leaves the wav files in the base directory. This way you drag and drop a lot of files on Audiograbbers main window and they will end up in correct sub directories after encoding.
You can also check the advanced naming box to get more control of your file names. Press the info button to read a text that describes which parameters you can use for advanced naming.
- **Delete silence automatically.** There is usually quite a bit of silence between tracks on a CD. From here you can select how much silence in the beginning and end of a track should be kept. Values from -127 to 127 are treated as silence. Read more under [checksums](#) about what these 127's are. This option does not work with direct rip and encode, you need to rip to a wav file first.
- **Rip offset.** If your CD reader has problems finding the exact start and stop points of tracks, different offsets can be given here. You can also tell Audiograbber to simply discard a few of the first frames if your CD reader produces a click at the beginning of the track. Normally these values should be set to zero if you don't experience any problems though. Some crappy drives has problems to start reading from a non moving disc. By checking the "spin up disc" box Audiograbber will try to spin up the disc for 3 seconds before it starts reading.
Some drives also tends to try to read outside the last frame of the last track due to bad positioning of the read head and gives an "ASPI error" just before they shall finish the last track. Check the "Shorten..." box to simply skip some frames and be on the safe side.
- **Partial rips.** Use this if you want to make short samples of the tracks. You can define how long each sample shall be and from what position in the track it shall start. It is also possible to have the tracks faded in and/or out to avoid abrupt starts/endings. The fade time is normally 2 seconds and is a logarithmic fade type. The fade time and type can be changed by modifying these lines in audiograbber.ini:
FadeTime=1 (It is possible to use 1, 2 or 3 seconds).
LogarithmicFade=False
- **Time Estimation.** [Normalize](#) speed indicates how quickly your computer can normalize a track. Both processor speed and the speed of the hard disk are important for this value. Here is also the value for how quickly the CD reader reads digital audio. 100% means that it reads just as quickly as the track is normally played. These values are recalculated when the program is used but they can also be

adjusted manually. The values will not be recalculated if the check box "Lock values" is checked. The speed of [external MP3 encoders](#) is not adjustable from here since there are too many factors to be taken into consideration (like hq-bit and bitrate). The speed of encoders are also dependent on the actual type of music that is being compressed. Sometimes it's quicker; sometimes it's a bit slower. The external encoder speed values can be adjusted manually by editing the file Audiograbber.ini. The values there correspond to the different bitrates without hqbit and they too are recalculated. They should be compared with normal audio speed, for example 5.0000 means that the compression is 5 times slower than playing the track.

- **Miscellaneous**, with the following settings:

Select all tracks by default

If this option is selected then Audiograbber will put a checkmark in the boxes in front of the tracks as default. If not selected then all tracks will become unchecked by default.

Continue even if synchronization fails

The copy will not be perfect if the synchronization fails. By clicking "Continue even if synch is lost" will save the track regardless of how the synchronization is achieved, otherwise the track will be skipped if synch problems are detected. (This is called "jitter correction" in some programs).

Don't calculate checksum during the rip process

By choosing this alternative, the program won't automatically calculate a checksum when reading a track. This frees up more processor time which can be used for the actual reading and can be useful when using Burst copy and ASPI in conjunction with a fast CD-ROM.

Use cdplayer.ini

Audiograbber stores disc info in an internal database file named discs.txt. It is also possible to store and retrieve disc info from Windows cdplayer.ini file. With this option checked Audiograbber will first look for the disc in its own database file and if not found it will look for it in cdplayer.ini. It can be a good idea to use cdplayer.ini so other programs also will be able to pick up track names saved from Audiograbber.

(Note: An ini file in Windows can normally not exceed 64 KB. If your cdplayer.ini should be full then it can not hold any more disc data. In that case you can simply rename it and start all over with a new one. Discs.txt does not have that limitation).

Shut down the computer when finished

This function only works with certain mother boards that support automatic "power off" to the computer. If the board doesn't support this operation, then only Windows will be shut down. You may find this function useful if you want to read in and create some MP3's but at the same time are in a rush to the office/school or bed! Audiograbber waits for two minutes after reading and then shuts down the computer. The result of the reading can then be read from the copyinfo.txt file, which will be found in Audiograbbbers program catalogue.

Eject disc when finished

The disc will be ejected when all tracks has been ripped if this option is selected.

Track list color

The font color for the tracklist is usually blue but it is also possible to select your own color for this list (In case you should be using blue as background color for windows for example).

The values corresponds to what you have set in the control panel, display properties, appearance.

Window text is the font color of standard window text.

Highlight is the background color of a selected item in the tracklist.

Active caption is the color of the title bar of an active window.

Inactive caption is the color of the title bar in a not active window.

- **Misc 2**, with the following settings:

Use internal sound effects

Audiograbber will play a little click sound when the buttons on the main form are clicked.

Allow multiple instances of Audiograbber

Multiple instances lets you run more than one audiograbber at a time. You can have two copies of Audiograbber.exe in separate directories and they will then be used with individual settings. This can be useful if you have more than one CD-ROM drive. It is however not recommended that you rip from two programs to the same harddisk at a time. That would make the read head of the harddisk to move a lot between the two files ripped to and may cause problems. Two programs ripping to different harddisks may work however.

Continuous ripping

This is an interesting feature. You need to run two copies of Audiograbber for this feature to be of any use. If you rip from one copy of Audiograbber then it will post a message to the other copy of Audiograbber when finished. The other copy of Audiograbber will then start ripping as soon as the first one is finished. When the second copy of Audiograbber has finished ripping it will post a message to the first one. If there has been a new disc inserted in the first Audiograbber's selected CD-ROM drive then it will start ripping again. If it is still the same disc which has already been ripped then no action is taken. This feature is intended to be used with two copies of Audiograbber, if you are running more than two copies of Audiograbber then it is undefined which of the other copies that will continue ripping.

By using this feature and have the option to "select all tracks by default" and have the Auto Insert Notification for the drives selected in Control Panel then you will be able to not do anything more than just changing discs. Rip, normalize and encode without any clicks at all! Just insert a new disc once in a while.

Replace spaces by underscores in filenames

If selected then all spaces in filenames will be replaced by underscores. Some people that moves their files to a Unix system has asked for this feature.

Prevent screensaver from starting while ripping

A screensaver can quite often be CPU intensive and it is not good if it "steals" valuable CPU time during the ripping process as this can cause "speed problems". It is also a nuisance to disable/enable a screensaver manually before and after using Audiograbber.

Select this option to prevent the screensaver from activating during the ripping process.

Show splashscreen

Selects if Audiograbber shall show its startup picture when it starts. Audiograbber will not start any faster even if this option is turned off.

Soundcard

Tells Audiograbber which soundcard to use for playing audio in the [advanced normalizing function](#). This is also the soundcard that is hooked to the volume meter.

CD-ROM access method

[ASPI settings](#) (this is the preferred rip method).

[MSCDEX settings](#)

[Analog](#)

Other useful programs

Internal encoders

Audiograbber does not come with any MP3 encoder of its own but it can use some other encoders. It is recommended that you download and install **LAME**'s freeware MP3 encoder dll and use that one with Audiograbber. LAME is fast and gives very good sounding MP3 files.

Download LAME's MP3 dll from <http://dkutsanov.chat.ru>. Unzip the lame_enc.dll to the same directory as audiograbber.exe and you will have an internal MP3 encoder that goes all the way up to 320 Kbit/s in Audiograbber.

LAME's official homepage is at <http://www.mp3dev.org>

Another good encoder that works good with Audiograbber is **BladeEnc**. This encoder is also freeware so it won't cost you a dime. It is far from as fast as Xing's encoder but many people thinks this encoder produces better sound quality.

<http://bladeenc.mp3.no>

A new interesting compression format is **OGG Vorbis**. It gives better sounding files than MP3 and the main advantage is that it is completely royalty free and open source (Fraunhofer has some patents regarding MP3 in some countries). Get an OGG dll to use as internal encoder in Audiograbber from <http://www.audiograbber.net/vorbisdll.zip>. OGG Vorbis homepage is at <http://www.vorbis.com> and you need to go there to get a plugin for Winamp and other players to play OGG files.

Other encoders that are supported internally by Audiograbber are [Fraunhofers MP3 acm codecs](#) and Qdesigns MP2 acm codec. Microsofts [Windows Media Audio V2](#) is also supported (which gets installed if you install Winamp).

External encoders

Xing technology has made a very fast MP3 encoder that works good as external codec in Audiograbber. You must have their registered version and use x3enc.exe as external encoder (x3enc.exe does currently not come with their trial version). It cost however only \$19.95 to register (27 September 2000).

<http://www.xingtech.com/products/mp3encoder/>

A freeware MP3 encoder is called **Pluggger+** and works fine with Audiograbber. It's fast too!

<http://members.tripod.com/~mp3nkoder/>

GOGO MP3 encoder (freeware): This one is based on LAME's sourcecode and it has been optimized for speed. Homepage http://homepage1.nifty.com/herumi/gogo_e.html Download:

<http://homepage1.nifty.com/herumi/soft/gogo2/gogo235-win-con.zip>

MP3Enc: Fraunhofers new standalone encoder, slow and expensive (\$199!) but with good sound quality.

<http://www.opticom.de>. Download a demo version from

http://www.sonicspot.com/mp3enc/mp3encdemo_3_1_win32.zip

Fraunhofers **L3enc** was previously used to create MP3's. It can no longer be downloaded from

<http://www.iis.fhg.de/audio> but I keep the link here for nostalgic reasons.

Monkey's Audio (freeware). This is another interesting format that differs from the others in the way that it is lossless compression. Lossless means that a file can be encoded and decoded back without losing quality, just like the .zip format. MP3 is lossy compression which means that some data is lost during encoding and it can never be fully restored. Lossless will of course not compress as much as lossy but Monkey's Audio compress the file by around 50%. And with Monkey's Audio you will not have to worry/listen for artifacts in the song. There are simply no artifacts since what you hear is identical to the original wav file. You can encode and decode as much as you want without losing quality!

<http://www.monkeysaudio.com>. (Set to Mac.exe as external encoder in Audiograbber).

FAAC (freeware). This encoder makes AAC (Advanced Audio Compression) files.
<http://faac.sourceforge.net>.

MPG+. http://www.stud.uni-hannover.de/user/73884/audiocoder_eng.html.

There has also been a general Encoder plugin released by Alexander von Gostomski. With his plugin you can use both AAC, **VQF** and a **Real Audio** encoder! The homepage has unfortunately vanished but the plugin can still be found here:

http://www.audiogalaxy.com/software/plugins/windows/jaep_plugins.zip

You can also check out <http://www.audiograbber.com-us.net> and see if there are any new encoders available.

Miscellaneous

Cool Edit is one of the best programs for general editing of audio files and it can be found here:
<http://www.syntrillium.com/>

Winamp is a good MP3 player. <http://www.winamp.com/>

MuzicMan is a good MP3 player and organizer. <http://www.muzicman.com/>

Helium is a good ID3 tagger program which lets you edit and view ID3 tags.
<http://www.intermedia.st/helium/>

DR. Tag is another good ID3 tagger. <http://www.asterius.org/>

Hints

- CD-Ripping is a jungle! It usually works fine without the need for special adjustments, but not always ... When the process doesn't work it's often hard to be able to suggest a definitive remedy other than trial and error. Try both the [ASPI](#) and [MSCDEX](#) methods to see which works best on your system. If you still experience problems, try with another computer. When you find a working system, swap CD readers to see if the problem lies with your computer or reader. By a process of elimination, you can maybe determine what the problem is. *Some CD readers quite simply can't read digital audio which is not the same as being able to play a track!*
In the worst case you will have to use [Analog](#) as copy method.
- If the CD-ROM drive reads audio very slowly (at 0.1x something) then it often helps if the CD-ROM is moved to the secondary IDE connector and is the only device located there.
- If wave files are incorrectly read and are found to contain pops and clicks then it sometimes helps if you click the box "Synch data transfer" under Settings -> Control Panel -> System -> Device Manager -> CD-ROM. This may help for both IDE and SCSI drives.
- If you get poor ripping results you will hear that as clicks and pops in the final file. In that case do not disturb Audiograbber when it is reading from the CD. Whilst Audiograbber itself will not develop any problems, the device driver between Audiograbber and the CD player may.
- If you are about to buy a new CD player put Audiograbber and the ag12free.dll / ag12full.dll on a floppy disk, go to the retailer and test the CD player before you buy it. There is a list of tested CD's on <http://www.audiograbber.com-us.net/cddrives.cgi>
- You can drop wav files from Windows Explorer into Audiograbber's main window to easily normalize / create MP3's (assuming you already have the wav's.)
- If you want to record many different cd's and encode them at a later time you can use the option "Append ID3 info to wav file" under MP3 Settings in combination with "Use sub directories only for MP3 and WMA" under General Settings. Then all your ripped wav files will be stored in Audiograbbers base directory. The final filename/destination and ID3 tag is taken directly from the wav file when it is later encoded.
- If you want a track span over more than one song you must first double click the last one of them. Check the end frame for that track and double click the first one. Change the end frame of the first track to the value you got from the last one. *Note: This can only be done with the full version. The free version won't let you go outside the original start / end frames.*
- If you get ASPI error on the last track it usually helps if you doubleclick the track and lower the last sector a little. Try with 75 which equals one second.
- No, I don't know why some CD readers occasionally gives an ASPI error. You have to test yourself to see if you can find any settings that make your CD reader work reliable.
- If you hear sound from your speakers but the volume meter on Line in sampling doesn't move then you shall press the mixer button and select the correct recording source. You may need to click properties and make sure that all volume controls are shown.
- You need a free MP3 encoder? Download [Lame's or Blade's DLL](#) which works damn good with Audiograbber!

Credit goes to:

I would like to thank the following persons for direct or indirect help with this program:

- **Michiel Overtoom**, For releasing the source code to DIDO, another CD-ripper.
- **Simon Chang**, for providing a way to thunk down from 32-bit to 16-bit code.
- **Jordan Russell**, for an excellent component for the toolbar buttons.
- **François Piette**, for his Internet Component Suite.
- **Steve Scherf** and **Ti Kan**, for making the CDDB server.
- **John Mertus** for the AudioIO component. **Dean Bowers** that converted the component from Delphi to Builder.
- **Ian Kennedy**, For helping me with the calls to Fraunhofers CODEC.
- **Clark Tisdale**, for tips on the volume meter.
- **Ivan Azic**, for an enhanced status bar.
- **Kenn Nesbit**, for a component that simplifies the read / write to .ini files.
- The beta testers, especially **moonroy** has been very thoroughly and demanding ;-)
- **Hillary Cutler**, **Sean O'Neill**, **Paul Genins** and **Randolph MacKenzie**, for help with the english translation.
- **Andy Key**, for helping with the calls to DeviceIoControl for internal ripping under Windows NT/2000. Some sourcecode is found at <http://www.geocities.com/SiliconValley/Byte/7125/>
- This version of Audiograbber has been packed with UPX, <http://www.upx.tsx.org>, to reduce the file size. UPX is made by **Markus Oberhumer** & **Laszlo Molnar** and it reduced the file size from 2.9 MB to 830 KB!

If anyone of you above would like to have the full version of Audiograbber, just mail me and I will send you one for free.

AUDIOGRABBER

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How to buy Audiograbber

[Buying Audiograbber](#)

[The difference between the full version and the free version](#)

[Version information](#)

Fraunhofers acm CODEC

An acm codec is a kind of program that works almost like a DLL. That means that it has functions that can be called from other programs but it can not be run separately.

Fraunhofer has made such a codec for MP3 creation and it is a lot faster then the L3enc program. The codec comes in two versions, the advanced and the professional. The advanced version was previously shipped with Microsofts Netshow which is free but unfortunately it supports only low bitrates for encoding. It does however support higher bitrates for decoding.

The professional version is better and supports bitrates up to 128 Kbit for encoding. This one has to be purchased and is quite expensive however... Contact [Fraunhofer](#) directly for price information. (It seems as Fraunhofer now has stopped distribution of this acm codec for good. I believe the reason is all the software piracy that went on with this codec. Don't ask me for information where to find a pirated version of that codec, I won't tell you. Use [LAME or BladeEnc's freeware DLL](#) instead).

Last minute update: The acm codec is sold on http://www.opticom.de/order/order_internat.htm. It is very expensive though.

The filename of the codecs are l3codeca.acm and l3codecp.acm. They should be placed in the c:\windows\system directory. The file c:\windows\system.ini must also be updated in the section [drivers32] with this:

```
[drivers32]
msacm.l3codecp=l3codecp.acm ;for the professional one.
msacm.l3acm=l3codeca.acm ;for the advanced one.
```

If you want to install them in Windows NT they should be placed in the c:\windows\system32 directory. You also need to change in the registry. The easiest way to do that is to create a textfile with the following and name it to l3codec.reg. Then you just have to doubleclick it.

```
REGEDIT4
[HKEY_LOCAL_MACHINE\SOFTWARE\Microsoft\Windows NT\CurrentVersion\drivers.desc]
"L3codecp.acm"="Fraunhof L3 Codec Professional"
[HKEY_LOCAL_MACHINE\SOFTWARE\Microsoft\Windows NT\CurrentVersion\Drivers32]
"msacm.l3codecp"="l3codecp.acm"
```

You can look in the Control Panel -> Multimedia and the advanced tab to see what sound compression codecs you have installed.

Audiograbber can use these Fraunhofer codecs to create MP3's in a way that it looks like Audiograbber is making the compression itself.

Only 56 Kbit/s?

Audiograbber does not come with any MP3 encoder of its own but it can use some other MP3 encoders it finds on the computer. It has now found Fraunhofers advanced MP3 encoder on your computer. The only "advanced" with this encoder is its name though, it really should have been named simple instead. Don't misunderstand, this MP3 encoder produce great sound for its bitrate but the problem is that it goes only up to 56 Kbit/s. The common bitrate for MP3 files is 128 Kbit/s and even if 56 Kbit/s may sound just fine on cheap computer speakers it will not sound perfect when played on a quality stereo system.

Fraunhofer has also made a professional MP3 encoder which is better but it is no longer sold. See the section [l3codecp.acm](#) to understand more about Fraunhofers acm codecs that works with many Windows programs, including Audiograbber.

Anyhow, it is recommended that you go to <http://www.audiograbber.com-us.net/download.html> and gets a better MP3 encoder. I recommend LAME's freeware MP3 encoder dll which gives very good sounding MP3 files and at bitrates all the way up to 320 Kbit/s.

Lost username and password

If you have lost your username and password for Audiograbber then send an e-mail to jackie@audiograbber.com-us.net. Include your full name and the e-mail address you used when you purchased Audiograbber. You can also tell me your street address and town to make it easier for me to locate and resend your download instructions. If you still have a copy of the full Audiograbber on your computer then you can also send me the StartCode= line from the audiograbber.ini file so I can check what your serial number is and resend your download instructions.

Keyboard shortcuts

A keyboard shortcut means that it is possible to simultaneously press Ctrl and a key to reach a function. Audiograbber supports the following keyboard shortcuts on the main window:

- A Select All.
- N Select None.
- T Toggle track numbers (on or off) in the tracklist.
- P Play single track (the currently highlighted track) or Pause when a track already plays.
- S Stop.
- L Play the checked tracks, starting at the highlighted track.
- R Play random tracks among the checked tracks.
- B Back to start of the track or previous track.
- F Next track.
- E Eject or insert disc.
- K Capitalize tracknames.
- G Grab checked tracks.
- Enter Opens the track properties window for the selected track.

ASPI

This lets Audiograbber know which preferences are to be used when reading digital audio via ASPI calls.

MSCDEX

This lets Audiograbber know which preferences are to be used when reading digital audio via MSCDEX calls.

Analog

This lets Audiograbber know which preferences are to be used when reading digital audio via Analog sampling.

CD-ROM unit

Choose here which CD reader should be used for Analog sampling.

Input Soundcard

All soundcards available on the computer are listed here. Choose which soundcard to use for Analog sampling. This soundcard will also be used for line in samplings.

Don't change volume

Select this option if you don't want Audiograbber to change the input volume of your soundcard. Make sure that you have selected your CD-ROM drive as an input source for the soundcard if this option is used.

Set to

Select this option to make Audiograbber set the input volume of your soundcard. Audiograbber will also make sure that the CD-ROM drive is selected as the input source for the soundcard.

Input volume

Drag this meter to a desired input level for your soundcard. You will get best results if the peak level for a track is around 98% when the track is sampled.

Mute Speakers

Select this option to mute the speakers while analog sampling is performed.

CD-ROM unit

Choose here which CD reader should be used for ASPI calls. If you change CD-ROM drives here, Audiograbber automatically alters other alternatives which it thinks best suite your needs. It may be necessary here to change these alternatives manually.

ASPI manager

This option is available if the computer has a working ASPI manager (wnaspi32.dll) installed. Win 95/98 comes with an ASPI manager but in NT and Win 2000 you will need to install a third party (Adaptec) ASPI manager yourself or use the native NT calls option.

Use native Win NT/2000 calls

This option is available under Win NT and 2000 only. With this option selected Audiograbber translates the calls that normally goes to the ASPI manger and sends them directly to Windows instead.

(DeviceIoControl calls). In most cases it really doesn't matter if ASPI manager or native calls are used but ASPI is theoretically slightly better to use if both options are available.

Rip to RAM

Check this box if you want Audiograbber to rip as much as possible of temporary wave files to internal RAM memory instead of the harddisk. The advantage is that ripping goes smoother with reduced "possible speed problems" . Wave files are flushed to harddisk when the track has been ripped (if the program is setup to create wav files at all).

Max allocated memory

Tells how much memory Audiograbber will maximally allocate for temporary files in RAM. The first 32 MB of memory is always reserved to Windows so if you for example have 128 MB RAM you can maximally set 96 MB here. Audiograbber does not allocate more memory than the largest track takes and the memory is released when ripping is finished.

CD-ROM type

The command set used for accessing the CD reader. All IDE CD-ROMs use the same command set but SCSI use several different ones. Because the SONY command set is the most common, all unknown SCSI drives are set here by default.

Rip method

Burst copy should be the best and fastest, however, some CD readers may display problems with it. If you do experience problems, first try using dynamic synch width, there after with the others. Burst copy doesn't use any synchronizing ("jitter correction".) The other rip-methods do use this.

Digital Audio Extraction speed

Not all CD readers support choice of reading speed. Speeds here are intended for those readers that claim to support it. Whilst many claim to have this function, the reality is often not the case.

Soundcard

All soundcards available on the computer are listed here. Audiograbber uses the soundcard when it plays a wav file in the preview function. It can change the CD volume for the choosen soundcard from its main window.

Detect

Rescans the SCSI bus for drives and refills the CD-ROM unit box.

Start offset

Some CD readers exhibit problems in finding the exact start/stop reading points on a CD. If you find it necessary, you can manually adjust the program to read a few frames earlier or later. A maximum of 150 frames can be set (75 frames=1 second.)

End offset

Some CD readers exhibit problems in finding the exact start/stop reading points on a CD. If you find it necessary, you can manually adjust the program to read a few frames earlier or later. A maximum of 150 frames can be set (75 frames=1 second.)

Discard

If your CD reader occasionally begins reading with a click, you can set an amount of frames to be read but not used. Max. 100 frames can be given.

Don't calculate checksum during the rip process

Audiograbber can experience problems with the re-calculating of data if a very fast CD reader is being used. This is most common when using the ASPI rip method and burst copy. By not calculating the checksum of a wave file valuable processor time can be saved and the problem avoided.

Use internal sound effects

With this option selected, Audiograbber will play a click sound when the big buttons on the main form are pressed. There is one sound when the button goes down and another one when the button goes up.

Allow multiple instances of Audiograbber

Normally only one session of Audiograbber will run at a time. Selecting this option will allow multiple sessions to run simultaneously. The benefits and drawbacks of this are explained in more detail in the help file.

Continuous ripping

If you run two sessions of Audiograbber simultaneously with this option selected, the second session of Audiograbber will continue ripping after the first one has finished.

Replace spaces by underscores

Select this option to have all spaces " " in filenames replaced by underscores "_". This is useful if the files are going to be moved to a Unix computer.

Disable screensaver when ripping

A screensaver can quite often be CPU intensive and it is not good if it "steals" valuable CPU time during the ripping process as this can cause "speed problems". Select this option to prevent the screensaver from activating during the ripping process.

Show splashscreen

Uncheck this box if you do not want Audiograbber to show the splashscreen when it starts. It will however not start any faster with this option unchecked.

Shut down the computer when finished

If so required, Audiograbber can shut down the computer when it has finished. On certain modern mother cards the power supply can also be turned off. If this function is activated, Audiograbber waits two minutes after finishing it's job and then tries to shut down the computer. The result of all copying can then be read in the copyinfo.txt file, residing in the Audiograbber program catalogue.

Eject disc

Opens the diskdrive and unloads the discs when ripping is finished.

Select all tracks

Check this box to put a checkmark in front of every track when a disc is refreshed. Uncheck the box to have all tracks unselected as default.

Autoplay after grab

If this box is checked then Audiograbber will start playing the tracks after the ripping is finished. The tracks that are played are the ones that have been ripped and are played in random order (if you choose to play the grabbed files then the settings in your MP3 Player dictates how they are played).

Say ripping complete

Check this box if you want Audiograbber to say "Ripping complete" when it has ripped all tracks.

Naming

Select what parts the filenames shall consist of. You can move the fields up and down with the arrows. Selected fields are separated by a space, a dash and another space.

Arrow

Press this arrow to move the selected item on position up.

Arrow

Press this arrow to move the selected item on position down.

File name items

Check the individual fields you want to be included in the filename. You can move the items up or down with the arrows to the left.

Directories

If you want to have your files stored in sub directories based on the artist and album names, select here.
Audiograbber will create the directory if necessary.

Advanced names

Check this box if you want more control over your filenames. Press the info button to get information on how to specify the names.

Advanced naming string

Type your file name options here if you want more flexibility naming your files. Press the info button to get information on how this name format string should be defined.

No Sub directories for wav files

This option is primarily intended to be used in conjunction with the option "append ID3 info to wav files" found under MP3 settings. Selecting it lets the optional ID3 info and track names in the wav files get the correct directory structure but leaves the wav files in the base directory.

Spin up disc

Some crappy drives has problems to start reading from a non moving disc. By checking this box Audiograbber will try to spin up the disc for 3 seconds before it starts reading.

Shorten last track

Check this box to automatically skip the last 10 frames of the last track. CD-ROM drives gives an ASPI error if they read outside the end of the last track. Some drives are not very good at positioning the laser exactly correct and it is a risk that they will go outside the last track. Check this box to be on the safe side.

Partial rips

Check this box if you want to make short sample clips of the tracks.

Rip only

Tells the duration in seconds of each clip. The clip will never be longer than the original track, no matter what value you enter here.

Start at

Tells how many seconds into a track ripping will begin. If this value is too high so that the remainder of the track is shorter than the specified time in the "rip only" box, the start point will automatically be set earlier. This occurs because Audiograbber always tries to rip the time specified in the "rip only" box.

Fade in

If the rip selection does not start at the beginning of the track and this option is selected, Audiograbber will "fade in" the first two seconds to create a less abrupt start.

Fade out

If the rip selection ends before the end of the song and this option is selected, Audiograbber will do a "fade out" the last two seconds to create a less abrupt ending.

Show detected speeds

This option works for IDE drives only. Audiograbber tries to ask the drive what DAE speeds it supports and displays the speeds it responds to.

Show all speeds

This option works for IDE drives only. When show all speeds are selected, all speeds are displayed. You will see yourself if the drive later responds to the speed you set.

Show direct in tracklist

State which value is to be displayed in the information column in the track list.

Copy result

Shows how the copying went. Cannot be removed.

Checksum

Shows the checksum calculated for the track.

Rip speed

Shows how fast the track was read compared to the speed it is normally played. At least 1x speed should be attained. Some readers however can go up to more than 20x speed!

Lost synchs

How many times it has not been able to synchronize between two blocks during reading. The value should be 0 in an error free file. Though this is no guarantee in the real world! Synchronization is not used at all with burst copy (only with ASPI.)

Possible speed problems

This value is only used with burst copy and ASPI as the rip method. When using burst copy, it's important that the CD reader delivers data in an even stream. When using a very fast reader, the computer can experience problems in keeping up. If possible, you should choose a lower reading speed, or choose the "don't calculate checksum" during the rip process. (under general settings, tab miscellaneous.)

Old peak value

Shows what the peak value was on the track on the disc.

New peak value

Shows the new peak value after an eventual normalization. This value is only calculated if the peak value has been changed.

Old average output level

Shows what the average output level was on the track on the disc. This is only displayed in case advanced normalizing has been used.

New average output level

Shows the new average output level for the track. This is only displayed in case advanced normalizing has been used.

Old compact ratio

Shows how compact the song was. This is only displayed in case advanced normalizing has been used. The higher value the more compact song and 100% is the maximum.

New compact ratio

Shows the new compact ratio for the song. This is only displayed in case advanced normalizing has been used. The higher value the more compact song and 100% is the maximum.

Rip offset

If your CD reader has problems finding the exact start and stop points of tracks, different offsets can be given here.

Miscellaneous

Here you'll find various settings that didn't fit into any other category.

ASPI manager

The name of the ASPI manager for the currently selected CD-ROM drive.

Next tip

Displays next tip.

Previous tip

Displays previous tip.

Show tips at startup

Uncheck this box when you're tired of all the tips. You can also reach this tip box from the help menu.

OK

Close this dialog box.

MSCDEX

MSCDEX is a tried and trusted method for reading digital audio from a CD reader. Originally it comes from the good ole DOS days but works fine with Windows 95. Unfortunately with the OSR2 release of Windows 95 and Windows 98 it is necessary to change the [scsi1hlp.vxd](#) file to the original version.

MSCDEX calls are not accessible with Windows NT, rather one needs to use [ASPI](#) calls. MSCDEX calls don't work with SCSI units, so again [ASPI](#) calls need to be used.

When the program uses MSCDEX calls to read a certain number of sectors/frames of audio, everything freezes and the program must wait until the reading of these frames is complete. During the time when the program then writes these frames to the hard disk, the compact disk manages to rotate a bit, meaning that synchronization is necessary. This however is normally corrected for by Windows caching a little data in advance.

One example is that Audiograbber asks for sectors 1-26. The CD-ROM reads these sectors and passes them to Audiograbber. During the time it takes for Audiograbber to convert them and write them to the hard disk the cd has managed to rotate a little. Windows 95 however takes care of this and caches a little data from the CD. When Audiograbber then asks for, and receives, sectors 25-50 it thinks they have come directly from the CD. In reality though sectors 25-27 have come from the cache memory! Only when the program asks for sector 28 does it receive it from the CD (These numbers are only illustrative!) In this case, Audiograbber thinks that synchronization should occur between sectors 25-26 and detects this to be true. A problem however may occur with the synchronization of sectors 27-28 and a small "click" can occur. Normally, the problem is solved, if however it doesn't sound good, install a [real mode driver](#). The data here will not be cached, all data will be read directly from the CD. In addition you could try using [ASPI](#) as a read method, again caching is not used here.

The following preference under General settings dictates how many frames are read at one go with the MSCDEX call:

- Read audio in blocks of. This tells how many frames of audio should be read at each chunk and how many frames should be used for synchronization. Every frame is 1/75 second. You can read more about [digital audio](#) and how the reading is done.

ASPI

ASPI is a relatively new way of reading data from a music CD. Originally it was devised for use with SCSI units but often works well with IDE CD-ROMs. (via a command set known as ATAPI.)

Certain versions of ASPI however don't seem to be able to detect IDE CD-ROMs. It appears that the `wnaspi32.dll` file between versions 4.01 and 4.53 doesn't work with IDE. Earlier and later versions do work though. To see which version is installed on your computer, right click on it with the mouse and chose properties, version.

Windows 95/98 originally came with a version of ASPI that worked with both SCSI and IDE. Unfortunately this is not the case with Windows NT and one is forced to install your own. If you don't have one read these [instructions](#) on how to obtain one. Audiograbber 1.62 and higher can now translate the ASPI commands and let Windows NT and 2000 internally send the calls to the drive instead so an ASPI manager is no longer needed. An ASPI manager is still slightly better because it can handle more than one call at a time.

The configurations needed by Audiograbber to read via ASPI are as follows:

- **Call drive via.** Choose if you want to let the ASPI manager send the commands to the CD-ROM drive or if Windows NT shall do it instead. The ASPI manager option is normally always available in Windows 95/98 but not in Windows NT. You need to manually install an ASPI manager in NT to enable this option. The Win NT/2000 calls option is never available in Windows 95/98 but is supposed to always be available in Win NT/2000. ASPI manager is theoretically slightly better to use.
- **CD-ROM unit.** The CD readers that Audiograbber has detected via the ASPI interface are displayed here.
- **CD ROM type.** The command set to be used for the CD reader . Unfortunately, different CD readers use ASPI calls in different ways. All IDE CD readers use ASPI calls in the same way. This is not the case with SCSI units. Those SCSI CD readers that are recognized are listed with the correct drive type. Unrecognized readers are listed as SONY by default (As SONY is the most common.) If however your SCSI drive doesn't work, you can try another call method. If no call method works, send an e-mail to your CD-ROM manufacturer and ask them to send a description of their command set for audio extraction to me.
- **Rip method.**
 - Buffered Burst copy** means that no overlapping or synchronization occurs. This is the fastest method of copying and should be attempted first. With this method it's important that the CD reader is requested for data continuously. As long as the reader doesn't need to reposition it's head and Audiograbber asks for 3 seconds of music and processes all the data in step with the process, this shouldn't be a problem. Directly after the reader has delivered some frames, Audiograbber requests some new ones, in this way a schedule of 3 seconds is maintained. If Audiograbber (or the computer) is unable to process the incoming data, the 3 second buffer will diminish. If the buffer is reduced to 0 a "possible speed problem" will occur, which generally means that the reader has been "idle" for a time. A "[possible speed problem](#)" doesn't necessarily mean that there is a problem, it is only a warning signal. If the CD reader reads at 8x speed Audiograbber's buffer will be filled in less than 0.4 seconds. It is therefore important that no other program interferes. The slower the reading speed is, the less danger the buffer underrun is exposed to.
 - Unbuffered Burst copy** is basically the same as buffered burst copy but the CD-ROM drive is never asked for more than one block at a time. Theoretically the buffered burst should be better but testings have proven that this copy method is better for some drives.

Dynamic synch width means that sector synchronization is occurring. The buffer that's used to synchronize the previously read block is dynamic and grows or reduces in size as required. If synchronization fails, Audiograbber re attempts the process with a slightly larger overlap area.

Fixed synch width means that a set amount of frames are used for sector synchronizing. Some CD readers seem to work best with this alternative, though theoretically dynamic synch width should yield the best results. "When in the real world "

- **DAE speed**, Digital Audio Extraction speed. This block is occasionally filled by the value requested from the reader as to which speed it supports. Different readers can manage different speeds, but often the speed they can actually read at isn't the speed reported to the program. In addition this only works with IDE readers. I have sent an e-mail to Plextor and asked how to adjust the reading speed for their readers but have received no reply If you experience problems whilst reading and are able to decrease the read speed, do so.
- **Show speeds**, This option works only with IDE drives. If "show detected speeds" are selected Audiograbber will ask the drive what DAE speeds it supports. Some drives does not respond to speed selection before the reading actually starts and in that case you can try with all possible speeds. It can also be a good idea to use "show all speeds" in case you already know what speeds works for your drive, Audiograbber will then start just a little faster.
- **Rip to RAM**, Check this box if you want Audiograbber to rip as much as possible of temporary wave files to internal RAM memory instead of the harddisk. The advantage is that ripping goes smoother with reduced "possible speed problems" . Wave files are flushed to harddisk when the track has been ripped (if the program is setup to create wav files at all). The maximum RAM Audiograbber will attempt to use depends on how much memory the computer has. The first 32 MB is always reserved for Windows and other programs so if your computer has 128 MB RAM it will be possible to use up to 96 MB to rip to. Audiograbber does not allocate more RAM than it needs to hold the largest selected track and the memory is released when all track has been ripped and optionally normalized and encoded.

ASPI for Windows NT

Note: Audiograbber 1.62 and higher can rip through native (internal) Windows NT and 2000 calls so an ASPI manager is no longer needed for digital ripping under NT and Win 2000. Audiograbber can however not measure possible speed problems without an ASPI manager so that meter is set to zero with native NT calls.

When being run with Windows NT, Audiograbber reads data via ASPI calls for both IDE and SCSI CD-ROMs. Unfortunately, no ASPI manager is distributed with NT, so a manual installation is sometimes required. If you have a SCSI reader you probably already have an ASPI manager installed. There is currently an ASPI manager delivered with Windows 95 and 98 which works with both IDE and SCSI drives but it may have been replaced by some other software. In case you need an ASPI manager you can find a good freeware one here that works with both Windows 95/98 and NT:

<http://www.grc.com/freesetuff.htm>

(Direct link: ftp://grc.com/aspi_me.exe)

UPDATE: The very good and reliable aspi_me ASPI manager seems to be no longer available from the above link. Maybe you can find it somewhere else on the internet if you search for aspi_me. If you find it then you need to set the data back to 1998 when you install it since the installation program checks for the date for some reason. Correct the date as soon as the installation program has started.

Here is another old one that works for windows NT:

ftp://ftp.irnet.ru/pub/Windows/NT/ASPI/aspi_update.zip

It appears to work with NT4 even though it was written for NT 3.51. The installation instructions may however be inaccurate. You can clip out the text given below (at the end of this page) and save it as a file called aspi.reg. By Double clicking on the file the registry will be updated. Either way, be sure to read the installation instructions that came with the aspi_update.zip. Pay close attention to the ExcludeMiniports part as it is essential in making it work.

You can also try to get an ASPI manager from Adaptec at

<http://www.adaptec.com> Or get it directly from: ftp://ftp.adaptec.com/software_pc/aspi/aspi32.exe

Here is an article about how to install an ASPI manger without having any Adaptec hardware:

http://www.datman.com/tbul/dmtb_028.htm

REGEDIT4

[HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\ASPI32]

"ErrorControl"=dword:00000001

"Start"=dword:00000001

"Type"=dword:00000001

[HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\ASPI32\Parameters]

"ExcludeMiniports"=""

NO CD-ROM drive access

This error occurs under the following conditions:

- You don't have a CD-ROM installed.
- You are running Windows NT but you have not installed any [ASPI](#) manager and MSCDEX calls can not be used under Windows NT.
- You are running Windows NT and have an ASPI manager installed but it hasn't been started.
- Audiograbber was unable to allocate enough memory in the low memory area (640 Kb.) As a result [MSCDEX](#) can't be used. In addition you have no ASPI manager installed. In this case, try re starting Windows and start Audiograbber before any other program.
- You are running Windows 95/98 but have no [ASPI](#) manager installed and Audiograbber can't find it's MSCDEX-DLL.

CD-ROM tab

Use these tabs to set the rip method for your CD-ROM drive. If the ASPI tab is grayed out it means that no ASPI manager is loaded (wnaspi32.dll). If the MSCDEX tab is grayed out it means that you are running Windows NT or that MSCDEX could not be loaded. If Analog is grayed out it means that there is no soundcard available.

Various settings tab

Use these tabs to set various options for Audiograbber.

Freedb server

The internet domain name of the database to ask for disc information.

Freedb server

This is where the server is physically located. Select one close to where you are.

Get list

Press this button to get a list of available freedb servers.

Connect via

Select whether you want to connect to the disc database via TCP/IP or HTTP.

Direct TCP/IP

This should be the fastest way to retrieve disc information.

HTTP

Not all servers support connection through HTTP. Select another freedb server if this option is disabled.

Proxy settings

Proxy can only be used under the HTTP protocol. As a rule, you shouldn't use this alternative but if you are accessing internet behind a firewall this can be used. If your proxy server needs username and password you can enter this in audiograbber.ini:

ProxyUsername=jackie

ProxyPassword=hacker

and replace jackie hacker with your own username and password.

Use Proxy

This box is only enabled if you select the HTTP protocol. Use it if you are accessing internet from behind a firewall.

Proxy port

The port number you use for proxy access.

Proxy Server

The IP address for your proxy server.

E-mail settings

The freedb server wants to know your e-mail address so enter it here.

Credits

These are the guys that originally made the Cddb (freedb) server software and protocol.

Freedb settings

Freedb is used for retrieving disc title and track names from an internet database. This means of course that you have to be connected to internet to use this function. It is a huge database available on the internet where individuals has submitted their disc data. Programs like Audiograbber can connect to this database to see if the disc is found in the database and download various disc information.

[Freedb](#) started out as a hobby project a few years ago and was named CDDb, acronym for Compact Disc DataBase. The whole project was open source, gratis and released under GPL license. In 1998 or 1999 a company named Escient somehow took over the project and they have since then tried to cash in on the original CDDb project. Now in 2001 Escient has changed name to Gracenote (or if they was acquired, I don't know) but Gracenote now has the rights to the original CDDb database. Gracenote has also started a slightly improved CDDb version named CDDb2 which is far from free or open source and Gracenote now tries to get developers like myself to sign a contract to use their CDDb2 database. They plan to stop the old CDDb database completely but they still keep it up for programmers who have signed up to use CDDb2.

I do not like their contract at all, it is far too complicated and restrictive, and I know that there are many persons who doesn't like that this whole project where individuals has built up the database has now become commercial. Fortunately this project was originally an open source project so in 1999 a similar service named freedb started. Audiograbber 1.80 has now been modified to connect to freedb.org instead of CDDb and since Gracenote has registered CDDb as their trademark Audiograbber has been modified to mention freedb on all places where it previously used CDDb.

Freedb does not have as many discs in their database as CDDb but now when more and more programs connects to freedb it is just a matter of time before that database also gets filled with a lot of discs. So, if you have some discs that are not found in freedb then please [submit](#) them so other users can find them on freedb in the future.

This freedb database is hosted on various servers around the world, so select one that is close to you. By pressing the "Get list" button, your computer will try to fetch the current list directly from the internet. If this fails for some reason (30 seconds timeout) Audiograbber will let you choose from the latest known list.

There are two ways to connect to the freedb server: Direct TCP/IP (also called CDDbP) and through the HTTP protocol. Both methods are about equally fast and generate the same result, so it really doesn't matter which you choose. If you are accessing internet from behind a firewall and must use proxy you will however need to use the HTTP protocol. It is not possible to use proxys under direct TCP/IP. Not all servers accepts HTTP transfers though. You can check the list (from "Get list") and check the HTTP-path to see if HTTP is supported.

If your proxy server needs a username and a password you will have to enter that manually in the audiograbber.ini file. Simply type these two line in audiograbber.ini:
ProxyUsername=jackie
ProxyPassword=hacker
and replace jackie hacker with your own username and password.

Audiograbber needs to send an e-mail address to the freedb server when asking for data so you will have to state your e-mail address (or at least something that looks as an e-mail address!)

AutoQuery freedb if a new disc is not recognized

If this option is selected and the disc is not found in Audiograbbers [database](#) or cdplayer.ini then Audiograbber will look up the disc in the freedb database on internet automatically. Audiograbber will only try once per disc, ie if the disc is not found in the freedb database then Audiograbber will remember that it has already asked about that disc and not ask again. It is recommended that you also check the "Autosave freedb queries" box.

Autosave freedb queries

If this option is selected then your successful disc queries from freedb will be saved in the database and cdplayer.ini automatically. (cdplayer.ini only if you have selected to use cdplayer.ini under General Settings, misc tab).

Path to local freedb database

It is possible to go to <http://www.freedb.org> and download the whole disc database and store it on your local computer. This way you will not have to go online to find most of the discs. If you use a local database then Audiograbber will first look for the disc info in its own database file discs.txt and if not found there it will look in cdplayer.ini. If still not found it will automatically look in the local freedb database. You will not need to press the penguin button to get the tracknames from the local database.

Discs found in the local freedb database will not be automatically saved in your discs.txt even if you have selected to "Autosave freedb queries", you will need to press the "Store disc in database now" on the CD menu to save disc info from a local freedb database.

If you decide to download the local freedb database then make sure you get the zip file with the database in Windows format. Audiograbber can use a local freedb database regardless if it is in Windows or Unix format but the Unix database has one file for each database entry and Windows is not so good at handling 200,000 files or more.

There is no checkbox to turn off the local freedb database as you can see. If you decide to not use a local freedb database anymore then press the Browse button and empty the text field with the directory name on top of the "Browse for database" window and press OK.

Audiograbber can also [Submit](#) (upload) disc info to freedb. This was new in version 1.50 of Audiograbber.

The homepage for the original CDDb™, Escient® and Gracenote is <http://www.cddb.com>.

Get from freedb

This function is found under the CD menu.

The purpose of this function is to retrieve the track names from an internet database. You can read more about what this is under [freedb settings](#). You can also retrieve info for more than one discs by using the [batch query](#) function.

Undo freedb Query

This function is found under the CD menu.

If you weren't satisfied with the answer from the [freedb](#) server you can then go back to the track names prior to the query.

Freedb batch query

Freedb is an Internet database that holds disc info such as artist and track names. Users from all over the world can upload and download disc info from this database and you can use this so you don't have to fill in the track names by yourself. Normally people download disc info for only the disc they are currently using but Audiograbber can also download info for a whole bunch of discs at once. That can be useful if you are on a dial up connection to Internet and have high telephone rates.

Insert a disc and press the "Store disc for later freedb query" option under the CD menu. Repeat that step for all discs you want to batch query. All these queries will be saved in Audiograbbers database [discs.txt](#). When you have stored all discs you want to query then connect to Internet and press the "Freedb query stored discs now" function on the CD menu. Audiograbber will now download info for all stored discs and you can disconnect to Internet when finished. It takes normally 5-10 seconds per disc. Then the database file will be filled with tracknames for the discs and Audiograbber will find that next time one of the discs is inserted.

Copy status

This function is reached by right clicking a track in the track list. Here you can tell Audiograbber what information to show in the information field.

Create M3U list

This function is reached by right clicking a track in the track list.

It simply creates an M3U list which is a play list that can be used by [Winamp](#). The M3U list is a plain textfile and is stored in your default directory. The tracks that are selected with a checkbox will be included in the M3U list. The name of the M3U list is taken from the album field.

Advanced Normalizing

Selects which options you want to use for normalization of your files. Normalization simply means how loud a song is played. Instead of adjusting the volume for each song you play, you can now ensure they will sound equally loud already when you rip them.

Use normalizing

Select this box if you wish to automatically adjust the recording level, leave this checkbox unchecked if you wish to copy the track exactly as it is. Also set the compression option to "Never" if you wish to get an exact digital copy of the tracks.

Peak level to

Select this if you want to use the peak level as criteria for how loud the song shall be. Peak level means the absolutely loudest part of the song.

Average output

Select this option if you want the output level adjusted, based on the average output level of the song.

Normalize to

Select the recording level between 0 -100%. 0% is equal to silence and 100% is the highest volume. 98% is recommended.

Average output to

Select how loud you want the song to be on average. 65% is recommended

Peak condition

Select if you want to exclude songs, within a certain range, from being modified. If this option or the average output condition is not realised the song will be neither normalized nor compressed. It will simply be left unchanged.

Average output condition

Select if you want to exclude songs within a certain range from being modified. If this option or the peak level condition is not realised the song will be neither normalized nor compressed. It will simply be left unchanged.

Compression

Compression essentially means making the sound more compact and giving the song a more equal output level. This is often used by radio stations to have the volume around the same level all the time.

Never

Choose this to not compress your songs.

Always

Choose this to compress all songs. This option is not recommended to be left in this state but can be useful if you are only modifying a few songs and you know what you are doing.

Only when needed

This option is only available if you have chosen to adjust the average output level. If the desired average level cannot be reached by ordinary normalizing it will use as much compression as needed to reach the desired average output level (but not more than the maximum compress value of course).

Maximum compress

This tells Audiograbber by how much it is allowed to compress the sound. A value of 60% means that every sample in the sound will get a new value that is within the range of 100% - 160% of the original value.

Don't compress songs

It is a good idea to not compress songs that have already been extensively compressed. Chose this option to leave already highly compressed songs in their original form. It is recommended that this option is used.

Max compact ratio

This is the limit for when a song is considered "already compact enough". Compact ratio basically means how close to the average output level the song is generally playing. For example classical music usually changes the volume a lot from low to high and does not stay on the average output level much. That gives a low compact ratio. Rock'n roll songs often play at the same output level throughout (often very loud!) which gives a higher compact ratio. 100% is max and would mean that there is no dynamic in the song. Recommended value here is 80%.

To new file

By choosing this option your original song will be left unmodified and a new modified song will be created. This option works only when you are in this dialog already. If you normalize/compress files from Audiograbber's main window, or drop them on the main window, it will modify the chosen file directly regardless of this setting. The same goes for ripped tracks.

Add file

Add files to the list from here. Newly added files will be placed on top of the list and not the bottom! The files must be .wav files, 44,1 KHz, 16 bit, stereo.

Remove file

Removes the selected file. They can also be removed from the list by pressing "Delete".

Modify

Press this button to start modifying the selected songs.

Test

This function will scan through the selected files so you can easily see what values they will get if they are modified with the chosen settings.

Preview

Press this button to view and listen to the last modified/tested song.

Abort

Aborts testing or modifying of the songs.

List

The order of the songs can easily be rearranged by drag and drop. Mark the songs you want to test/modify with a checkbox.

OK

Leaves this dialog box and performs the changes you might have made.

Note that you must leave the dialog in advanced state to have the advanced options used when ripping.

Toggle

Switch between the simple and the advanced normalizing/compression.

Advanced normalizing

This function is found under the settings menu.

Occasionally it simply isn't enough with ordinary normalizing. The peak level value does not say much about the average output level of a song; the average output is better to use as an indication for how loud the song is. That's why I made this function. I realized that I had to change the volume on my car stereo all the time when I played my own compilation discs. This function will put a clampdown to the movement of that volume button!

Audiograbber can now scan through a song and split the song into parts of 133 ms. The peak value in each 'frame' is then used to calculate an average output volume. This average is based on the 90% loudest 133 ms frames in the song, leaving the silent parts of the song out of the calculation.

This average value can then be changed up or down just like with ordinary normalizing. Usually with no problems.

Some songs do however have many silent parts and can not reach a certain average output level by only altering the peak level for the song. Here is where compression plays a role. Compression is something that radio stations and music studios use to even out the output level.

Basically, it is something like a continuous and rapid movement of the volume level, raising the silent parts of the song more than the loudest.

There are some concepts that I would like to explain:

- **Peak level:** Easy. This is simply the loudest sample of the song.
- **Average output level:** A value that tells how loud the song is on average.
- **Compact ratio:** Tells how compact the song is and how much each 133 ms frame differs from the average output level. The closer to 100% the more the song is playing at its average output level all the time. Can vary between 0% and 100%.
- **Maximum compress by:** Tells how much this function is allowed to compress the song. For a maximum value of 60% no single sample can be modified by more than 60%. This means that every sample will be between 100%-160% of its original value. I have limited this value to between 0% and 100% but there is theoretically nothing that says that this value can't be higher (or even less).

There is also a preview function where you can view and listen to a track, both in its original form and how it will sound after it has been changed. You can quickly switch between listening to the original or modified song which gives good testing options.

To change position in the song you simply click the position where you want it to play from and it will play from there. It's a good idea to use a pair of good quality headphones for more professional listening tests. The green drawing is the original song and the yellow is the modified.

If you have a special song, maybe one you have made yourself and want to make sure it is not over compressed you can do this:

Compress it by 100% and save it to a new file. Then compress the new file by another 100%. Now listen. You should try to listen for general changes in the volume in some parts of the song. You may have to compress by another 100% to make sure that you can hear the parts that do not sound correct. Now you know in which parts of the song an eventual error will be first noticeable.

Go back to the original song and compress by 60-70%, listen carefully to the parts where you know the error will first occur.

If you choose to always compress then Audiograbber will first compress the sound and there after change the peak level to achieve the correct average output. If you instead use compression only when needed,

Audiograbber will first change the peak level and if a peak of 100% is not enough to achieve the desired average, it will use as much compression needed to reach the desired average output. It will of course never use more compression than specified by the maximum compress meter.

Play grabbed files

This option will send a playlist (an .m3u list) with the grabbed files to your MP3 Player. The settings in your MP3 Player dictates if they are played sequentially or in a random order.

Play selected tracks

This option will make Audiograbber play the selected tracks when the ripping is finished. They will be played in a random order.

Default MP3 player

Press the browse button and select your preferred MP3 player or a wave file you want to be played when ripping is finished.

Audiograbber icon

Select this option to let Audiograbber display its own nice player icon in the main window.

Icon from MP3 player

Select this option to let Audiograbber display the icon from the MP3 player in the main window. You can click the icon here to select which icon you want to use (in case that the MP3 player has more than one icon). Left click for next icon and right click for previous.

Which icon

Left click for next icon and right click for previous icon. This works only if the MP3 player has more than one icon.

MP3 Playback

This function is found under the settings menu.

If you want to be notified when the computer has finished ripping it is a good idea to use this MP3 playback function. You can either let the computer play the tracks from the cd or launch some external MP3 program and let that program play the ripped tracks. Select here which MP3 program you want to integrate with Audiograbber. This MP3 program can then also be started by clicking on the MP3 player button on the main window. (If you right click that MP3 player button then the MP3 player will be launched with a .M3U list of the currently selected tracks.)

Audiograbber looks in the registry to find out which is your default MP3 player if you have not selected one of your own choice here.

If you rather want Audiograbber to play a short sound when finished then you simply select a wav file to be played. Audiograbber will play that file internally and not launch any MP3 player.

You can also have Audiograbber saying "Ripping complete" when it has ripped all tracks by checking that option.

Artist

The name of the artist who made the disc. If this is a compilation disc with more than one artist then type "Various" here.

Album

The album name.

Freedb Category

The disc category. These are predefined values on the freedb server and you can not make up your own categories. It has nothing to do with the categories in the ID3v1 tag for MP3 files.

Additional info

Additional info for this disc. You can for example include the year the disc was released and the musicians that performs on the disc. Not used by Audiograbber but other programs may use this info.

Clear list

This button clears the list and makes it easier to fill in the track names for a new disc. When all information is entered you can then press the "Export to tracklist" button to send the info back to Audiograbber.

Track names

Fill in the names of the tracks here. Check the spelling carefully.

Additional track info

Type additional info for the track here if there is something special to mention. This info is not used by Audiograbber but may be used by other programs.

E-mail address

If the disc submission is not accepted then an e-mail with the reason will be sent to this e-mail address.
Simply enter your e-mail address here.

Submit

Press this button to upload your disc to the database. Watch the penguins for a few seconds and then you should get a confirmation message back.

Export to tracklist

Click here to send your changes back to Audiograbber.

Back

Press this button to go back to the previous page.

Next

Press this button to move on to the next page.

Help

Press this button to open the helpfile with detailed help on how to submit a disc.

Cancel

Press this button to abort your disc submission and go back to the program.

OK

Press this button to go back to the program.

Analog sampling from cd

If your CD-ROM drive can't read digital audio or if the disc is so scratched that digital reading fails then you will have to use Analog sampling instead of digital reading (ripping). Some people like to say analog ripping but I don't like that expression. Analog sampling is a more appropriate expression.

When you sample analog you will lose quality twice: First when the CD-ROM drive converts the digital data to an analog signal for transport to the soundcard. Second when this signal is reconverted back to digital data by the soundcard. However, if your CD-ROM drive has a digital out and your soundcard has a digital in, consider yourself lucky, as this conversion becomes unnecessary. In this case quality loss should be minimal or nonexistent. Nonetheless, white noise (hiss) can be added to the track due to processing in the soundcard. With a decent soundcard though, this should be minimal.

In either case your CD-ROM drive has to be connected to the soundcard via a thin little wire that goes from the rear of the CD-ROM drive to the soundcard. Some soundcards have more than one socket to connect this wire. If this is the case make sure it is connected to the CD-ROM socket. (If this wire for example is connected to a TV input on the soundcard then normal playback from the CD will work but Audiograbber will not use the TV input and no sound will be sampled).

For best results with analog sampling try to set the input volume to a level so the peak level of the sampled file ends up at around 98%. If you have told Audiograbber to set the input level to a specific value then Audiograbber will also choose the CD-ROM drive as input source. If you tell Audiograbber not to change the input volume then the currently selected input source will not be changed either. You can manually start the mixer control to check this by double clicking on the speaker icon in the system tray (or by starting the program sndvol32.exe). When the mixer program start it shows the output settings for the soundcard instead of the input (recording) settings. You will have to go to "options", "properties" and select "recording" under "adjust volume for" to see the input settings.

Audiograbber

Line in sampling.

This function is found under the file menu.

The "line in" sampling function is used to record from audio sources through the soundcard either internally, ie. from the cd rom, or even a music file on the hard disc, or from an outside audio source such as a phonograph or radio, through the "line in" jack on the soundcard. You can plug whichever audio source you desire into your soundcard and let Audiograbber record from it. For best results, always try to send as good a signal to the soundcard as possible. To record from vinyl it is best to connect the turntable to the stereo and then the stereo to the sound card. Do not connect the turntable directly to the soundcard, as the signal from the turntable is too weak.

The "line in" function has three different recording modes: "**manual**", "**track splitting**", and "**time scheduled**". These different modes are described in more detail later on in text.

To name the tracks is easy. Just enter all the names in the fields in the upper right hand corner. The current track is the one marked with an arrow. It is possible to change current track by clicking on another number. The arrow will move to verify this change.

The "volume meter" (middle of the window) displays the recording level of the incoming sound and is always active even if Audiograbber is not recording. This allows for easier adjustment prior to actual recording. When there is no incoming sound the meter should stay under 0.10% (Good soundcards should however be able to reach as low values at 0.00% - 0.02%). To achieve the optimum recording level adjust the "mixer" so the meter occasionally peaks at 100%, but not too often.

You can click on the "**Mixer**" button to start Window's internal soundcard mixer. All the audio sources that are not going to be used, ie. "mic, midi", "wav", etc... should be deselected and their respective slide bars dropped to the lowest value to ensure minimal undesired background noise. You must of course also select the audio source you want to record from (usually Line in or AUX) and set an appropriate volume level.

Right above the status bar there is a field which displays which recording settings you are using. These refer to the settings you have selected in [Normalize settings](#) and [MP3 settings](#). You need to go out of the "line in sampling" window and back to the main window to change these settings. You can "shortcut" to the main window however, by "right clicking" on this field and selecting "enable main form" from the pop up menu. Make your changes and then right click on the same field again to close. Be careful however, as it isn't too hard to crash the program by misuse of this shortcut).

At the bottom there is the status bar which tells you what is going on and optionally what you are supposed to do.

A button will be visible right below the "Close" button if there are any files waiting to be normalized and/or encoded. Press the button to initiate this process. This button will not display when the program is recording.

It is also a good idea to click on the help menu, choose "Help for this window" and thereafter click on whatever you want to learn more about.

If you need more professional editing options of your wav files, like click removal, noise reduction and manual fading options etc, then [Cool Edit](#) is a very good program.

Manual recording mode, do not split tracks:

This is the most basic recording mode. You will have to press the "Start" and "Stop" button for each track you want to record to a separate file. It is wise with this method of recording to limit recording to a certain amount of minutes using "the minutes to stop after" option in case you forget to come back and press the stop button!

Manual recording mode, auto split tracks:

Enabling this useful function when recording two or more tracks from vinyl (LP's) will allow Audiograbber to detect the silence between tracks, and create separate destination files for each one. Thus saving you the bother of doing it manually. Just play the record straight through in one go and let Audiograbber do the rest! It is a complicated process for Audiograbber to differentiate between record noise during silence and an actual quiet passage of music but it does its best! Use the "track split sensitivity" slide bar to change Audiograbber's perception of silence. Slide the bar to a lower value if Audiograbber clips the songs during the final fade and to a higher value if the program fails to separate tracks.

Time scheduled recording:

This mode is useful for recording from radio when you are away from your PC. It works like a VCR timer, but for radio (or for that matter whatever audio source is connected to your soundcard).

Times can be set flexible. When the program is in "time scheduled" mode it will display the current date and time below the "Close" button. How this is shown is dependent of your computers regional settings (in the Control Panel). Use this same format when entering information in the "Date" and "Time" field (see below).

"Date" field:

Enter a date here (using the proper format) to record only on that specific day. You can leave the field empty if you want, whereby Audiograbber will then record at the same specified time every day. It is also possible to specify a weekday like Monday or Tuesday etc, (in English) whereby Audiograbber will record once a week (on that day) during the specified time.

"Time" field:

Specify the start time (time of the day) for the recording in this field. Remember to use the proper format.

"Length" field:

Specify the duration of the recording here using the following format: hh:mm:ss where "hh" are hours, "mm" are minutes, and "ss" are seconds (for example, One hour two minutes and three seconds should be written as following: 01:02:03).

Tracknames, Times

Tracknames and times are saved in plain textfiles with the extension .Nam for tracknames and .Tim for times. The filename for auto loaded tracknames is "Auto.Nam" and for auto loaded times "Auto.Tim".

Record

Press this button to start recording. In "track splitting" mode Audiograbber will wait until sound is coming in before starting to record. In "time scheduled" mode this button may be disabled when the program is either waiting for a time to be reached or is already recording.

Stop

Press this button to end the recording. The sound that has already been recorded is saved. If there are files waiting to be normalized and/or encoded, a button will appear in the lower right hand corner. Press that button to proceed with normalizing and/or encoding or press the "record" button to record more files first.

Abort

Press this button to abort the recording. The track that has been recorded is not saved. If there are files waiting to be normalized and/or encoded a button will appear in the lower right hand corner. Press that button to proceed with normalizing and/or encoding or press the "record" button to record more files first.

Recording mode

Choose between "manual recording" or "time scheduled" mode. "Manual recording" requires you to press the "start" and "stop" button yourself. "Time scheduled" means that Audiograbber will record at predefined times.

Do not split tracks

Choose this option if you want to split the tracks from an LP manually. You will have to press the "start" and "stop" button yourself for each track in order to save each one as a separate file.

Auto stop after

This option is useful if you want to be sure that not too much data is recorded and written to the harddisk. Audiograbber will stop recording after the specified number of minutes if this option is selected. You can of course press the "stop" button to stop earlier.

Minutes to stop after

Select how many minutes (maximum) shall be recorded.

Auto split tracks

Choose this option if you are recording two or more track from an LP and want Audiograbber to create a separate destination file for each individual copied track, thereby sparing you the hassle of having to do it manually.

Track split sensitivity

Audiograbber splits tracks by detecting silence. This slide bar affects Audiograbber's perception of silence. If the bar is set too high there is a risk that tracks will be cut off during the fade. If it is set too low there is a risk that the tracks will not be split at all.

All tracks are at least 35 seconds

Choose this option to overcome the problems of tracks being split due to silence after intros.

Date

This field holds the date the recording shall start. The date format is dependent upon your computers regional settings. You can see in the lower right hand corner how dates are expressed on your PC. This field can also be left blank or you can specify a weekday (Monday, Tuesday, Wednesday, Thursday, Friday, Saturday or Sunday).

Time

This field holds the time the recording shall start. The time format is dependent upon your computers regional settings. You can see in the lower right hand corner how times are expressed on your PC.

Length

This field defines long time Audiograbber shall record. Use the following format: hh:mm:ss (hours, minutes and seconds).

Time scheduled recording

You can specify up to 24 different recording times. All recordings are faded in and out for 3 seconds.

Append date and time to filename

If you have many files recorded at different times it will sooner or later be difficult to tell which file was recorded when. Select this option to include the date and time the recording started in the filename.

Artist

In this field you can write the artist or band that has created the track. If you record from radio you can type "Radio" instead.

Album

In this field you can write the album name or leave it blank.

Track number / Current position

The arrow in this column tells which tracks are presently being recorded or will be recorded next. You can easily change by clicking on one of the track numbers.

Track name

Enter the track names in this field. If you are recording from radio it is a good idea to type the name of the radio channel or another identifying word or number. If you leave these fields empty the default name "track x" (x being the current track number) will be assigned.

Copy status

In this field Audiograbber will fill in the recording time for that specific track. It will also tell how many speed problems occurred (if any). If no speed problems occurred it will show "Copied OK".

Clear

Clears the copy status column.

Used CPU time

Displays if the computer is fast enough to handle the incoming data. If you get speed problems there will be dropouts in the files. The CPU speed should not be a problem if you record to a wav format but it can be problems if you record directly to MP3 or WMA format. In that case use an intermediate wav format/file instead.

Volume meter

Displays the level of the incoming sound. Press the mixer button to adjust this level. When there is no incoming sound the meter should stay under 0.10%. Set the mixer so the volume occasionally peaks at 100%, but not too often.

Mixer

Press this button to launch Window's internal sound card mixer. The mixer should start in the recording mode. Deselect all the sources you are not recording from and drag their respective volume meters to "0" to get minimal undesired background noise.

Close

Press this button to exit from "Line in" sampling and return to the main part of the program.

Normalize/Encode now

This button is shown when there are files waiting to be normalized and/or encoded and there is no recording in progress. In "track split" and "time scheduled" modes it will pop up automatically and start a countdown from 10 before normalizing and/or encoding starts. If you have previously pressed the "stop" or "abort" button you will have to resume the normalizing/encoding process manually by pressing this button.

Date and time

In "time scheduled" mode you will see the current date and time here. Use this same format to enter the dates and times in the "time schedule" field.

Name of the song

Displays the name of the current song and recording time.

Recording to:

Displays in what order things are going to happen and what settings you are using.

Status bar

Displays what is presently occurring and periodically what you are supposed to do.

Recording in progress

This indicator button blinks when recording is in progress.

Auto load/save tracknames

Select this option to load and save the track names automatically when Audiograbber starts and exits. The file name used for holding the file names is "Auto.Nam".

Auto load/save times

Select this option to load and save the scheduled times automatically when Audiograbber starts and exits. The file name used for holding the times is "Auto.Tim".

Start Audiograbber in time scheduled mode

Select this option to make Audiograbber start in "time scheduled" mode. This is useful if used in conjunction with the option "Start Audiograbber when the computer starts".

Start Audiograbber when the computer starts

If this option is selected then Audiograbber will start when the computer starts. Useful if you have some electronic timer to start up your computer. This is the only instance where Audiograbber uses the registry to store information.

Shut down when finished

Select this option if you want Audiograbber to shut down the computer when recording is finished. It works in "track split" and "time scheduled" mode. If there is another time scheduled in less than 15 minutes then the computer will not be shut down even if this option is selected.

Audiograbber starts when the PC starts

If Audiograbber starts when the PC starts then that is because you have told it to under File, Line in sampling. Go there and uncheck the box "Start Audiograbber the next time the computer starts".

Delete wav file after encoding

Use this if you do not want to keep the wav files. The wav files will be deleted one by one as soon as they have been encoded and harddisk space will be restored.

Normalize all files as if they were one file

Selecting this option lets you normalize all files together rather than individually. The difference is that the peak value is not found for each file, instead only one peak value is found for all files together. This option is only available for plain (simple) normalizing.

Different stereo modes

A common question is what the different stereo modes is used for in MP3 files.

- **Joint Stereo:** Joint Stereo shares certain bits between high frequency left and right channels. This improves compression efficiency at a slight loss of stereo separation. Lower frequencies are treated as normal stereo. Use Joint Stereo to obtain the best overall quality at mid-to-lower bit rates. < 224 Kbit.
- **Stereo:** Stereo includes two independent channels. The total bit rate remains constant, but the split between the channels can vary. The Encoder uses this flexibility to improve quality by allocating more bits to the channel with the more dynamic signal. Use the Stereo setting for best quality stereo audio at higher bit rates.
- **Dual Stereo:** Dual Stereo includes two completely independent channels (left/right), each with half the total bit rate. In effect, it is two mono files packed into a single file. Dual is generally used for multi-lingual audio programs.

The higher bitrate the larger file and better sound quality. 128 Kbit is MP3 standard and 64 Kbit is WMA standard. 128 Kbit is the same as 16000 bytes ($128000 / 8$). That is 16 Kbytes per second or just a little bit less than one MB per minute of music. That is equal to a compression ratio of 1:11.025. 128 Kbit is usually referred to as cd quality. That might be true for most songs but some songs needs higher bitrates to sound perfect. Use headphones to listen for artifacts in the MP3 or WMA files.

Choose Language

This function is found under the file menu.

Audiograbber uses English as default language but from version 1.61 it can also run in other languages. When selecting "Choose language" from the file menu another menu with the available languages will show up. Just click the language you like and it changes instantly.

The language files are simple textfiles ending in .lng and the ones found in Audiograbbers program directory are shown as available. (Maximum 10 language files can be shown). The setup program for Audiograbber will automatically install the most common languages but other language files, and improved versions of the pre-distributed, can be found on Audiograbbers site at <http://www.audiograbber.com-us.net/download.html>

It is easy to make your own language file, instructions and a template file can be found at the above web site.

Extra settings in audiograbber.ini

Some rarely used features did not fit into the GUI (Graphical User Interface) of Audiograbber but they can still be modified by manually changing audiograbber.ini.

MP3VBRMaxBitRate=

MP3VBRMinBitRate=

Sets the max and/or min bitrate for LAME's mp3 encoder dll. If you for example have a hardware MP3 player that can only play back MP3 frames of 256 Kbit/s or lower then set the max bit rate to 256 and no frames in the MP3 file will get higher than 256 Kbit/s. Default values are 0 which means no limits. Possible values are: 32, 40, 48, 56, 64, 80, 96, 112, 128, 160, 192, 224, 256 and 320.

SendStartUnit=False

Change this to True and Audiograbber will send a "Drive get ready" ASPI command before it rips each track and before it asks the drive for the tracklist. A few problematic drives seems to gain if that command is sent.

If your proxy server needs a username and a password you will have to enter that manually in the audiograbber.ini file. Simply type these two line in audiograbber.ini:

ProxyUsername=jackie

ProxyPassword=hacker

and replace jackie hacker with your own username and password.

LoadASPI=True

Change this to False and Audiograbber will not load the ASPI manager. If you have problems to start Audiograbber then it is often the ASPI manager that fails and with this code line the program shall at least start.

LoadNTCalls=True

Change this to False and native NT calls (for CD-ROM drive access) will not be loaded when Audiograbber starts. This option is only valid under Windows NT and 2000. If your PC gets a Blue Screen Of Death (BSOD) when it starts then it is probably these NT calls that fails. Setting this option to False will at least make it possible to start Audiograbber. You need to manually install an ASPI manger if you want to rip digitally under NT/2000 with this option set.

FadeTime=

Set to 1, 2 or 3 (seconds). Default value is 2. This changes the fade in and out time for partial ripping and tracks that are ripped with the "Fade tracks" checkbox checked.

LogarithmicFade=False

Change to true if you want the fade type for partial rips to be logarithmic.

AutoRip=False

Change to True if you want Audiograbber to start ripping automatically as soon as a new disc is inserted. It will first connect to freedb and get disc info if that option is also selected.

MinimizeToSystemTray=False

Change to True if you want Audiograbber to minimize to the system tray rather than the normal task bar when it is minimized. Unfortunately the task bar can only show icons with 16 colors and Audiograbbers little 16x16 pixels 16 colors icon does not look good!

Year

The year the song was released. ID3v1 uses 4 characters from this field even though it is possible to enter more characters. ID3v2 uses all characters entered in this field.

Fade tracks

Tick this box if you want the tracks to be faded at start and end. Useful for live recordings.

The fade time is normally 2 seconds and is a logarithmic fade type. The fade time and type can be changed by modifying these lines in audiograbber.ini:

FadeTime=1 (It is possible to use 1, 2 or 3 seconds).

LogarithmicFade=False

ID3v2

Click these buttons to modify the ID3v2 settings for each track. Buttons will get red text with a checkmark if individual ID3v2 settings has been applied to the track.

Cut

Press this button to split the current recording (stopping the current recording and immediately start a new one). The radio buttons to the right defines the cut type.

Abrupt cut

When the cut button is pressed the cut is an instant cut and a new track will be recorded immediately with this option selected. The start of the second file will get a seamless fit to the end of the previous one.

Faded cut

This fades the end of the current recording slightly before it cuts and it also fades in the start of next track. In track splitting mode it will wait for incoming sound before it starts recording next track. Good to use for live recordings when you do not want the ends of the songs to be too abrupt.

Start now

This button is only visible when Audiograbber waits for sound to come in in track splitting mode. Press the button if you do not want Audiograbber to wait any more.

Reduce noise on incoming sound

This setting is only available if you have installed the noise reduction dll AlgoCtrl.dll from Algorithmix. The noise is reduced on the fly so it takes a little more CPU power with this setting on.

Noise reduction settings

Press this button to adjust noise reduction settings. This is only available if you have installed a third party noise reduction dll from Algorithmix (AlgoCtrl.dll).

Path to local freedb database

It is possible to download the whole disc database from freedb and have Audiograbber look up the disc locally on the computer rather than going online to get the tracknames. Here is where you enter the path to the local freedb if you have installed the database on your PC. If the disc is found in the local database Audiograbber will find the tracknames automatically and you will not have to press the penguin button.

Browse for freedb path

Press this button to tell Audiograbber where your local freedb database is found (regular users does usually not use any local database). Browse for the path and have a blank line on top of the directory list window if you want to stop using a local database.

Settings tabs

There are three ways to go to this window and the tabs are enabled depending on how you got here. If you came from the tracklist then all tabs are enabled. If you came here from line in sampling then the track properties tab is not enabled since there are no track properties like this in line in sampling. If you came here from the ID3v2 settings menu then only the ID3v2 settings tab is enabled.

ID3v2 genre

You can either use one of the 147 ID3v1 genres or manually enter an ID3v2 genre of your own choice. ID3v1 genres must be one of the predefined texts but with ID3v2 it is possible to make up your own genre.

ID3v2 year

This is the year or date the disc was released. You can either simply use the ID3v2 year or enter an individual year for the track here. ID3v1 and ID3v2.3 years must consist of four characters (like 2001) but ID3v2.4 can optionally hold a full date and shall in that case be in this form: yyyy-MM-ddTHH:mm:ss. Use as much of it as you desire, ie if you only want the year, month and date then enter something like 2001-03-24.

ID3v2 comment

While ID3v1 has only one single comment field with a fixed length of 30 characters ID3v2 can have many different comment fields of any length. Either you can use the comment from ID3v1 or make your own comment for this track here.

Track properties

Set various properties for this track here.

ID3v2 properties

Set various ID3v2 properties for this track here.

ID3v2 settings

ID3v2 is info about the song that is stored first in the MP3 file. Make your general ID3v2 settings here.

Enlarge

Press this button to make the lyrics field larger or to go back to normal size.

Load lyrics

Press this button to load the lyrics from a text file.

Save lyrics

Press this button to save the lyrics to a text file.

Save to database

Press this button to save the disc info in discs.txt and cdplayer.ini (if you have chosen to use cdplayer.ini under general settings). If this is for a track from line in sampling then you will be asked for a file name to save the info to because LP's etc has of course no disc ID that can be used to look up the disc again.

Lyrics field

Type the lyrics as plain text here. You can press the "Enlarge" button to make this field larger.

Play button

Play and stop playing the track with this button. You can also press "F5" to start playing the track and "F8" to stop playing. That is useful so you don't have to grab the mouse when you try to keep up your typing with the singer.

Play progress slider

Drag this slider to the position in the track you want it to play from.

Use ID3v2

Check this box to make Audiograbber create ID3v2 tags for the tracks. The ID3v2 tag is always first in the MP3 file in contrast to the ID3v1 tag which is last in the file. There are a lot of various frames that can be used with ID3v2 and Audiograbber uses only the most useful :)

ID3 version 2.3 and version 2.4

There is only one slight difference between the ID3v2 frames Audiograbber creates for ID3v2.3 and ID3v2.4; ID3v2.3 uses a frame named TYER but this frame was replaced with a TDRL frame in ID3v2.4. Both these frames holds the year/date of the recording. ID3v2.3 can only hold a 4 bytes date as in 2001. ID3v2.4 can optionally hold a date of this format:

yyyy-MM-ddTHH:mm:ss. Use as much of it as you desire, ie if you only want the year, month and date then enter something like 2001-03-24.

So, ID3v2.4 is just a little more flexible than ver 2.3 but it is possible to rather use ver 2.3 because some MP3 players like Winamp 2.72 can only show 2.3 tags and not at all show version 2.4 tags.

Pad ID3v2 tag

There is no fixed length of the ID3v2 tag and since it is stored first in the file it would be bad if it was completely filled up if you later on want to correct a spelling or add something. (Audiograbber can not modify ID3 tags but there are other programs that can). If the tag is completely filled up (padded with 0 bytes) then the other program has to flush the whole file if it wants to update the tag. Therefore it is possible to pad (fill out) the tags with some bytes.

Software used

Check this box if you want to have the software used to create the MP3 stored in the ID3v2 tag.

User defined

Enter any string here that describes the software you have used to make your MP3's.

Auto detect

Audiograbber autodetects which Audiograbber version and encoder and what settings the encoder used when the MP3 was made.

Add to comment field

ID3v2 can have many comment fields of any length. Check the additional info you want to add to the comments fields in the ID3v2 tag.

Track number

Tells how the track number shall be stored in the ID3v2 tag. Track numbers are only used for cd tracks. Audiograbber does not create ID3v2 track number fields for line in samplings.

Encoded by

Check this box and enter your nickname or whatever info you want to be stored in this field in the ID3v2 tag.

Music CD Identifier

This field is named "MCDI" in the ID3v2 specifications and it holds the table of contents (TOC) of the cd. It is normally not used but it could be useful in the future though since it would be possible to look up the cd the track originally came from with this info. Someone would have to write one such program first though :)

This field is of course not used for line in samplings.

Tracklist color

This is the color of the tracks in the main window. The values corresponds to what you have set in the control panel, display properties, appearance.

Window text is the font color of standard window text.

Highlight is the background color of a selected item in the tracklist.

Active caption is the color of the title bar of an active window.

Inactive caption is the color of the title bar in a not active window.

Store disc in local freedb database

Press this button to store the disc in your local freedb database.

ID3v2 properties

This function is reached by double clicking or right clicking a track in the track list. It is also reached via the ID3v2 buttons on line in sampling.

ID3 is a standard to add text info to an MP3 file and there are two different versions of ID3; version 1 and 2. [ID3v1](#) (version 1) is the oldest standard and it simply appends 128 bytes to the end of the MP3 file. The drawback with this old standard is that it has fixed length of the fields so track names etc can only be up to 30 characters. Another drawback is that it is not so useful for streamed MP3 files since the tag is at the end of the file but at least almost all MP3 players can read ID3v1 tags.

ID3v2 is a newer and much more advanced standard which is now being more and more used. ID3v2 has not fixed field lengths so track names etc hasn't to be truncated if they are long. ID3v2 has also support for much more fields than ID3v1 so it is possible to add lyrics etc to an ID3v2 tag. It is also possible to add other info than plain text, pictures for example even though that is not possible with Audiograbber. Another advantage is that the ID3v2 tag is first in the MP3 file so MP3 players capable of displaying ID3v2 tags can show the info directly even with streamed MP3's. Not all MP3 players can display ID3v2 info yet though. It is however possible to use both ID3v1 and v2.

ID3v1 is in Audiograbber set per disc so it is for example not possible to have different comments, year and genre for the tracks from a cd. ID3v2 on the other hand is set on per track basis so all tracks can have their own comment etc. It is of course tedious to type the same info for each track so it is possible to simply set the year, genre and comment to the same value as the one used for ID3v1. A shortcoming in Audiograbber is that if you for example want to use the same genre for all tracks and the genre is not one of the predefined 147 [ID3v1](#) genres you will have to enter the genre manually on all tracks even though it is the same. But the 147 predefined ID3v1 genres should cover most songs anyhow.

The field to enter lyrics is not so large so a button to enlarge the field is found on the ID3v2 settings window. There is also a play button and a progress slider which can be used to play and change position in the song so you can play it and type the lyrics you hear when they sing. You can also press "F5" to start playing from the position in the progress slider and "F8" to stop playing. It is useful to use these keys instead of grabbing the mouse when you are not quick enough to keep up with the singer.

This button and slider is of course not shown when you are setting ID3v2 properties for a track from an analog source. (Audiograbber can not control your turntable!).

Tips:

- Many bands has the lyrics for their songs on their official homepage and you may be able to do some cut/copy and pasting in order to not have to type it all yourself. (Ctrl+C is copy and Ctrl+V is paste).
- You can also find lyrics on <http://www.lyrics.com> and <http://www.lyrics.ch/index.htm>

Lyrics can be loaded and saved to/from plain text files with the buttons. There is also a button to save the disc info in the database (discs.txt). All disc info is saved when you press this button so it doesn't just save the info for this track. Lyrics are not stored in discs.txt in order to not make that file too large, instead lyrics are saved into separate files for each disc. These files are saved in a subdirectory named lyrics so your Audiograbber directory is not cluttered with too many files. The name of a lyrics file is the freedb disc id with .lyr as extension. (If the disc ID is 01234567 then the lyrics file will be named 01234567.lyr). Since there is no disc id for line in samplings the lyrics are here saved in the .nam file used to save line in samplings info.

You can use the program [Helium](#) or [Dr. Tag](#) to view and edit both ID3v1 and v2 tags. The author of Helium, Mikael Stalvik, has also made a good ID3v2 plugin (Argon) for Winamp and you find that on Helium's homepage.

Tracks with individual ID3v2 info gets a yellow checkbox in the main window.

More info in ID3v2 is found on <http://www.id3.org>

More on ID3v2 on [ID3v2 settings](#).

ID3v2 settings

This function is reached from the settings menu.

ID3 version 2 is a quite new standard that adds text and other info about a song first in an MP3 file. Audiograbber can use either the **ID3v2.3** or **ID3v2.4** standard when making ID3v2 tags. There are some differences between ID3v2.3 and v2.3 but for the ID3v2 frames that Audiograbber supports there is only one minor difference. ID3v2.3 uses a frame named TYER but this frame was replaced with a TDRL frame in ID3v2.4. Both these frames holds the year/date of the recording. ID3v2.3 can only hold a 4 bytes date as in 2001. ID3v2.4 can optionally hold a date of this format:

yyyy-MM-ddTHH:mm:ss. Use as much of it as you desire, ie if you only want the year, month and date then enter something like 2001-03-24.

So, ID3v2.4 is just a little more flexible than ver 2.3 but it is possible to rather use ver 2.3 because some MP3 players like Winamp 2.72 can only show 2.3 tags and not at all show version 2.4 tags.

Padding means to fill out the ID3v2 tag with zeroed bytes. Since the ID3v2 tag is first in the MP3 file it would be extra complicated for ID3v2 editor programs like [Helium](#) to add some extra info to the tag. (Not too complicated, it is just that file has to be flushed if it gets larger). If the ID3v2 tag has been padded with 100 bytes that means that an ID3v2 editor program can easily add 100 bytes to the tag without flushing the file. 100 is the default value here.

Software used is a frame with info on what software that was used to make the MP3 file. Either you can enter your own text string here or simply let Audiograbber auto detect what settings that was used.

In ID3v1 it is only possible to one single **comment field** but in ID3v2 there can be as many comments as you want. Audiograbber only lets you add one comment to the tag but it can also create sub comments with the freedb disc id, the cdplayer.ini disc id and/or the checksum of the ripped file.

The **track number** can also be stored in the ID3v2 tag and you can select how you want the track number to appear in the tag. Audiograbber does not store track numbers in the ID3v2 tag for line in samplings though.

Encoded by is a frame that lets you enter who you are. If you want you can enter your nickname, company name or whatever here.

Music CD identifier is the table of contents (TOC) from the cd. The frame consists of a binary dump of the TOC from the CD, which is a header of 4 bytes and then 8 bytes/track on the CD plus 8 bytes for the 'lead out' making a maximum of 804 bytes. This info can for example be used by other programs to calculate a freedb disc id and thus be able to locate from which cd the track originally came from. There are however no programs yet that uses this info but it may be useful in the future, who knows? This field is of course not used for line in samplings.

The following ID3v2 frames can be created by Audiograbber:

- TIT2 (Track name).
- TALB (Album name).
- TPE1 (Artist name).
- COMM with subset comment (Comment for the track).
- COMM with subset CDDb Disc ID (The freedb disc id).
- COMM with subset CDPlayer.ini Disc ID (The cdplayer.ini disc id).
- COMM with subset Checksum (Audiograbbers checksum of the ripped file).
- TCON (Genre).
- TYER (v2.3), TDRL(v2.4) (Date).
- TRCK (Track number).
- TSSE (Software used).
- TENC (Encoded by).

- USLT with subset Lyrics (Song lyrics).
- MCDI (Table of contents from the cd).

If you want to edit your ID3 tags and add pictures etc I suggest you use the program [Helium](#) or [Dr. Tag](#).

More info in ID3v2 is found on <http://www.id3.org>

Info on [ID3v2 properties](#).

Capitalize tracknames

This function is found under the cd menu.

Quite simple. Press this function to spell the first character in each word in the tracknames as uppercase and the rest as lowercase. "Dancing with MySelf" will for example be corrected to "Dancing With Myself".

Keyboard shortcut for this function is Ctrl+K.

