1 jack_playfile

jack_playfile — play audio files with JACK

Synopsis

jack_playfile [OPTION]... FILE...

DESCRIPTION

jack_playfile is a simple audio file player for JACK.

Main features:

- Plays most RIFF, AIFF and CAFF PCM WAVE files, also plays FLAC, Ogg Vorbis, Opus and MP3 files
- High quality resampling to match JACK sample rate (custom SR supported)
- Keyboard control while playing (can be turned off)
- Fast and frame accurate seeking
- Supports multichannel files
- Can play files at alternative speed/pitch
- Adapts to a broad range of file formats and JACK settings
- Versatile command line options and live control
- Survives JACK restart (resume playing at the same position)

Please note:

- Opus files are limited to 8 channels by libopusfile.
- FLAC files are limited to 8 channels per stream by spec.
- MP3 supports a maximum of two channels (no "MP3 Surround" support).

Best results are achieved when playing back 32-bit (IEEE 754 single-precision) floating-point RIFF wave files at JACK sample rate (no resampling involved).

Limitations:

jack_playfile can read and play local audio files only. There is currently no support for standard playlists or extended metadata. No DSP except optional resampling and amplification is applied to the audio signal. To meter the signal and apply equalization look out for corresponding JACK clients to chain with *jack_playfile*. This plugin collection is a good starting point: https://github.com/x42/x42-plugins

The term "frame" refers to samples of multiple channels i.e. one frame includes one sample of every channel.

THIS PROGRAM COMES WITHOUT ANY WARRANTY

/!\ *jack_playfile* is still work in progress. It can happen that output is of unexpected nature (i.e. noise, unaligned buffers etc.) in special circumstances due to bugs in the code. If you find such cases that can be reproduced, please consider reporting a bug (see *BUGS*).

This document refers to *jack_playfile* version 0.91.

OPTIONS

-h, --help (w/o argument)
Display help and quit.

-V, --version (w/o argument)
Display version and quit. Static builds of *jack_playfile* will also show the versions of the invovled audio libraries.

-F, --file (string)
Get files to play from playlist file. One audio file per line (full path, unescaped). If a playlist file is given, command line file arguments are not allowed and vice versa. See also --dump.

-R, --recurse
If a given argument looks like a directory, recursively scan it for playable audio files. See also --dump, --file.

-n, --name (string)
JACK client name. **Default:** "jack_playfile"

-s, --sname (string)
JACK server name to start *jack_playfile* in a specific JACK server. **Default:** "default"

-w, --ports (integer)
The port count defines how many output ports *jack_playfile* will expose to JACK, independently of other variables. If the file can deliver less channels than offset+count, the remaining JACK output channels will be filled with zero to match given --ports. This can be handy if files with different channel counts are played in a row (no port recreation or reconnection involved). A fixed port count i.e. --port 4 will ensure that even if a first file would initialize *jack_playfile* with only one JACK output channel (mono file), a following 4-channel file can still play all its 4 channels through the available JACK output ports. **Default:** Dynamically set, all available channels (respecting given offset). This means the file channel count (-offset) of the first file sets the number of JACK output ports *jack_playfile* will have. See also --choffset, --chcount.

-N, --noconnect (w/o argument)
Don't connect to JACK playback ports. **Default:** *jack_playfile* tries to connect all available channels to all available physical output/playback ports 1:1.

-E, --noreconnect (w/o argument)
Don't wait for JACK to reconnect as a client. **Default:** If JACK goes down, *jack_playfile* waits for the server to come back and then continues operation. If *jack_playfile* was playing when JACK went down, it will continue right at the position where it was before JACK went down. If JACK settings were changed between a restart, *jack_playfile* tries to adapt to the new settings as good as possible. If *jack_playfile* is started and no JACK server is availabe, it will wait until it is ready. Reconnecting to JACK means switching the hard- or software playout "backend" *jack_playfile* is using via JACK. JACK supports a variety of backends, including ALSA (Advanced Linux Sound Architecture) (jackd -dalsa), FFADO (Free Firewire Audio Drivers) (jackd -dfirewire), dummy (not attached to any hardware) (jackd -ddummy) and more.

-D, --nocontrol (w/o argument)
Disable keyboard control. **Default:** *jack_playfile* accepts keyboard input while playing. For a detailed overview on available control actions, see *KEYBOARD SHORTCUTS* below or hit *F1* or *h* while *jack_playfile* is started and control is enabled.

-Q, --filtersize (integer)
Manually set the resampler filter size. This value must be one of 16, ... 96. A high value means more use of CPU power. **Default:** 64

-S, --samplerate (integer)
Override file sample rate. This affects how the resampler is setup. The resampler ratio is defined with two sample rates, one of which is the JACK sample rate. Overriding the sample rate that the meta data of the file is indicating results in a different (non-original) playback speed and pitch. For instance, if a file has an SR of 48kHz, overriding with -S 24000 would play the file at half the original speed, i.e. the playback time is double the original length. Not every given SR for overriding the file sample rate is possible. Odd values etc. will not work. The zita-resampler documentation says: The Resampler class performs resampling at a fixed ratio F_out / F_in which is required to be greater than 1/16 and be reducible to the form b / a with a, b integer and b less than 1000. This includes all the *standard* ratios, e.g. 96000 / 44100 = 320 / 147. These restrictions allow for a more efficient implementation. **Default:** Use original file sample rate.

-A, --amplify (float)
Set initial playback volume. Amplify the signal by the given value in decibel [dB]. A value of 0 corresponds to zero amplification (=original volume). Values are limited to maximal +6.0 dB. The amplification can be changed during playback using the appropriate control keys (if control is enabled). **Default:** 0.0 dB

-p, --paused (w/o argument)
Start paused. **Default:** Start playing after successful file open and connection to JACK.

-m, --muted (w/o argument)
Start muted. **Default:** Not muted, i.e. hear sound.

-l, --loop (w/o argument)
Enable loop. If end of track is reached (given offset+count), start over at offset. **Default:** disabled. What happens when the end of a track is reached depends on other conditions.

-e, --pae (w/o argument)
Pause at end: If end of track is reached (given offset+count), *jack_playfile* won't quit but pause playback instead. If loop is disabled, the position will correspond to end. If loop is enabled, the position will be set to start. While paused at end, play, toggle play and forward seeks are blocked i.e. not executed. **Default:** Off. If the end of a track is reached and loop is not enabled, *jack_playfile* is done and will quit.

-j, --transport (w/o argument)
JACK transport: If enabled, *jack_playfile* will follow JACK transport signals "Stopped" and "Rolling", mapped to paused and playing. In the opposite direction, *jack_playfile* sends transport signals to JACK when the play status changes, i.e. spacebar was hit to toggle play. No positional information from JACK transport is processed or sent by *jack_playfile* to JACK. The position of *jack_playfile* is independent, i.e. it can loop while the JACK transport position increments linearly. **Default:** Off.

-f, --frames (w/o argument)
Show time as frames. A number of (multichannel) frames in native file sample rate. Note: The sample rate for Opus and MP3 files is always 48000. This is a consequence of how *jack_playfile* treats these formats, not a limitation of the formats. **Default:** Show time as seconds.

-a, --absolute (w/o argument)
Show absolute time. The frames and seconds indication relate to absolute position 0 of audio samples in file. **Default:** Show relative time. Frames and seconds indication relate to given offset of audio samples in file (offset=relative position 0).

-r, --remaining:: (w/o argument)
Show remaining time. How many frames or seconds until the end of the track is reached (offset+count). **Default:** Show elapsed time. How many frames or seconds away from start (offset).

-k, --noclock (w/o argument)
Disable clock display. This can save some resources. **Default:** Enabled. The display is updated approximately with every JACK cycle.

-o, --offset (integer)
Set frame offset: The first n (multichannel) frames (given number) in the file will be ignored. The frame offset relates to native file sample rate (not JACK's). The offset is relative frame/time position 0 and will be used for seeking to start and looping. Offsets beyond the end of available frames in the file will be set to 0. **Default:** 0 (At first audio sample in file).

-c, --count (integer)
Frame count: A number of (multichannel) frames to play from given offset position. The frame count relates to native file sample rate (not JACK's). Counts resulting in positions beyond the end of available frames in the file will be set to default. **Default:** All available frames, full length of track (respecting given offset).

-O, --choffset (integer)
Set channel offset: The first n channels (given number) in the file will be ignored. Offsets beyond the total file channel count result in *jack_playfile* skipping the file, i.e. nothing will be set to a sane value and nothing will be played. **Default:** 0 (At first channel in file).

-C, --chcount (integer)
Channel count: How many channels to read from file, counting from offset (--choffset). This value could be limited by --ports. **Default:** All available channels (respecting given offset).

-d, --dump (w/o argument)
Print usable files (audio files that *jack_playfile* could possibly play). This applies to files found either in the playlist (--file) or files given as command line arguments. *jack_playfile* will quit (and not play anything) after all files were printed to stdout. All other options are ignored (except --file).

-v, --verbose (w/o argument)
Display more information about loaded audio files and JACK properties.

-L, --libs (w/o argument)
Show license and library information (see *LIBRARIES AND DEPENDENCIES*)

Count and offset relate to the sample rate and duration (frame count) indicated when *jack_playfile* starts up. For the audio formats Opus and MP3, frame offsets and counts always relate to a fixed sample rate of 48k.

To play multiple files, each file can be appended to the command line, i.e.:

```
$ jack_playfile a.wav b.ogg c.flac
```

Please note that files containing spaces or special characters should be enclosed in ″ and if ″ is part of the filename, it needs escaping like \″.

Of course all the shell filename expansion mechanisms can be used, so that

```
$ jack_playfile *.wav 0??.ogg
```

will play all files matching the patterns.

Playing directories recursively:

With Bash globstar:

```
$ shopt -s globstar; jack_playfile music/**
```

If your shell supports this syntax, it will play any file in music/ and all subdirectories therein.

Without Bash globstar:

```
$ jack_playfile --recurse music/
```

Attention: reading file arguments from STDIN will only work if keyboard control is off!

```
#THIS WILL FAIL
$ ls -1 *.wav | ... | while read line; do jack_playfile "$line"; done

#SHOULD WORK
$ ls -1 *.wav | ... | while read line; do jack_playfile --nocontrol "$line"; done
```

However the option to provide a playlist file can be used to achieve a similar operation with enabled keyboard control.

```
$ ls -1 *.wav | ... > pl.txt; jack_playfile --file pl.txt
```

A playlist file is just a list of audio files, one per line. Filenames must not be escaped.

jack_playfile can check which of the files could possibly be played, without playing anything. This is done by trying to open and read (up to a certain degree) every file. If it looks like being an audio file, the filename (and path, equal to provided input) is printed to the screen (stdout). *jack_playfile* will quit afer all files are tested. A scenario is to find all audio files from a list of mixed type files:

```
$ find $PWD/. >files.txt; jack_playfile --file files.txt --dump >audio_files.txt
```

Please note that using the environment variable *$PWD* will make the list more robust (i.e. to play it from another path) since every file in the list is relative to root (*/*).

In the following example, *a.txt* will be sorted out:

```
$ jack_playfile --dump a.wav a.txt a.ogg
```

Other examples:

```
$ jack_playfile --dump --recurse /music
$ jack_playfile -dF files.txt
```

If multiple files are available as arguments or from a playlist file, they will be played in a row without recreating or reconnecting JACK ports. Using keyboard control < and > will browse through the list of files. If a file can't be played, the next file will be tried until there is a valid file or no more files left to try.

KEYBOARD SHORTCUTS

- Start refers to the relative start given with --offset which is 0 by default. Relative start is always 0.
- End refers to relative end which is always equal to --count.
- Default Values are marked with "*"

h, f1
: Help (this screen)

space
: Toggle play/pause

enter
: Play

arrow left
: Seek one step backward

arrow right
: Seek one step forward

arrow up
: Increment seek step size

arrow down
: Decrement seek step size

home
: Seek to start

0
: Seek to start and pause

backspace
: Seek to start and play

end
: Seek to end

< less than
: Load previous file

> greater than
: Load next file

1
: Reset volume (zero amplification)

2
: Decrease volume

3
: Increase volume

m
: Toggle mute on/off*

l
: Toggle loop on/off*

p
: Toggle pause at end on/off*

j
: Toggle JACK transport on/off*

c
: Toggle clock display on*/off

, comma
: Toggle clock seconds*/frames

. period
: Toggle clock absolute*/relative

- dash
: Toggle clock elapsed*/remaining

q
: Quit

If the clock is set to seconds, changing the seek step size will use the following grid:

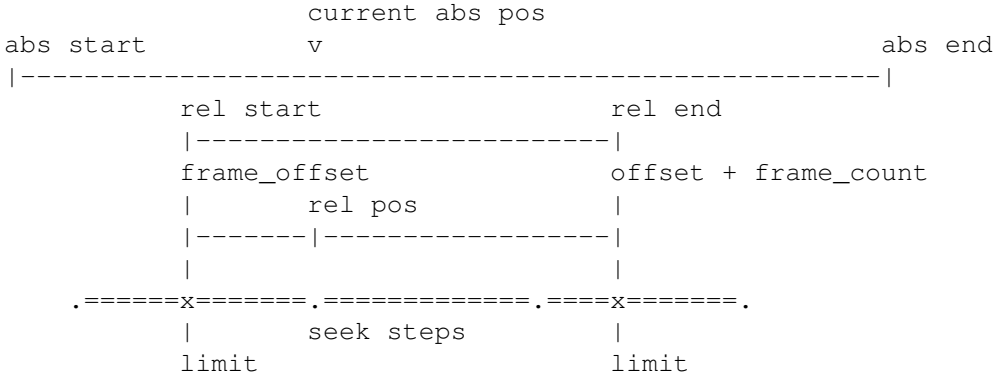
- 0.001, 0.010, 0.100, 1, 10*, 60, 600, 3600 (seconds)

If the clock is set to frames, changing the seek step size will use the following grid:

- 1*, 10, 100, 1000, 10k, 100k, 1000k, 10M, 100M (frames)

TIMELINE

The relation of absolute and relative start and end using offset and count, limited seek steps:



EXAMPLES

- Play RIFF wave file:

```
$ jack_playfile audio.wav
```

Example output of *jack_playfile* using option -v:

```
file #   1/  1: audio .wav
size:        57274264 bytes (57.27 MB)
format:      Microsoft WAV format (little endian)
             Signed 16 bit data (0x00010002)
duration:    00:05:24.684 (14318555 frames)
sample rate: 44100
channels:    2
data rate:   176400.0 bytes/s (0.18 MB/s)
frame_count set to 14318555 (all available frames)
playing frames (offset count end): 0 14318555 14318555
playing channels (offset count last): 0 2 2
amplification: 0.0 dB (1.000)
JACK sample rate: 48000
JACK period size: 128 frames
JACK cycles per second: 375.00
JACK output data rate: 384000.0 bytes/s (0.38 MB/s)
total byte out_to_in ratio: 2.176871
play range:      00:05:24.684
resampler out_to_in ratio: 1.088435
autoconnect: jack_playfile-01:output_1 -> firewire_pcm:000a9200d6012385_MainOut 1 ↩
   L_out
autoconnect: jack_playfile-01:output_2 -> firewire_pcm:000a9200d6012385_MainOut 2 ↩
   R_out
>  playing       S rel    10          4.3  (00:00:04.321)
```

(the last line is being updated in an interval)

Note on ratios:

- byte_out_to_in_ratio: Bytes delivered to JACK divided by bytes read from file. For lossy compressed formats (Ogg, Opus, MP3), the total file size is used for calculation.
- resampler out_to_in ratio: JACK sample rate divided by file sample rate (file sample rate can be customized with -S).
- data_rate: Bytes to read from file per second to satisfy constant flow to JACK output. For lossy compressed formats (Ogg, Opus, MP3), the total file size is used for calculation.

Legend (example prompt):

```
|| paused   JMALP  S rel 0.001       943.1  (00:15:43.070)
^           ^^^^^  ^ ^   ^     ^     ^     ^ ^             ^
1           23456  7 8   9     10    11   10 12           13
```

```
 1): status playing > paused || or seeking ...
 2): JACK transport on/off J or ' '
 3): mute on/off M or ' '
 4): amplification A, ! (clipping) or ' ' (no amp.)
 5): loop on/off L or ' '
 6): pause at end on/off P or ' '
 7): time and seek in seconds S or frames F
```

```
 8): time indication rel to frame_offset or abs
 9): seek step size in seconds or frames
10): time elapsed ' ' or remaining -
11): time in seconds or frames
12): time in HMS.millis
13): keyboard input indication (i.e. seek)
```

- Play Opus file, starting at an offset of 480000 frames (10 seconds), playing 48000 frames (1 second), showing remaining absolute time, pause at end and loop:

```
$ jack_playfile -o 480000 -c 48000 -r -a --pae -l audio.opus
```

- Play a short snippet of all wave files in a directory, using only first channel

```
$ jack_playfile --offset 5000 --count 10000 --chcount 1 samples/*.wav
```

ERROR MESSAGES

jack_playfile does not automatically start a JACK default server if there is none running. If *jack_playfile* is started with the option --noreconnect, this will lead to the following message:

```
Cannot connect to server socket err = No such file or directory
Cannot connect to server request channel
jack server is not running or cannot be started
jack_client_open() failed, status = 0x11
Unable to connect to JACK server
```

Simply start JACK before using *jack_playfile*. If --noreconnect is not present, *jack_playfile* will wait until JACK is reachable:

```
waiting for connection to JACK server...
```

To find out how to start JACK, see *jackd* manpage and tutorials on http://jackaudio.org. There is an excellent graphical JACK control program called *qjackctl*, http://qjackctl.sourceforge.net/.

In a nutshell:

- Starting JACK in realtime mode from a terminal with ALSA backend (i.e. onboard audio), using first available audio card

```
$ jackd -R -dalsa -r48000 -p512 -n3 -dhw:0
```

This can fail for several reasons:

- *jackd* is not installed → check repository for "jackd" or similar and install
- The default JACK server is already running → no need to start again
- The device at hw:0 is already in use by another audio server, i.e. *pulseaudio* → try to stop pulse or try another card (i.e. hw:1)
- You don't have permissions to run *jackd* because of security limits (rtprio, memlock) → check /etc/security/limits.d/audio.conf, check that user is part of group "audio", eventually log out and login to make group changes take effect.
- Other reason

If *jackd* is installed, it's possible to start JACK with a dummy backend where no physical audio devices are involved:

- Starting JACK with dummy backend, server name "virtual"

```
$ jackd --name virtual -ddummy -r4800 -p128
```

- Telling *jack_playfile* to use JACK server "virtual"

```
$ jack_playfile --sname virtual audio.ogg
```

If you have trouble starting *jackd* on your host, please consult JACK FAQ at http://jackaudio.org/faq/ and join IRC #jack on freenode. There is a mailinglist too.

jack_playfile returns 0 on regular program exit, or 1 if there was an error.

PROGRAM STATUSES

(Simplified)

- Initializing (JACK, first/next/prev file)
- Paused (||)
- Playing (>)
- Seeking (. . .)
- Shutting down

PROGRAM LIFE CYCLE

jack_playfile procedure (simplified):

0) Initializing, starting up with given parameters

1) Trying to open given file(s) with several decoders, building an internal list of playable files

2) Check if JACK libraries are available on host, quit on fail

3) Eventually wait for JACK server to become available

4) Register JACK client, register ports, optionally connect ports, quit on fail

5) Start operation based on playback settings (paused, muted etc.)

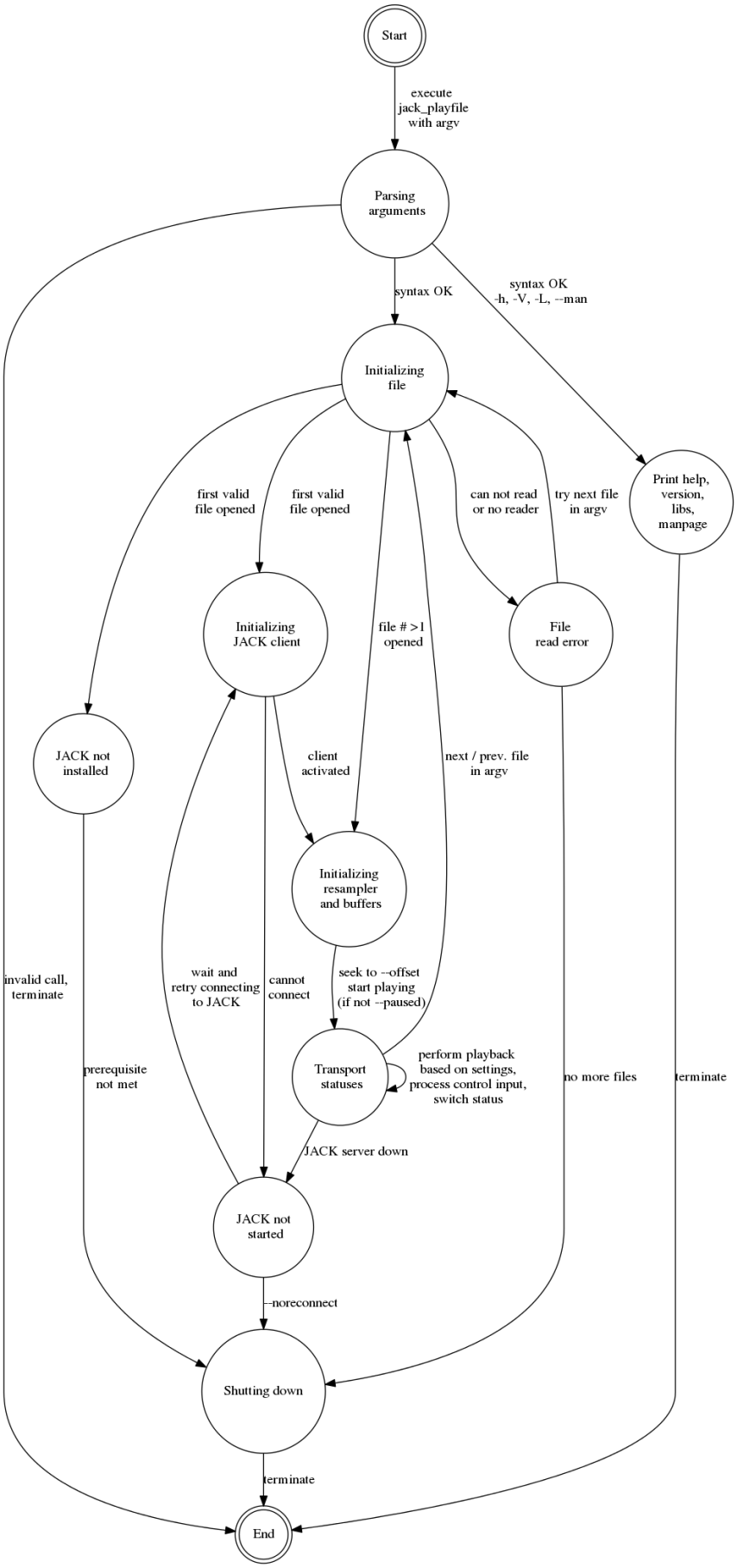
6) Eventually stop operation if JACK away

7) Eventually resume operation if JACK available

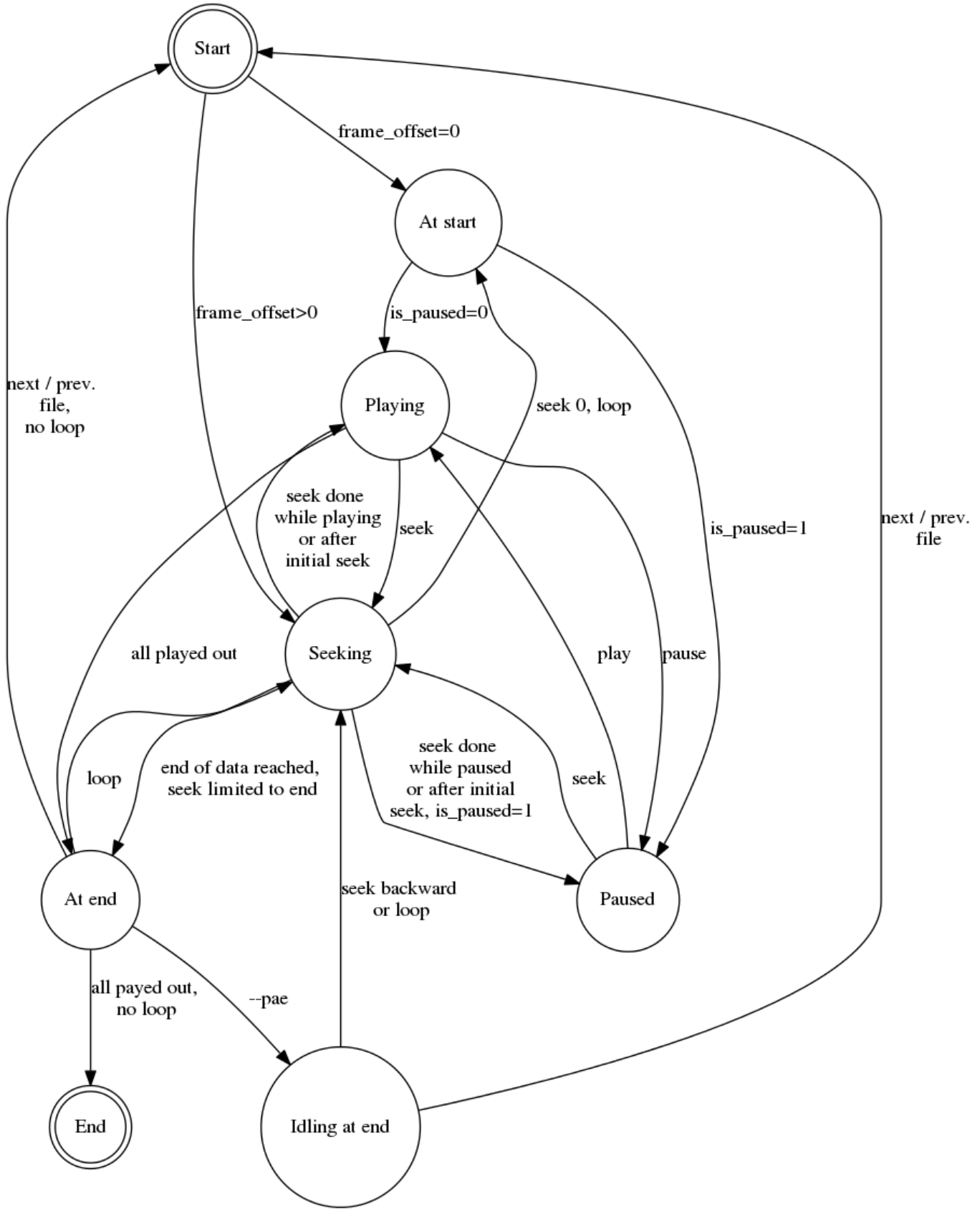
8) Eventually play next or previous file in args list or given playlist file

9) Release resources and quit nicely if all done or quit was requested

The diagram below is not "pure" in any way. It should show some statuses and possible program flows. Unexpected errors that could happen anywhere aren't graphed. (This diagram isn't visible in ASCII text based representations of this document).


Start
execute
jack_playfile
with argv
Parsing
arguments
syntax OK
syntax OK
-h, -V, -L, --man
Initializing
file
Print help,
version,
libs,
manpage
first valid
file opened
first valid
file opened
can not read
or no reader
try next file
in argv
Initializing
JACK client
file # >1
opened
File
read error
JACK not
installed
client
activated
next / prev. file
in argv
Initializing
resampler
and buffers
invalid call,
terminate
wait and
retry connecting
to JACK
cannot
connect
seek to --offset
start playing
(if not --paused)
Transport
statuses
perform playback
based on settings,
process control input,
switch status
prerequisite
not met
no more files
terminate
JACK server down
JACK not
started
--noreconnect
Shutting down
terminate
End

The diagram below is a more detailed view of *Transport statuses*. (This diagram isn't visible in ASCII text based representations of this document).



Example playhead movements:

- Default playback

```
1)
    v Playhead
----|-----------------|-----> time
    offset            offset+count
    (=rel. start)     (=rel. end)

2)
    .(advancing)...v Playhead
----|-----------------|-----> time
    offset            offset+count

3)
    .(end reached)....v Playhead
----|-----------------|-----> time
    offset            offset+count

4) all played out, program terminates
   OR loads next file if available, restart at 1)
```

- Loop playback (--loop)

```
1) - 3) as default

4) seek to offset, repeat from 1)
```

- Pause at end (--pae)

```
1) - 3) as default

4) Idling at end (playhead stays at end, paused)

5) Optional: seek backwards, load previous or next file
```

- Pause at end combined with Loop (--loop --pae)

```
1) - 3) as default

4) seek to offset (playhead at start, paused)
```

During all operation *jack_playfile* tries to prevent to cause JACK X-runs or *jack_playfile* internal buffer underflows. It's very likely that underruns happen inside *jack_playfile* though (not enough data available to play in buffer), i.e. while seeking, during startup or shutdown. *jack_playfile* relies on constant fast file read access. Files can be copied to a RAM disk (i.e. /dev/shm/) before playing to prevent physical disk access on non-SSD disks.

LIBRARIES AND DEPENDENCIES

Major audio libraries *jack_playfile* depends on:

- JACK audio connection kit - http://jackaudio.org/ - *jack_playfile* works exclusively with JACK as audio backend. JACK is available for Linux, Windows and OSX.
- libsndfile - http://www.mega-nerd.com/libsndfile/ - This is the main library to read audio files.
- libzita-resampler - http://kokkinizita.linuxaudio.org/linuxaudio/ - High quality resampler.
- libopus, libopusfile - http://www.opus-codec.org/ - RFC 6716, incorporates SILK & CELT codecs.
- libvorbisfile - http://xiph.org/vorbis/ - Fast seeking in Ogg Vorbis files.
- libmpg123 - http://www.mpg123.org/ - Reading MPEG audio layer 3 files.

Libraries abstracted by libsndfile:

- libFLAC - http://xiph.org/flac/
- libvorbis, libvorbisenc - http://xiph.org/vorbis/
- libogg - http://xiph.org/ogg/

RESOURCES

Github: https://github.com/7890/jack_tools in subdirectory jack_playfile

BUGS

Please report any bugs as issues to the github repository. Patches and pull requests are welcome.

SEE ALSO

jackd(1) **jack_capture**(1) **sndfile-info**(1) **zresample**(1) **flac**(1) **oggenc**(1) **opusenc**(1) **mpg123**(1) **sox**(1) **patchage**(1)

AUTHORS

Thomas Brand <tom@trellis.ch>

Last Update: Sun Jan 17 12:31:03 CET 2016

COPYING