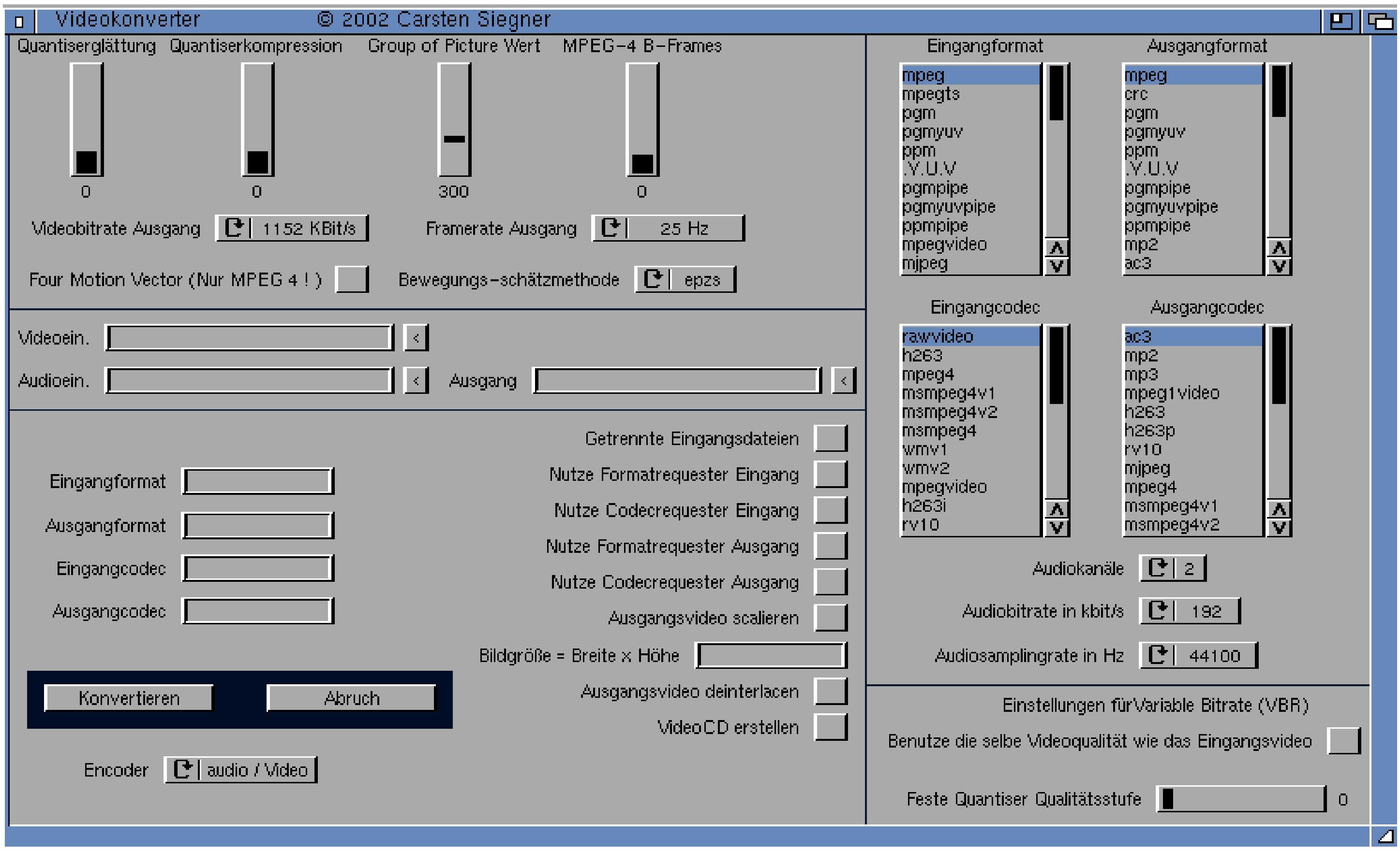
# Videokonverter




Videokonverter

# Features

- Es werdem alle Features von ffmpeg unterstützt
- Auf jedem System lauffähig
- Decoder, Encoder, Streammultiplexer und VideoCD-authoring in Einem
- Plugin-Schnittstelle (VideoCD-authoring)
- Multitaskingfähig ( Beliebig viele ffmpeg-Tasks )
- Auslesen von unverschlüsselten DVD-Videos und konvertieren in andere Formate ( benötigt Filesystem AllegroCDFS und ein DVD-Laufwerk auf dem Rechner)
- Nutzung fester Videobitrate oder Variable Videobitrate (VBR)
- Einfache Nachbearbeitungsmöglichkeit im Programm (deinterlace, Bildglättung)
- Erstellung von Multitrack-VideoCd’s ( Bis zu 5 Tracks)
- Abspeicherbare Voreinstellungen
- Ein und ausschaltbare online Hilfe
- Appwindow und Appmenü Gegebenheiten

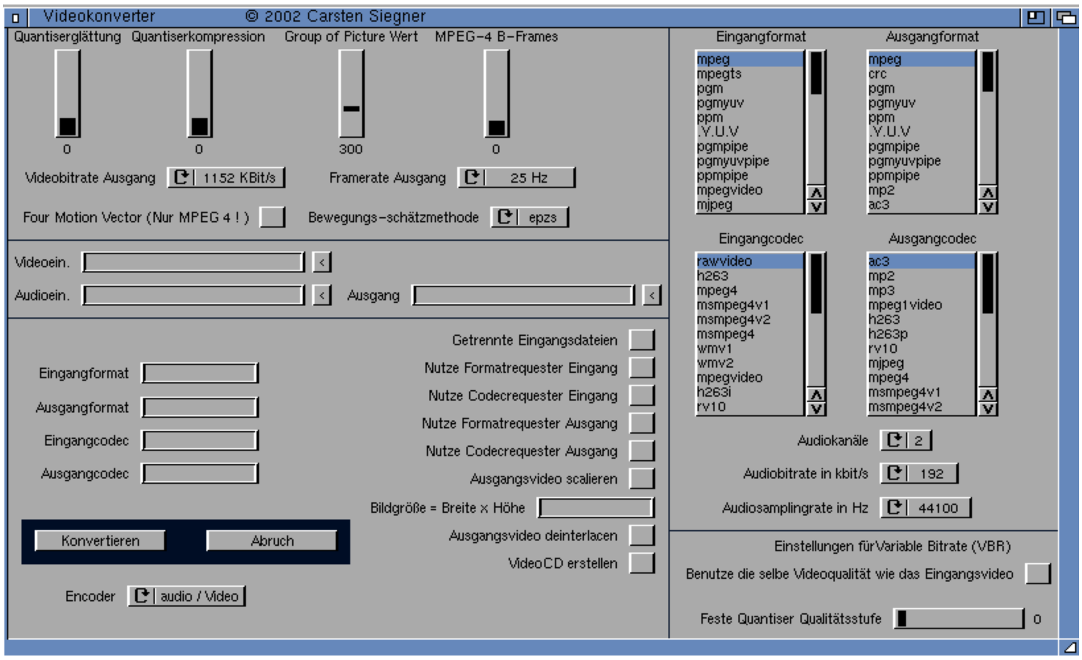
# Es werden zum Programmstart benötigt

68020 - 68040
Amithlon
WarpOs, PowerUp, MorphOs
Ixemul.library (Aminet)
nicht unter 32MByt Ram
ffmpeg (liegt bei)
vcdimager (liegt bei)

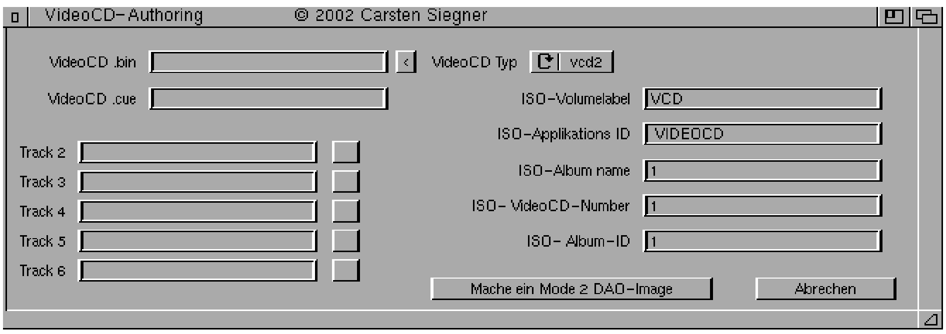
# Installation

Endpacke das Lzh-Archiev irgendwo hin, und es läuft :)

# Funktionen




**Quantiserglättung**: Zusätzliche Möglichkeit zum nachbearbeiten der Videobilder. **Quantiserkompression**: Einstellbarer Wert für die zusätzliche verlustbehaftete kompression bei dem Encodiervorgang. **Group of Picture Wert:** Anzahl der I-Frames pro 10-Sekundentakt und pro Framerate. **MPEG-4 B-Frames:** Anzahl der B-Frames pro Framerate in einem Mpeg 4Video. **Four Motion Vector:** Dies ist eine Berechnungsmethode in Mpeg 4 Videos. **Bewegungs-Schätzmethode**: Methodeo zur Encodierberechnung. Die Methode beeinflußt die Rechenzeit ganz enorm. **Bildgröße**: Damit kann man die Bildgröße der Ausgangsstream bestimmen. Man muß nur „800x600“, oder Anders eingeben. Außerdem kann man Schlüsselworte eingeben. Für VideoCD Erstellung gild : vcdpal, vcdntsc,svcdpal und svcdntsc. Es gelten außerdem Bildgrößenabkürzungen: Sqcif, qcif, cif, 4cif. **Benutze die selbe Videoqualität wie das Eingangsvideo:** Hiermit kann man ffmpeg veranlassen, den Quantiserfaktor vom Decoder beim encoden zu benutzen. Dadurch erhält man nahezu verlustfreie Ergebnisse. Als Nebenefekt hiervon hatt die Einstellung der Videobitrate keine Auswirkung auf das
Endergebniss, da der Encoder die Videobitrate zu jedem Frame neu bestimmt.
**Feste Quantiser Qualitätsstufe:** Einstellbare Größe des Quantiserfaktors. Mit diesem Faktor kann man bestimmen, ob noch eine zusätzliche verlustbehaftete Kompression beim encoden ausgeführt wird. 0 bedeutet,
das diese Methode abgeschaltet ist. 1 bedeutet, das Verlustfreies encodieren möglich ist (sehr gute Bildqualität,
aber große Dateien). 31 bedeutet das stark zusätzlich komprimiert wird (schlechte Bildqualität, aber kleine Dateien).




**VideoCD.bin:** Dort kann man mit dem ASL-Requester ein Ausgabedatei anwählen. Diese Datei darf dabei keine Endung besitzen ! Die Endungen .bin und .cue werden automatisch vom Programm angehängt. **Track 2 - 6:** Hier werden das Ergebniss vom Videokonverter eingelesen, wenn man die endsprechende Checkbox betätigt. Es können nur die Dateien eingelesen werden, die in Ausgangsrequester von den Videokonverter stehen. Die Videotracks fangen deshalb erst bei 2 an, weil der erste track, der ISO9660-track mit dem Verzeichniss ist.**ISO Volumelable:** Hier kann man den Nahmen der VideoCD eingeben (vorgegeben ist der nahme VCD). **ISO-Applications ID:** Hier kommt das Programm rein, von dem die VideoCD hergestellt wurde (Vorgegeben ist der Nahme VideoCD).

## MPEG1 Requirements

| | VCD 1.1 | VCD2.0 | CD-i | MediaMogul |
|---|---|---|---|---|
| **Video:** | | | | |
| **Picture Size:** | | | | |
| NTSC | 352 x 240/29.97Hz ,30Hz (or 23.976Hz film)[1] | | | |
| PAL | 352 x 288/25Hz (or 24Hz film)[1] | | | |
| Stills | MPEG stills should use the same dimensions as motion video.[2] | | | |
| **Pixel Clock :** | | | | |
| Frequency | 13.5Mhz | | 15Mhz | |
| Horiz. Pixel Distortion Ratio | 1.09[3] | | NA[3] | |
| **Bitrates:** | | | | |
| **Maximum** Video bitrate[4] | 1151929 bits/sec | | 1183200 bits/sec | |
| **Maximum** MPEG1 bitrate[5] (multiplexed audio/video) | 1411200 bits/sec | | | |
| **VBV Buffer Size:** | | | | |
| Video stream[2] | 40960 bytes | | | |
| Still Picture stream[2] | 47104 bytes | | | |
| **PCB Channel :** | **Must** be in Channel 1 | | Normally in Channel 0. | |
| **Streams :** | | | | |
| Number of Streams allowed | One. | | Up to 16. | One. |
| Stream Number | **Must** be in Stream 0. | | Typically in Stream 0. | **Must** be in Stream 0. |
| | | | | |
| | **VCD 1.1** | **VCD2.0** | **CD-i** | **MediaMogul** |
| **Audio:** | | | | |
| ALL | 16 bit, 44.1K sampling rate | | | |
| Joint Stereo/Stereo/Dual Channel | 224kbit/sec | ONLY these bitrates: 128, 192, 224, or 384kbit/sec[6] | 32-448kbits/sec/PCB Chan. Most commonly 192kbit/sec/ PCB Chan. | |
| Mono: | No. | ONLY these bitrates: 64 or 96 or 192kbit/sec[6] | 32 to 448kbits/sec[7] | |
| **Streams :** | | | | |
| Number of Streams allowed: | One | | 1-32 | 1-16 |
| Stream Number | Stream 0 . | | Streams 0 to 31, starting with Stream 0. | Streams 0 to 15, starting with Stream 0. |

# Anhang : VideoCD II

[1] *Dimensions up to 384 x 288 are supported, though not at full frame rate , but may not display properly on normal TV equipment. CD−i allows for dimensions below 352 x (240 or288), as long as both x and y are multiples of 16. Smaller images are automatically centered on the screen by the decoder card.*

[2] *MediaMogul does not support MPEG stills. However, any I−frame of a motion stream may be displayed as a still image. The maximum size of any one still picture(or frame) is the same as the VBV buffer size.*

[3] *VCD MPEG1 fills the screen, but dimensions are distorted slightly. CD−i MPEG1 is not distorted, but doesn't fill the screen fully−−at 352 pixels horizontally, there will be black vertical bands on both sides.*

[4] *Maximum video rate based on these assumptions:*

- One audio stream at standard bit rate: CD−i: 192kbit/sec, VCD: 224kbit/sec.
- MPEG play is from CD. Hard drives may support up to 5Mbit/sec video streams.

[5] *Maximum mux rate based on CD play. Depending on the decoder, playing MPEG1 from a hard drive **might** work up to 5.6448Mbit/sec.*

[6] *For VCDAV (Trackable MPEG), only224kbit sec is allowed. VCDAV also has these requirements:*

- Only motion video allowed, no stills.
- Must contain both motion video and audio.
- Only stereo audio, no mono.

[7] *1kbit=1000 bits*

# Anhang: Formate und Codecs

File formats:
Encoding: mpeg crc pgm pgmyuv ppm .Y.U.V pgmpipe pgmyuvpipe ppmpipe mp2 ac3 h263 mpeg1video mjpeg s16le s16be u16le u16be s8 u8 mulaw alaw rawvideo rm asf avi wav swf au gif mpjpeg singlejpeg jpeg ffm
Decoding: mpeg mpegts pgm pgmyuv ppm .Y.U.V pgmpipe pgmyuvpipe ppmpipe mp3 ac3 mpegvideo mjpeg s16le s16be u16le u16be s8 u8 mulaw alaw rawvideo rm asf avi wav swf au mov jpeg ffm
Codecs:
Encoders: ac3 mp2 mp3 mpeg1video h263 h263p rv10 mjpeg mpeg4 msmpeg4v1 msmpeg4v2 msmpeg4 wmv1 wmv2 rawvideo pcm_s16le pcm_s16be pcm_u16le pcm_u16be pcm_s8 pcm_u8 pcm_alaw pcm_mulaw
Decoders: rawvideo h263 mpeg4 msmpeg4v1 msmpeg4v2 msmpeg4 wmv1 wmv2 mpegvideo h263i rv10 mjpeg mp2 mp3 ac3 pcm_s16le pcm_s16be pcm_u16le pcm_u16be pcm_s8 pcm_u8 pcm_alaw pcm_mulaw
Supported file protocols: file: pipe:
Frame size abbreviations: sqcif qcif cif 4cif
Motion estimation methods: zero(fastest) full(slowest) log phods epzs(default) x1

# Anhang: Tips

********** FFMPEG soft VCR documentation *********

0) Introduction
---------------

FFmpeg is a very fast video and audio encoder. It can grab from
files or from a live audio/video source.

The command line interface is designed to be intuitive, in the sense
that ffmpeg tries to figure out all the parameters, when
possible. You have usually to give only the target bitrate you want.

FFmpeg can also convert from any sample rate to any other, and
resize video on the fly with a high quality polyphase filter.

1) Video and Audio grabbing
---------------------------

* FFmpeg can use a video4linux compatible video source and any Open
Sound System audio source:

ffmpeg /tmp/out.mpg

Note that you must activate the right video source and channel
before launching ffmpeg. You can use any TV viewer such as xawtv by
Gerd Knorr which I find very good. You must also set correctly the
audio recording levels with a standard mixer.

2) Video and Audio file format convertion
-----------------------------------------

* ffmpeg can use any supported file format and protocol as input:

Examples:

* You can input from YUV files:

ffmpeg -i /tmp/test%d.Y /tmp/out.mpg

It will use the files:
/tmp/test0.Y, /tmp/test0.U, /tmp/test0.V,
/tmp/test1.Y, /tmp/test1.U, /tmp/test1.V, etc...

The Y files use twice the resolution of the U and V files. They are
raw files, without header. They can be generated by all decent video
decoders. You must specify the size of the image with the '-s' option
if ffmpeg cannot guess it.

* You can input from a RAW YUV420P file:

ffmpeg -i /tmp/test.yuv /tmp/out.avi

The RAW YUV420P is a file containing RAW YUV planar, for each frame first
come the Y plane followed by U and V planes, which are half vertical and
horizontal resolution.

* You can output to a RAW YUV420P file:

ffmpeg -i mydivx.avi -o hugefile.yuv

* You can set several input files and output files:

ffmpeg -i /tmp/a.wav -s 640x480 -i /tmp/a.yuv /tmp/a.mpg

Convert the audio file a.wav and the raw yuv video file a.yuv
to mpeg file a.mpg

* You can also do audio and video convertions at the same time:

ffmpeg -i /tmp/a.wav -ar 22050 /tmp/a.mp2

Convert the sample rate of a.wav to 22050 Hz and encode it to MPEG audio.

* You can encode to several formats at the same time and define a
  mapping from input stream to output streams:

  ffmpeg -i /tmp/a.wav -ab 64 /tmp/a.mp2 -ab 128 /tmp/b.mp2 -map 0:0 -map 0:0

  Convert a.wav to a.mp2 at 64 kbits and b.mp2 at 128 kbits. ‘-map
  file:index’ specify which input stream is used for each output
  stream, in the order of the definition of output streams.

* You can transcode decrypted VOBs

  ffmpeg -i snatch_1.vob -f avi -vcodec mpeg4 -b 800 -g 300 -bf 2 -acodec
  mp3 -ab 128 snatch.avi

  This is a typicall DVD ripper example, input from a VOB file, output to
  an AVI file with MPEG-4 video and MP3 audio, note that in this command we
  use B frames so the MPEG-4 stream is DivX5 compatible, GOP size is 300
  that means an INTRA frame every 10 seconds for 29.97 fps input video.
  Also the audio stream is MP3 encoded so you need LAME support which is
  enabled using ‘–enable-mp3lame’ when configuring.
  The mapping is particullary usefull for DVD transcoding to get the desired
  audio language.

  NOTE: to see the supported input formats, use ‘ffmpeg -formats’.

2) Invocation
-------------

* The generic syntax is :

  ffmpeg [[options][-i input_file]]... {[options] output_file}...

  If no input file is given, audio/video grabbing is done.

  As a general rule, options are applied to the next specified
  file. For example, if you give the ‘-b 64' option, it sets the video
  bitrate of the next file. Format option may be needed for raw input
  files.

  By default, ffmpeg tries to convert as losslessly as possible: it
  uses the same audio and video parameter fors the outputs as the one
  specified for the inputs.

* Main options are:

-L show license
-h show help
-formats show available formats, codecs, protocols, ...
-f fmt force format
-i filename input file name
-y overwrite output files
-t duration set the recording time
-title string set the title
-author string set the author
-copyright string set the copyright
-comment string set the comment
-b bitrate set video bitrate (in kbit/s)

* Video Options are:

-s size set frame size [160x128]
-r fps set frame rate [25]
-b bitrate set the video bitrate in kbit/s [200]
-vn disable video recording [no]
-bt tolerance set video bitrate tolerance (in kbit/s)
-sameq use same video quality as source (implies VBR)
-ab bitrate set audio bitrate (in kbit/s)

* Audio Options are:

-ar freq set the audio sampling freq [44100]
-ab bitrate set the audio bitrate in kbit/s [64]
-ac channels set the number of audio channels [1]
-an disable audio recording [no]

* Advanced options are:

-map file:stream set input stream mapping
-g gop_size set the group of picture size
-intra use only intra frames
-qscale q use fixed video quantiser scale (VBR)
-qmin q min video quantiser scale (VBR)
-qmax q max video quantiser scale (VBR)
-qdiff q max difference between the quantiser scale (VBR)
-qblur blur video quantiser scale blur (VBR)
-qcomp compression video quantiser scale compression (VBR)
-vd device set video device
-vcodec codec force video codec
-me method set motion estimation method
-bf frames use 'frames' B frames (only MPEG-4)
-hq activate high quality settings
-4mv use four motion vector by macroblock (only MPEG-4)
-ad device set audio device
-acodec codec force audio codec
-deinterlace deinterlace pictures
-benchmark add timings for benchmarking
-hex dump each input packet
-psnr calculate PSNR of compressed frames
-vstats dump video coding statistics to file

The output file can be "-" to output to a pipe. This is only possible with mpeg1 and h263 formats.

3) Protocols

ffmpeg handles also many protocols specified with the URL syntax.

Use 'ffmpeg -formats' to have a list of the supported protocols.

The protocol 'http:' is currently used only to communicate with ffserver (see the ffserver documentation). When ffmpeg will be a video player it will also be used for streaming :-)

4) File formats and codecs
--------------------------

Use 'ffmpeg -formats' to have a list of the supported output formats. Only some formats are handled as input, but it will improve in the next versions.

5) Tips
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- For streaming at very low bit rate application, use a low frame rate and a small gop size. This is especially true for real video where the Linux player does not seem to be very fast, so it can miss frames. An example is:

ffmpeg -g 3 -r 3 -t 10 -b 50 -s qcif -f rv10 /tmp/b.rm

- The parameter 'q' which is displayed while encoding is the current quantizer. The value of 1 indicates that a very good quality could be achieved. The value of 31 indicates the worst quality. If q=31 too often, it means that the encoder cannot compress enough to meet your bit rate. You must either increase the bit rate, decrease the frame rate or decrease the frame size.

- If your computer is not fast enough, you can speed up the compression at the expense of the compression ratio. You can use

'-me zero' to speed up motion estimation, and '-intra' to disable
completly motion estimation (you have only I frames, which means it
is about as good as JPEG compression).

- To have very low bitrates in audio, reduce the sampling frequency
(down to 22050 kHz for mpeg audio, 22050 or 11025 for ac3).

- To have a constant quality (but a variable bitrate), use the option
'-qscale n' when 'n' is between 1 (excellent quality) and 31 (worst
quality).

- When converting video files, you can use the '-sameq' option which
uses in the encoder the same quality factor than in the decoder. It
allows to be almost lossless in encoding.

# Fragen / Antworten

1) ffserver doesn't not work.

ffmpeg development is now concentrated on the codec and format
handling. New developments broke ffserver, so don't expect it to work
correctly. It is planned to fix it ASAP.

2) I cannot read this AVI file.

Even if ffmpeg can read the AVI format, it does not support all its
codecs. Please consult the supported codec list on the ffmpeg project
page.

3) I cannot read this Real Audio/Video file.

See (2). ffmpeg only supports Real Video 1.0 and AC3 as Real
codecs. It mean that you cannot view files encoded with real tools
after version 5.0.

4) I get audio/video synchro problems when grabbing.

Currently, the grabbing stuff does not handle synchronisation
correctly. You are free to correct it. It is planned to fix it
ASAP.

5) How do I encode jpegs to another format ?

If the jpegs are named img1.jpg, img2.jpg, img3.jpg,..., use:

```
ffmpeg -i img%d.jpg /tmp/a.mpg
```

'%d' is replaced by the image number.

'img%03d.jpg' generates img001.jpg, img002.jpg, etc...

The same system is used for the other image formats.

6) FFmpeg does not support codec XYZ. Can you include a Windows DLL
loader to support it ?

No ! FFmpeg only supports open source codecs. Windows DLLs are not
portable, bloated and often slow.