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Linux Ubuntu 22.4.5 Manual Pages on command 'mpg123.bin.1'

$ man mpg123.bin.1

mpg123(1) General Commands Manual mpg123(1)

NAME

mpg123 - play audio MPEG 1.0/2.0/2.5 stream (layers 1, 2 and 3)

SYNOPSIS

mpg123 [options] file-or-URL...

DESCRIPTION

mpg123 reads one or more files (or standard input if ``-'' is specified) or URLs and plays them on the audio device (default) or outputs them to stdout. file/URL is assumed to be an MPEG audio bit stream.

OPERANDS

The following operands are supported:

file(s) The path name(s) of one or more input files. They must be valid MPEG-1.0/2.0/2.5 audio layer 1, 2 or 3 bit streams. If a dash ``-'' is specified, MPEG data will be read from the standard input. Furthermore, any name starting with ``http://'' is recognized as URL (see next section).

OPTIONS

mpg123 options may be either the traditional POSIX one letter options, or the GNU style long options. POSIX style options start with a single ``-'', while GNU long options start with ``--''. Option arguments (if needed) follow separated by white? space (not ``=''). Note that some options can be absent from your installation when disabled in the build process.

INPUT OPTIONS

-k num, --skip num

Skip first num frames. By default the decoding starts at the first frame.

-n num, --frames num

Decode only num frames. By default the complete stream is decoded.

--fuzzy

Enable fuzzy seeks (guessing byte offsets or using approximate seek points from Xing TOC). Without that, seeks need a first scan through the file be? fore they can jump at positions. You can decide here: sample-accurate oper? ation with gapless features or faster (fuzzy) seeking.

-y, --no-resync

Do NOT try to resync and continue decoding if an error occurs in the input file. Normally, mpg123 tries to keep the playback alive at all costs, in? cluding skipping invalid material and searching new header when something goes wrong. With this switch you can make it bail out on data errors (and perhaps spare your ears a bad time). Note that this switch has been renamed from --resync. The old name still works, but is not advertised or recommened to use (subject to removal in future).

--resync-limit bytes

Set number of bytes to search for valid MPEG data once lost in stream; <0 means search whole stream. If you know there are huge chunks of invalid data in your files... here is your hammer. Note: Only since version 1.14 this also increases the amount of junk skipped on beginning.

-p URL | none, --proxy URL | none

The specified proxy will be used for HTTP requests. It should be specified as full URL (``http://host.domain:port/''), but the ``http://'' prefix, the port number and the trailing slash are optional (the default port is 80). Specifying none means not to use any proxy, and to retrieve files directly from the respective servers. See also the ``HTTP SUPPORT'' section.

-u auth, --auth auth

HTTP authentication to use when recieving files via HTTP. The format used is user:password.

--ignore-mime

Ignore MIME types given by HTTP server. If you know better and want mpg123 to decode something the server thinks is image/png, then just do it.

--no-seekbuffer

Disable the default micro-buffering of non-seekable streams that gives the parser a safer footing.

-@ file, --list file

Read filenames and/or URLs of MPEG audio streams from the specified file in addition to the ones specified on the command line (if any). Note that file can be either an ordinary file, a dash ``-'' to indicate that a list of filenames/URLs is to be read from the standard input, or an URL pointing to a an appropriate list file. Note: only one -@ option can be used (if more than one is specified, only the last one will be recognized).

-l n, --listentry n

Of the playlist, play specified entry only. n is the number of entry start? ing at 1. A value of 0 is the default and means playling the whole list, a negative value means showing of the list of titles with their numbers...

--continue

Enable playlist continuation mode. This changes frame skipping to apply only to the first track and also continues to play following tracks in playlist after the selected one. Also, the option to play a number of frames only ap? plies to the whole playlist. Basically, this tries to treat the playlist more like one big stream (like, an audio book). The current track number in list (1-based) and frame number (0-based) are printed at exit (useful if you interrupted playback and want to continue later). Note that the continua? tion info is printed to standard output unless the switch for piping audio data to standard out is used. Also, it really makes sense to work with ac? tual playlist files instead of lists of file names as arguments, to keep track positions consistent.

--loop times

for looping track(s) a certain number of times, < 0 means infinite loop (not with --random!).

--keep-open

For remote control mode: Keep loaded file open after reaching end.

--timeout seconds

Timeout in (integer) seconds before declaring a stream dead (if <= 0, wait forever).

-z, --shuffle

Shuffle play. Randomly shuffles the order of files specified on the command line, or in the list file.

-Z, --random

Continuous random play. Keeps picking a random file from the command line or the play list. Unlike shuffle play above, random play never ends, and plays individual songs more than once.

--no-icy-meta

Do not accept ICY meta data.

-i, --index

Index / scan through the track before playback. This fills the index table for seeking (if enabled in libmpg123) and may make the operating system cache the file contents for smoother operating on playback.

--index-size size

Set the number of entries in the seek frame index table.

--preframes num

Set the number of frames to be read as lead-in before a seeked-to position. This serves to fill the layer 3 bit reservoir, which is needed to faithfully reproduce a certain sample at a certain position. Note that for layer 3, a minimum of 1 is enforced (because of frame overlap), and for layer 1 and 2, this is limited to 2 (no bit reservoir in that case, but engine spin-up any?
way).

OUTPUT and PROCESSING OPTIONS

-o module, --output module

Select audio output module. You can provide a comma-separated list to use the first one that works.

--list-modules

List the available modules.

-a dev, --audiodevice dev

Specify the audio device to use. The default is system-dependent (usually /dev/audio or /dev/dsp). Use this option if you have multiple audio devices and the default is not what you want.

-s, --stdout

The decoded audio samples are written to standard output, instead of playing them through the audio device. This option must be used if your audio hard? ware is not supported by mpg123. The output format per default is raw (headerless) linear PCM audio data, 16 bit, stereo, host byte order (you can force mono or 8bit).

-O file, --outfile

Write raw output into a file (instead of simply redirecting standard output to a file with the shell).

-w file, --wav

Write output as WAV file. This will cause the MPEG stream to be decoded and saved as file file , or standard output if - is used as file name. You can also use --au and --cdr for AU and CDR format, respectively. Note that WAV/AU writing to non-seekable files, or redirected stdout, needs some thought. Since 1.16.0, the logic changed to writing the header with the first actual data. This avoids spurious WAV headers in a pipe, for example. The result of decoding nothing to WAV/AU is a file consisting just of the header when it is seekable and really nothing when not (not even a header). Correctly writing data with prophetic headers to stdout is no easy business.

--au file

Does not play the MPEG file but writes it to file in SUN audio format. If - is used as the filename, the AU file is written to stdout. See paragraph about WAV writing for header fun with non-seekable streams.

--cdr file

Does not play the MPEG file but writes it to file as a CDR file. If - is used as the filename, the CDR file is written to stdout.

--reopen

Forces reopen of the audiodevice after ever song

--cpu decoder-type

Selects a certain decoder (optimized for specific CPU), for example i586 or MMX. The list of available decoders can vary; depending on the build and what your CPU supports. This options is only availabe when the build actu?

ally includes several optimized decoders.

--test-cpu

Tests your CPU and prints a list of possible choices for --cpu.

--list-cpu

Lists all available decoder choices, regardless of support by your CPU.

-g gain, --gain gain

[DEPRECATED] Set audio hardware output gain (default: don't change). The unit of the gain value is hardware and output module dependent. (This pa? rameter is only provided for backwards compatibility and may be removed in the future without prior notice. Use the audio player for playing and a mixer app for mixing, UNIX style!)

-f factor, --scale factor

Change scale factor (default: 32768).

--rva-mix, --rva-radio

Enable RVA (relative volume adjustment) using the values stored for Replay? Gain radio mode / mix mode with all tracks roughly equal loudness. The first valid information found in ID3V2 Tags (Comment named RVA or the RVA2 frame) or ReplayGain header in Lame/Info Tag is used.

--rva-album, --rva-audiophile

Enable RVA (relative volume adjustment) using the values stored for Replay? Gain audiophile mode / album mode with usually the effect of adjusting album loudness but keeping relative loudness inside album. The first valid infor? mation found in ID3V2 Tags (Comment named RVA_ALBUM or the RVA2 frame) or ReplayGain header in Lame/Info Tag is used.

-0, --single0; -1, --single1

Decode only channel 0 (left) or channel 1 (right), respectively. These op? tions are available for stereo MPEG streams only.

-m, --mono, --mix, --singlemix

Mix both channels / decode mono. It takes less CPU time than full stereo de? coding.

--stereo

Force stereo output

-r rate, --rate rate

Set sample rate (default: automatic). You may want to change this if you need a constant bitrate independent of the mpeg stream rate. mpg123 automag? ically converts the rate. You should then combine this with --stereo or --mono.

-2, --2to1; -4, --4to1

Performs a downsampling of ratio 2:1 (22 kHz) or 4:1 (11 kHz) on the output stream, respectively. Saves some CPU cycles, but at least the 4:1 ratio sounds ugly.

--pitch value

Set hardware pitch (speedup/down, 0 is neutral; 0.05 is 5%). This changes the output sampling rate, so it only works in the range your audio sys? tem/hardware supports.

--8bit Forces 8bit output

--float

Forces f32 encoding

-e enc, --encoding enc

Choose output sample encoding. Possible values look like f32 (32-bit float? ing point), s32 (32-bit signed integer), u32 (32-bit unsigned integer) and the variants with different numbers of bits (s24, u24, s16, u16, s8, u8) and also special variants like ulaw and alaw 8-bit. See the output of mpg123's longhelp for actually available encodings.

-d n, --doublespeed n

Only play every n'th frame. This will cause the MPEG stream to be played n times faster, which can be used for special effects. Can also be combined with the --halfspeed option to play 3 out of 4 frames etc. Don't expect great sound quality when using this option.

-h n, --halfspeed n

Play each frame n times. This will cause the MPEG stream to be played at 1/n'th speed (n times slower), which can be used for special effects. Can also be combined with the --doublespeed option to double every third frame or things like that. Don't expect great sound quality when using this op? tion.

-E file, --equalizer

Enables equalization, taken from file. The file needs to contain 32 lines of data, additional comment lines may be prefixed with #. Each data line consists of two floating-point entries, separated by whitespace. They spec? ify the multipliers for left and right channel of a certain frequency band, respectively. The first line corresponds to the lowest, the 32nd to the highest frequency band. Note that you can control the equalizer interac? tively with the generic control interface.

--gapless

Enable code that cuts (junk) samples at beginning and end of tracks, en? abling gapless transitions between MPEG files when encoder padding and codec delays would prevent it. This is enabled per default beginning with mpg123 version 1.0.0 .

--no-gapless

Disable the gapless code. That gives you MP3 decodings that include encoder delay and padding plus mpg123's decoder delay.

--no-infoframe

Do not parse the Xing/Lame/VBR/Info frame, decode it instead just like a stupid old MP3 hardware player. This implies disabling of gapless playback as the necessary information is in said metadata frame.

-D n, --delay n

Insert a delay of n seconds before each track.

-o h, --headphones

Direct audio output to the headphone connector (some hardware only; AIX, HP, SUN).

-o s, --speaker

Direct audio output to the speaker (some hardware only; AIX, HP, SUN).

-o l, --lineout

Direct audio output to the line-out connector (some hardware only; AIX, HP, SUN).

-b size, --buffer size

Use an audio output buffer of size Kbytes. This is useful to bypass short periods of heavy system activity, which would normally cause the audio out? put to be interrupted. You should specify a buffer size of at least 1024 (i.e. 1 Mb, which equals about 6 seconds of audio data) or more; less than about 300 does not make much sense. The default is 0, which turns buffering off.

--preload fraction

Wait for the buffer to be filled to fraction before starting playback (frac? tion between 0 and 1). You can tune this prebuffering to either get faster sound to your ears or safer uninterrupted web radio. Default is 0.2 (wait for 20 % of buffer to be full, changed from 1 in version 1.23).

--devbuffer seconds

Set device buffer in seconds; <= 0 means default value. This is the small buffer between the application and the audio backend, possibly directly re? lated to hardware buffers.

--smooth

Keep buffer over track boundaries -- meaning, do not empty the buffer be? tween tracks for possibly some added smoothness.

MISC OPTIONS

-t, --test

Test mode. The audio stream is decoded, but no output occurs.

-c, --check

Check for filter range violations (clipping), and report them for each frame if any occur.

-v, --verbose

Increase the verbosity level. For example, displays the frame numbers dur? ing decoding.

-q, --quiet

Quiet. Suppress diagnostic messages.

-C, --control

Enable terminal control keys. This is enabled automatically if a terminal is detected. By default use 's' or the space bar to stop/restart (pause, un? pause) playback, 'f' to jump forward to the next song, 'b' to jump back to the beginning of the song, ',' to rewind, '.' to fast forward, and 'q' to quit. Type 'h' for a full list of available controls.

--no-control

Disable terminal control even if terminal is detected.

--title

In an xterm, rxvt, screen, iris-ansi (compatible, TERM environment variable is examined), change the window's title to the name of song currently play? ing.

--name name

Set the name of this instance, possibly used in various places. This sets the client name for JACK output.

--long-tag

Display ID3 tag info always in long format with one line per item (artist, title, ...)

--utf8 Regardless of environment, print metadata in UTF-8 (otherwise, when not us? ing UTF-8 locale, you'll get ASCII stripdown).

-R, --remote

Activate generic control interface. mpg123 will then read and execute com? mands from stdin. Basic usage is ``load <filename> '' to play some file and the obvious ``pause'', ``command. ``jump <frame>'' will jump/seek to a given point (MPEG frame number). Issue ``help'' to get a full list of com? mands and syntax.

--remote-err

Print responses for generic control mode to standard error, not standard out. This is automatically triggered when using -s N.

--fifo path

Create a fifo / named pipe on the given path and use that for reading com? mands instead of standard input.

--aggressive

Tries to get higher priority

-T, --realtime

Tries to gain realtime priority. This option usually requires root privi? leges to have any effect.

-?, --help

Shows short usage instructions.

--longhelp

Shows long usage instructions.

--version

Print the version string.

HTTP SUPPORT

In addition to reading MPEG audio streams from ordinary files and from the standard input, mpg123 supports retrieval of MPEG audio files or playlists via the HTTP pro? tocol, which is used in the World Wide Web (WWW). Such files are specified using a so-called URL, which starts with ``http://''. When a file with that prefix is en? countered, mpg123 attempts to open an HTTP connection to the server in order to re? trieve that file to decode and play it.

It is often useful to retrieve files through a WWW cache or so-called proxy. To accomplish this, mpg123 examines the environment for variables named MP3_HTTP_PROXY, http_proxy and HTTP_PROXY, in this order. The value of the first one that is set will be used as proxy specification. To override this, you can use the -p command line option (see the ``OPTIONS'' section). Specifying -p none will enforce contacting the server directly without using any proxy, even if one of the above environment variables is set.

Note that, in order to play MPEG audio files from a WWW server, it is necessary that the connection to that server is fast enough. For example, a 128 kbit/s MPEG file requires the network connection to be at least 128 kbit/s (16 kbyte/s) plus protocol overhead. If you suffer from short network outages, you should try the -b option (buffer) to bypass such outages. If your network connection is generally not fast enough to retrieve MPEG audio files in realtime, you can first download the files to your local harddisk (e.g. using wget(1)) and then play them from there.

If authentication is needed to access the file it can be specified with the -u user:pass.

INTERRUPT

When in terminal control mode, you can quit via pressing the q key, while any time you can abort mpg123 by pressing Ctrl-C. If not in terminal control mode, this will skip to the next file (if any). If you want to abort playing immediately in that case, press Ctrl-C twice in short succession (within about one second).

Note that the result of quitting mpg123 pressing Ctrl-C might not be audible imme?

diately, due to audio data buffering in the audio device. This delay is system de?
pendent, but it is usually not more than one or two seconds.

PLAYBACK STATUS LINE

In verbose mode, mpg123 updates a line with various information centering around the current playback position. On any decent terminal, the line also works as a progress bar in the current file by reversing video for a fraction of the line ac? cording to the current position. An example for a full line is this:

> 0291+0955 00:01.68+00:28.22 [00:05.30] mix 100=085 192 kb/s 576 B acc 18 clip p+0.014

The information consists of, in order:

> single-character playback state (``>'' for playing, ``='' for pausing/loop? ing, ``_'' for stopped)

0291+0955
current frame offset and number of remaining frames after the plus sign

00:01.68+00:28.22
current position from and remaining time in human terms (hours, minutes, seconds)

[00:05.30]
fill of the output buffer in terms of playback time, if the buffer is en? abled

mix selected RVA mode (possible values: mix, alb (album), and --- (neutral, off))

100=085
set volume and the RVA-modified effective volume after the equal sign

192 kb/s
current bitrate

576 B size of current frame in bytes

acc if positions are accurate, possible values are ``acc'' for accurate posi? tions or ``fuz'' for fuzzy (with guessed byte offsets using mean frame size)

18 clip
amount of clipped samples, non-zero only if decoder reports that (generic does, some optimized ones not)

p+0.014

pitch change (increased/decreased playback sampling rate on user request)

NOTES

MPEG audio decoding requires a good deal of CPU performance, especially layer-3. To decode it in realtime, you should have at least an i486DX4, Pentium, Alpha, Su? perSparc or equivalent processor. You can also use the -m option to decode mono only, which reduces the CPU load somewhat for layer-3 streams. See also the -2 and -4 options.

If everything else fails, have mpg123 decode to a file and then use an appropriate utility to play that file with less CPU load. Most probably you can configure mpg123 to produce a format suitable for your audio device (see above about encod? ings and sampling rates).

If your system is generally fast enough to decode in realtime, but there are some? times periods of heavy system load (such as cronjobs, users logging in remotely, starting of ``big'' programs etc.) causing the audio output to be interrupted, then you should use the -b option to use a buffer of reasonable size (at least 1000 Kbytes).

BUGS

Mostly MPEG-1 layer 2 and 3 are tested in real life. Please report any issues and provide test files to help fixing them.

No CRC error checking is performed.

Some platforms lack audio hardware support; you may be able to use the -s switch to feed the decoded data to a program that can play it on your audio device.

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WEBSITE

http://www.mpg123.org

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