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Contents

Chapter 1

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1.1 PlayHD Documentation

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                PlayHD Documentation
Version 1.0907, September 1998
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1.2 introduction

PlayHD is a 16-bit audio harddisk-recording package, based on the AHI-system by Martin Blom. With PlayHD, you can do multitracking: play a number of samples directly from harddisk and simultaneously record another (stereo) track ↩ .

Thanks to AHI, an increasing number of samplers/soundcards can be used with

PlayHD without re-writing a single bit of code. Even the Amiga's own Paula chip can be used to play samples in a special 14-bit calibrated mode.

Features of PlayHD:

- This version can now play up to a maximum of 20 samples at a time. The amount of samples you can actually play depends on:
  * Processor power: since all the mixing is done by the processor, more channels can be mixed at higher frequencies on faster processors.
  * Speed of your harddisk and harddisk-controller. SCSI is preferred.
  * Samplerate: the higher, the better quality, but more data has to be processed and loaded from the harddisk per second.
  * The AHI-mode you have selected. Is it a stereo mode, stereo++ mode with panning, fast mode etcetera?
  * The kind of sampler used. Samplers that have their own sample buffers are heavily recommended.

- A mixing desk provides you with realtime control over volume, panning and mute per channel. A master volume slider and a stereo master-volume indicator are present too.

- Full mixer-automation based on event-list: channel faders, panning, mutes and master volume can be automated.

- Full duplex playback and record: if your soundcard allows it, you can playback samples while recording.

- Sample-editing with the usual features like cut,paste,copy,erase range. Direct to disk, not limited by memory.

- Effects like Delay, Noise Gate, Chorus, Comp/Limiter, Amplifier, Frequency filters and FIR-filters can be applied.

- MIDI synchronisation: by sending a MIDI start-command and a Song Position Pointer, you can synchronize audio with an external MIDI sequencer. You can also receive MIDI start-commands.

- Supports mono and stereo 16-bit AIFF-samples.

- Possibility to do HiFi non-realtime mixing to a new AIFF-file, thanks to AHI.

- A Time Line Display provides you with an overview of the loaded samples and their durations. Samples can be moved along the timeline.

- A Locator window enables you to set 4 cue-points.

- A time-slider in the Tape Deck window allows for fast locating in the recording.

- etc..

1.3 requirements

```
* Hardware requirements:
   - 68020 processor
   - 0.5 Mb Chip Ram
   - 4 Mb Fast Ram + around 256 Kb per sample
   - A (fast) harddisk

* Software requirements:
   - AHI-system (V4) (from Aminet: ahiusr.lha,
                      or http://www.lysator.liu.se/~lcs/ahi.html)

* Recommended:
   - The faster the processor, the better (ie.68040,68060)
   - 8 Mb Fast Ram
   - A monitor which can display high resolution screens
   - A 16-bit sampler supported by the AHI-system
     (like Delfina, Prelude, Toccata, Wavetools etc.)
   - Fast SCSI harddisk


Benchmarks:
      On a 68030/25 Mhz with a recent IDE harddrive (2.7 Meg/s),
      I can get 11 mono channels at 25681 Hz with the Paula 14-bit stereo
      mode. With the same set-up but with Paula 14-bit++ stereo mode
      I get 7 channels. This is without automation events which cause
      some overhead too.
      With a Prelude soundcard I can play 3 stereo tracks at 44.1Khz
      while recording a stereo track at the same time.
```

1.4 howtostart

```
You should first install PlayHD with the installer that came with the
package. Running PlayHD straight from a CD will most likely give problems
with effectprocessing and recording because samples have to be written
someplace. No assigns are needed.

When PlayHD is installed, you can start it by either clicking on its
icon or by entering 'PlayHD' in a shell.

Please make sure that you select a 1:1 mode from the screenmode requester,
preferably 640x480 or higher. Displays that are too small can cause
problems with overlapping windows.

Note: You can use CyberGraphX screenmodes too, but make sure you choose
      an 8-bit mode when doing this.
```

1.5 usage

```
                This section explains the functionality of each window:
```

Time Line Display

Locator

Control

EffectList

Mixer
and the functionality of the menus:

File

Screen

Sync

Options

Tools
Note that these menus are available in almost every window.

Some other stuff:
- There is also a way to enter timecodes fast: more info can be found

here
.
- Online Help is available in many windows by pressing the 'Help' key
- F1/F2 selects previous/next sample in the Time Line Window
-

Using PlayHD with MIDI
- Notes about recording and master volume levels can be found

here
.

1.6 timelinedisplay

The Time Line Display offers you an overview of the currently loaded
samples and their durations. If a sample has been loaded, a white box
appears on the time-line, matching the duration of the sample.
You can scroll through the time-line using the slider at the bottom of the
window. The width of the time-line adapts itself automatically to the
largest sample.

- Pressing the 'Load Sample' gadget opens a file-requester in which you
can choose one AIFF-sample. The selected sample is then checked for the
right format and some initialisations are made. If this is the first
sample you loaded, the main samplerate is set to this sample's
samplerate. Note that no sample-data is really loaded into memory at
this time. The 'loaded' sample will show up in the listview underneath
the gadgets.
Note: If a sample appears to be an AIFF-file with a wrong samplelength

stored in the header, a window will open where you can choose to let the program fix it.

- 'Remove' removes a sample from the list. Note that this doesn't delete the actual sample from disk!

- 'Move sample' lets you select a different starting location. Just enter the timecode where you want the sample to start and you will see the corresponding white box move in the time-line.

- In the listview, all the currently loaded samples are displayed. Clicking on the name of a sample selects it. This selection can be used in combination with for example the 'Play Single' button in the

Control window or the effects in the EffectList window.

There are also 2 options in the menu Options that determine the look of the samples in the timeline: 'Extended Timeline Display' and 'Volume Overlay'. You can look here for an explanation.

A very important feature is activated when you click on a white sample box: the SampleEdit window is opened with the selected sample, which offers several editing options.

1.7 locator

The Locator window keeps track of four cue points, called L1 to L4. This can be very handy if you have to play or record often from the same location in the recording.
To use this feature, first go to the desired location (for example by using the timeslider in the Control window) and press one of the four 'Set' buttons. The timecode will be displayed in the box right next to the 'Set' and 'Goto' buttons. Now you can jump back from another timecode to the cue point by simply pressing the corresponding 'Goto' button.

Another quick way to jump to a cue point is to select 1-4 on the keyboard (non-numerical part).

1.8 control

The Control window (or also called 'Tape Deck' window) controls the heart of the program.

From left to right there are 5 buttons:

- 'Rewind to start' button: Sets the timecode to zero.

- Stop button: stops playing back. Note that at the moment this button only reacts between loading of sampleblocks.

- Play button: starts playing all samples synchronously. If the option in the 'Sync' menu is set to 'Receive', some sampledata is loaded into memory first and the program will wait for an incoming MIDI start command. See the
  Sync menu
  for more details.

- Play single button: when this button is pressed, the sample selected in the
  Time Line Display
  window is played on its own. Note that the main samplerate is used and not the samplerate of the selected sample. Also note that the mixing sliders are not active when playing single samples.

- Record button: Opens the
  Record window
  .
  This button will be disabled if the selected AHI audio mode doesn't support sampling.

Besides the buttons there is a timeslider which offers you a fast way of locating in the recording. Underneath this slider there is a timecode display in the format minutes:seconds:milliseconds. Note that the 'milliseconds' part doesn't show anything yet in this version.

At the bottom, a checkbox and a 'Bars' field are present. This can be used in conjuction with a MIDI sequencer: if you enter a number of bars in the field at the right bottom and activate the checkbox to the left, the timeslider isn't used anymore, but the starting position of the samples is calculated from the number of bars entered and the 'Tempo' that is set in the
MIDI Settings Window
.

1.9 effectlist

The EffectList window contains a listview with several 'effects'. Double-clicking an effect pops up a new window containing a number of parameters for the selected effect. Note that you must have at least one sample

loaded to view the parameter window, because the effect that is selected will act on the sample you have selected in the listview in the

Time Line Display
window.

All effects that are listed below have a certain 'Apply' button which you can press to actually perform the effect. All effects are non-destructive, unless stated otherwise and the 'effected' sample is saved to the file 'EFFECTSAMPLE' in the 'samples/' directory.
For convenience, the processed sample is loaded automatically into the samplelist.

Most effects can be applied to a selected range in the
Sample Edit window
:
first open the effect window and then open the Sample Edit window. Select the range where you want the effect to be applied and then press 'Apply' in the effect window. A copy of the sample will be made with the effect on the selected part.

At the moment, the following effects are available:

- Delay: Delay with feedback-line with three parameters:
  * Delay time: sets the delay time in milliseconds with a maximum of a two seconds.
  * Mix: the volume of the delayed sound that is mixed with the original sample.
  * Feedback: the volume-percentage that is fed back into the delay line.

- Sync: This is not really an 'effect'. As a matter of fact, this option may seem very strange, but this is actually what started me to write this program! So what does it do? Simply said, it looks for the first audible sound in the selected sample and cuts the first inaudible part of the sample. It does this by looking for a certain volume level above the threshold given by the 'Noise Threshold' slider. If it finds anything, the start position in samples is given in the info box, as a matter of debug info. You can try this out without actually cutting the sample by making sure that the 'Do for real' option is not selected, which isn't by default. Press 'Sync Sample' to try it out or do it for real. This action is destructive if the 'Do it for real' button is checked. If you ask yourself why I created this option, see the background info
here
.

- Noise Gate: This is a noise gate with two parameters:
  * 'Noise Threshold': this parameter sets the level in decibels (dB) at which point sound may pass.
  * 'Release Time': specifies the time in which the gate closes, after the amplitude went under the threshold.

- Change SampleRate: Changes the samplerate of the currently selected sample to the main samplerate that was implied by the first sample or set by the 'Set SampleRate' window in the
Options

menu. The main
samplerate is displayed in the window for reference.
This may be handy for users who sample with the Aura sampler: the sample-program that comes with the Aura writes samplerates not according to the AIFF-standard. If you load samples created by the Aura sampler into PlayHD, the samplerate will be higher. You can adjust this by using this 'effect'. This action is destructive, so the samplerate is actually written to the original file.

- Time Stretching: Time-stretching changes the length of the sample without affecting the pitch. As this is a not very straight-forward algorithm, this effect is prone to all kinds of 'sampling artifacts', especially with stretches beyond 30% of the original length.
  The 'Stretch Factor' slider selects the stretching time and the 'Window Size' is an optimization value you should play with.

- Pitch Shifter: Pitch shifting changes the pitch without affecting the length of the sample. Like time-stretching, this can produce certain sampling artifacts, especially when shifting more than 5 semi-tones up or down. Use the 'Pitch' slider to set the amount of semi-tones you want to shift.

- Amplifier: With the 'Amplifier' you can increase or decrease the overall volume level of the selected sample. When the volume exceeds the maximum level, unnatural sounding clipping can occur.

- Frequency Filters: Frequency filters remove a certain frequency spectrum from a sample. There are 5 different filters you can choose from:
  * Low Pass: Use the 'High Freq' slider to set the highest frequency that may pass.
  * High Pass: Use the 'Low Freq' slider to set the lowest frequency that may pass.
  * Bandwidth: Use both sliders to set the frequency range that may pass. Make sure that the 'Low Freq' slider is set lower than the 'High Freq' slider.
  * Reject: The opposite of the 'Bandwidth' filter: the frequency spectrum you select is rejected.
  * Resonator: Don't know about this one. I just found some code and implemented it :-)

- FIR Filter: You can load a FIR-filter file that was created by either AudioLab (not tested) or the program 'FIRdesigner' (Aminet: FIRDesigner.lha). FIR-filters are also frequency filters, but are far more advanced than the previous 'Frequency Filters' because many different frequency-bands can have a different gain, like an equalizer.
  If a FIR-filter file is loaded, the frequency graph is shown in the window and the amount of taps/coëfficients is displayed. The higher the amount of taps, the longer it will take to perform the FIR-filter. To create your own FIR-filters, use the 'FIRdesigner' program on Aminet by Harald Zottmann. I'm planning to write a FIR-designer myself in the future that will be integrated in the program.

- Chorus: A chorus gives the effect of multiple voices/instruments singing/playing at the same time. It has the following options:

    * Speed: how quick the modulation changes
    * Depth: determines how much modulation is used
    * Mix: the amount of 'chorus' that is added to the sample

- Comp/Limiter: A compressor fits a certain dynamic range on a smaller one: use the threshold slider to set the upper limit where the sound may pass freely. Above this setting, audio is compressed with the ratio you can set with the ratio-slider: for example, if you set this to 30:10 (ie. 3:1), every 3dB above the threshold will result in an output of only 1dB above this threshold (this is called gain-reduction).
  You can use the 'Limiter' threshold to set the maximum allowed output value: every sample that is louder than this limit will be set to this maximum value.

- Parametric EQ: Under construction!

- Ring modulation: A ring modulator adds a kind of 'metallic' sound to the sample.

- Reverse: Reverses the sample or samplepart.

- Reverb: Under construction!

1.10 mixer

The Mixer window is very straightforward: each channel has its own panning control, mute knob and volume slider. If a sample is loaded, its name will be displayed right next to the corresponding channel unit for an easy overview.

At the right side of the window, a red slider adjusts the master volume level. Each time a sample is loaded, the volume range (dynamic range) is increased: if one sample is loaded, the range is from 0 (silence) to let's say 1 (full volume). If the second sample is loaded, the range is from 0 to 2, and the master volume adjusts itself to the previous value. The next range is from 0 to 3 and so on... This may seem a little strange (and it is, actually), but because of the nature of mixing, volume levels will decrease when mixing more and more samples. To account for this, the volume levels can be increased by the above explained method. Be warned that too high settings of the master volume can result in unwanted clipping.

The two bars at the right show the left and right output volume. These bars respond quicker in play-single mode. You can turn these bars off in the
Options
menu.

The
Auto
button brings you to the mixer automation windows where you

can automize the whole mixer setup.

1.11 sampleedit

The SampleEdit window offers you a visual representation of the sample selected with the 'Time Line Display' window and allows you to edit the sample in many ways.

* SampleView area:
  The main part of this window shows the sample or part of the sample. You can use the scrollbar underneath it to scroll through the whole sample if it has been zoomed in. You can view the timecode at the current mouse position at the top-center of the window next to 'Position:'. This is shown in the format minutes:seconds:milliseconds.

* Range selection:
  To select a range, press the left mouse button where you wish the range to start and then move the mousepointer to the desired end-position while holding down the left mouse button. The range you select will appear in white. The start- and end-timecodes of the range will be displayed at the top-left of the window.

* Buttons at the bottom:
  - Play: Plays the whole sampleview area when no range is selected or otherwise only the range. A vertical line shows the current position during playback.

  - Stop: Stops playback of the sample.

  - Clear Range: Clears the range you have previously selected.

  - Show Range: Zooms in on the sample, resulting in the largest display of the selected range.

* Magnification:
  When a sample is displayed, it has a minimum magnification value, shown in the box just below the word 'Magnification:'. This value actually represents the 'pixels per second'. You can alter this value by entering a new value in this box or by using the '-' and '+' gadgets. Pressing one of the latter gadgets results in a halve or double magnification factor, but the sample display is not directly updated until you press the button called 'Display'.

* Tools:
  - Erase Range: Erases the selected range by writing all zeroes to that part of the sample. You can use this for eliminating some unwanted noises in recordings or doing manual noise gating. This action is DESTRUCTIVE (!), which means that the action takes place directly on the sample.

  - Cut Range: Cuts out the selected range, copies it into the buffer and joins the two 'non-selected' parts together, shortening the samplelength. This is also a destructive action.

- Copy to Buffer: Copies the selected range to a buffer-file, called TEMPSAMPLE. This file is an AIFF-file on its own, so it can be copied or renamed if you want to use the selected range as a separate AIFF-sample.

- Copy Over: Copies the buffer over a part of the selected sample. To use this feature, you must first set the position at where you want the copying to start. You do this by selecting a range with a one-pixel width at the desired starting position. Then press the 'Copy Over' button and the entire buffer will be copied over the original sample. The sample is expanded when needed. This action is destructive.

- Insert: Inserts the buffer into the selected sample at a given point. See the explanation of 'Copy Over' (above) for more information on how to set this point. This action is destructive.

- Save Buffer: Lets you rename the last saved buffer from TEMPSAMPLE to another name. After this, the sample can be used in your project.

- Copy to Track: Copies the selected range to another sample at the same timecode. A window will pop up where you can set the destination sample and select whether you want to overwrite the sample or want to mix the original and destination sample with given mix percentages.

1.12 filemenu

* New Project: Creates a new project and removes the current one from memory.

* Load Project: Opens a file-requester where you can select a previously saved project. A project contains the following items:

  * Samplenames with their offset, volume and panning
  * Automation events
  * Sync Mode
  * MonitorVolume
  * Recording input
  * Selected audio output device
  * Tempo
  * Initial mixer settings
  * Locators

  Note that a project doesn't contain the actual samples itself, it just keeps a reference to them.

* Save Project: Opens a file-requester where you can enter a filename for your current project. The items listed above will

be stored in the project file.

* Delete: - Sample: Deletes the currently selected sample in the

Time Line Display
window from (hard-)disk
and removes it from the samplelist.
Warning: you can't get it back once you have
deleted it!

- Project: Deletes the current project from harddisk and
starts a new project. A requester will ask you
if you want to delete every sample of the project
from harddisk.

* Online Help: Opens the main page of PlayHD.guide

* Audio Mode: Lets you change the current AHI audio mode

* Screen Mode: Lets you select another screen mode. If you want to use
the new screen mode, you will have to choose 'Save
Preferences' in this menu, quit and start PlayHD again.

* Save Preferences: Saves the following items in the 'PlayHD.prefs' file:

* Screen mode
* Audio mode
* Time Correction Factor

When the preference-file is present at start-up,
PlayHD will load and use these settings.

* About: Some version and author information

* Quit: quits the program

1.13 screenmenu

Currently, the midi screen is not available.

1.14 syncmenu

This menu has three options regarding MIDI synchronisation:

* Ignore: Ignores any MIDI start command or Midi Machine Control (MMC)
code. When the Play-button is pressed in the Control window,
the samples will be played without waiting for anything.

* Send: Sends a MIDI start command (and MMC-start when this option is
selected in the
Options

menu) after loading the first sample-
parts in memory when the Play-button has been pressed. A Song
Position Pointer is also send.

* Receive: If this option is selected, the program first waits for an
incoming MIDI start command before playing when the Play-
button has been pressed. Of course, the first sampleparts
are loaded into memory first for optimal synchronisation.

See the section
Using PlayHD with MIDI
for more details on MIDI.

1.15 optionsmenu

There are four options/actions at the moment in this menu:

* Volume Display: Turns the volume indicator in the Mixer window on or
off.

* MIDI Machine Control: Sends an MMC-start command when the option
'Send' is selected in the
Sync
menu.

* Automation Display: Turns the visual display of the automation events in
the Mixer window on or off.

* Volume Overlay: Toggles the visual display of the volume curve (determined ↩
by
initial volume and automation events) in the
Time Line Display
.

* Extended Timeline Display: Toggles between full sample view in the ↩
timeline
display or just a white bar representing the time and ↩
offset
of the sample.

* MIDI Bar Display: If this option is switched on, the timeline display
will show the time in MIDI bars instead of minutes.

* Recording options: When this option is selected, the
Record window
is
opened in which you can set some recording options.
Note that this can only be selected when the selected
AHI audio mode supports sampling.

* Set SampleRate: This opens a window in which you can alter the main
samplerate. Note that this doesn't write the sample-
rate to the samples. If you want to write a new
samplerate to a sample, see the
EffectList

window.

* MIDI Settings: Opens the MIDI Settings Window .

1.16 toolsmenu

The tools-menu contains only 1 option at the moment:

* Channel Splitter: This allows you to split a stereo sample into a left and right mono channel.

1.17 recordoptions

This window lets you select the following recording options:

* Input: This is a list with all available inputs for the selected audio mode. Refer to the documentation of your sampler for more info.

* Output: This is a list with all available outputs for the selected audio mode. Again, refer to the documentation of your sampler for more info.

* Monitor: This is the monitor volume that is passed to the selected 'Output' while recording.

* Input Gain: Sets the input level of the selected 'Input'. This option is only available if your sampler has variable input gain.

* Output Volume: Adjusts the overall output level of the selected 'Output'. This option is only available if your sampler has a variable output volume.

* Listen: This checkbox allows you to listen to the incoming signal of the selected 'Input'. You can adjust the monitor level while listening. Uncheck this box when you're done.

* Use/Cancel: Press 'Use' if you want to keep these options or press 'Cancel' if you wish to forget about them.

If you checked the 'Listen'-box, you can view your input level in the two level bars at the bottom of the window. To the right of these bars there are separate peak indicators for the left and right channel. Adjust the input gain while listening and make sure that you have a good dynamic range: peaks should be at around -2 dB. If you lower your input gain you can reset your peak indicator to -96 dB by pressing the 'Reset' button.

1.18 entertimecode

There is a way of entering timecodes manually by pressing the 'Enter' key on the numerical keyboard. This opens a small window in which you can enter the timecode in minutes and seconds very quickly. At the moment, there is a maximum of 9 minutes you can enter, so the first digit you enter represents the minutes. Then you can enter a dot ('.') followed by the seconds. You can close the window by pressing 'Enter' again.

1.19 future

The Future!

There is still a lot of work to do and I have a feeling that development is not going to cease in the following years.

Major things I have planned:

- Porting the program to PowerPC.
- When the pre/Box arrives, I'm going to port everything to the pre/Box!

Minor things I have planned in the near future:

- (Much) Better effect-processing: Timestretching and pitch-shifting sound really bad now. Things like backward delay,phaser,flanger and so on will be added.
- Adapting display to different screen sizes
- Better recording facilities
- Dynamically changing the amount of tracks
- Better sample-editing (multiple windows etc.)
- Updating the GUI
- etc.

1.20 background

This whole project started during the summer of 1996 when I wanted to do some home-recording. I am a guitar player and wanted to record some Joe Satriani covers for fun. The backline (drums,bassguitar,synth) was recorded with a synth and the midi-sequencer Music-X so I could play along with this. Normally, everything went fine if I only had to record one guitar part: I just mixed the output of my synth with my guitar and recorded directly to DAT.

But then I wanted to record some songs with multiple guitar parts.

This was a major problem because I didn't have a multitrack recorder. Then I lend an Aura sampler from a friend and wanted to record everything (multiple guitar parts + synth) track by track. Since I had two Amiga's available this was no problem: one for recording with the Aura and one for playing the midi-tracks with Music-X.

The idea was then to record everything to different samples and then to mix these samples. But then a big problem arised: there was no way of synchronizing the start of the recording to a MIDI start command with the Aura software! So recording of the different parts resulted in different start offsets, which made it practically useless. Then I had this great idea to record four metronome ticks with every sample as a reference and after recording I would 'shift' the samples to the left: I had to cut the first part until the first metronome tick was heard. I started to program the mixer and then a simple algorithm to recognise the first metronome tick, which was just a search for a certain sample value above a certain threshold. This worked all fine, but I was interested in doing more. I saw a program called 'Play16' (which still exists and is being updated regularly I believe...) by Thomas Wenzel which could play 16-bit samples in a special 14-bit mode using the standard Amiga audio outputs. As I wanted to do that too, I got some assembly and information from him but at the same time I most pleasantly bumped into the AHI home page on the internet. I downloaded the AHI-package and started playing with it with great success. So I rewrote my mixing program to make use of the AHI-device and then things started to develop and grow step by step, which it still does.

Meanwhile I received my Prelude soundcard which allows me to really use and test PlayHD. Now it's only a matter of waiting for my PowerUp-board!

So far for the moment... If you want to know more just
mail me
.

By the way, this program is mainly being programmed on an A4000 with a 68030/25 Mhz,68882/50 Mhz and 2Mb Chip + 18Mb Fast and a CyberVision64 graphics card.

1.21 problems

Known problems:

- Cybergraphics screens with a depth larger than 8-bit cause some windows to display very slowly.
  Solution: use 8-bit screens, only 16 colors are used at the moment.

- Some effects are mono (Pitch Shifter, Freq.filters, and Chorus)

- PlayHD and an external MIDI sequencer are playing out of sync.
  Solution: use the Time Correction Factor in the MIDI Settings window.

- The master volume automation doesn't update at the correct speed with a soundcard but is correct with Paula audio.

Solution: Not solved yet...

1.22 contact

If you have any questions, suggestions, criticism or wish to have the full version, feel free to contact me at the following address:

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1.23 acknowledgements

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- Misja van Waterschoot for testing PlayHD, giving tips and lending me your sampler!
- Martin Blom for his wonderful AHI-package
- Jeroen Vermeulen for providing me with the camd-package
- Everybody on c.s.a.audio who helped me with my troubles regarding the midi.library and camd.library

1.24 legal stuff

LIMITATIONS
The demo version is limited to only 3 (stereo) tracks and 90 seconds of playback- and record time. The full version will have an unlimited number of tracks and unlimited playback- and recordtime. For obtaining the full version, please
contact me
.

PRICING
There has been a lot of work in this program and there will be even more in the future. That's why this program probably is going to be shareware or commercial with an expected pricing between 100 DM / 58 US dollar and 200 DM / 116 US dollar.

DISCLAIMER
This software is provided as-is, without warranty of any kind, either expressed or implied. In no event will the author be liable for direct, indirect, incidental or consequential damages or data loss resulting from the use or application of this software. The entire risk as to the results and performance of this software is assumed by the user.

1.25 recordwindow

Not every option is working at the moment!

Things that are implemented:

* Record button: If you press this button, the program will start recording from the input you have selected at the Recording Options window.
  If you have selected 'Sync Receive' in the Sync Menu, the program will first wait for a MIDI start command before starting the recording.

* Set Rec-Start: Pressing this button pops up a window where you can enter a precise start time for recording. You can check the current start time to the right of this button.

* HD-light: To check whether the audio data is really written to disk, a harddisk-led is present here. When recording, if flashes red when something is written to disk. If you pressed 'Record' and nothing happens within in a few seconds, something probably went wrong. You should stop the recording then and try again.

After recording a window pops up in which you can choose to keep or delete the sample you just recorded.
If you select 'Keep', a file-requester will pop up in which you can rename the sample. Then, the sample is added to the current project.
If you select 'Delete', the sample will be removed from your harddisk.

1.26 automationwindow

With the automation window, you can automate all the parameters of the mixing window. For example, you can choose to fade in some channels at the beginning of your music piece, and at the end you fade them all out by automatically decreasing the master volume over lets say 10 seconds. This will give you smooth fades which you don't have to do manually while mastering your recording.

At the left side of the window, there is a listview which displays all the automation events that are currently present. From left to right it shows:

* the channel on which the event takes place (if it is a master volume event, nothing will be displayed here)
* the samplename (again, not with master volume events)
* start-time of the event (in minutes:seconds:milliseconds)
* stop-time of the event
* the event-type (one of MUTE, CHANNEL VOLUME, MASTER VOLUME, PANNING)

The events are ordered to their start-times, but of course, events may overlap.

You can add, edit or delete events by pressing their corresponding buttons

right next to the listview: adding or editing an event brings you to
the
Event Window
.

You can also create a new eventlist, load it or save it. Note that the
eventlist is also saved with the project, but you can create multiple
eventlists if you wish, for different mix-downs.

1.27 eventwindow

At the Event window, you can choose the sample on which the event will
take place, the actual event-type, the start- and stoptimes for the
event and, if applicable, some settings depending on the event-type:

* For MUTE events, you can only select the start- and stoptimes
and of course the sample to be muted.

* The CHANNELVOLUME event lets you fade a sample in or out. The
duration of the fade is determined by the entered start- and
stoptimes. You can set the begin-volume with the first slider
at the right bottom of the window. It will be initially set to
the current volume that is set in the Mixer window to prevent
'jumps' in the volume. The same goes for the end-volume.

* Choosing the MASTERVOLUME event will disable the samplename-
selection because the master volume will affect all samples.

* The PANNING event gives you two sliders for the start- and
stop positions in the panorama. A value of '-100' means
all the way to the left, '0' is in the middle and '100' is
all the way to the right.

You can choose 'Add Event' when you are done with your settings or you
can choose 'Cancel' to stop editing.

1.28 midi settings window

The MIDI Settings Window has the following options:

* Tempo Slider: if you use the 'Bars' indicator at the
Control
window,
this tempo is used to calculate the starting position.

* Time Correction Factor: For some reason, timing in MIDI sequencers is not ↩
100%
correct. So if you use PlayHD with an external sequencer or ↩
have
a second Amiga connected via MIDI running a sequencer, ↩
timing

problems may occur. As this error is cumulative, a correction
factor can be set to solve this problem. The start position of
the play routine will be corrected with (1/TCF). This means that
for every second, (1/TCF) seconds will be added.
To make life easier, I have included two TCF's for two common
MIDI sequencers which work for me:

- Music-X : TCF = 1500;

- Bars&Pipes Pro : TCF = 110;

This means that B&P has a very large error!

If you have another MIDI sequencer and want to find out what
its TCF is, record a long sample (let's say 12 minutes) in
which you clearly play sync to the beat.
Then see if everything is in sync from the start. If that's
not the case, something else is going wrong. See the section

Sync Menu
for more details. If everything is fine, check the
synchronisation at 1 minute and adjust the TCF if necessary:
start with a value of around 1000 and listen if your sample
is ahead or behind the MIDI tracks. If it's ahead, increase
the value and v.v.
Repeat this for 2,4,8 and 12 minutes.

A TCF of 0 means that it isn't used.

1.29 using playhd with midi

PlayHD can be used together with an external MIDI sequencer: a second Amiga
running a software MIDI sequencer can be connected via MIDI or you can
attach a hardware MIDI sequencer. To synchronize the two, you can set
the MIDI Sync to 'Send' or 'Receive' (see the
Sync Menu
for more details).

- If the Sync option is set to 'Send' and the play- or record button has been
  pressed, first the right start positions of the samples are searched,
  then a MIDI-start command is sent, together with a Song Position Pointer
  and playback or record is started.
  Because the MIDI sequencer might not react instantaneously, a delay might
  occur. In this case, you should select the Sync option 'Receive' and let
  the sequencer be the 'master'.

- If the option 'Receive' is set and the play- or record button has been
  pressed, the right start positions of the samples are searched and the
  program waits for a MIDI Start or Continue command. If one of these is

received, playback or record starts immediately.

It is preferred to use the 'Receive' option. Because it's not possible to receive a MIDI Song Position Pointer and to set the starting positions of the samples immediately thereafter, you will have to set the time counter or 'Bars' indicator by hand. The 'Bars' indicator is preferred, because the resolution of the time slider is in seconds which probably won't match the time indicator on your MIDI sequencer. The 'Bars' indicator in conjuction with the 'Tempo' setting in the
MIDI Settings Window
will
allow you to position your recording at any place you want.

See the section
MIDI Settings Window
for fixing timing problems
which may be occur in the middle of a recording.

1.30 notes about recording

Notes about recording and master volume settings

As you can read in the
Mixer
section, everytime a sample is loaded
or recorded, the volume range is increased. As the size of the master volume slider isn't changing, this means that your master volume setting on the slider will have a different meaning depending on the amount of samples you have loaded. On the other hand, because mixing is done by adding up all samplevalues for each track and then diving by the amount of samples, you'll probably loose some dynamics because not every track will be using the full dynamic range at every moment.
So, if you would keep the master volume fixed and then play more and more samples, you would loose volume instead of gaining it. Also, if you would record a track while playing another track, you would hear the already recorded track too soft.

PlayHD tries to prevent this by adding some volume everytime you load or record a sample. Note that this might lead to some distortion when you are playing or recording samples that are very loud.

Additionally, when you're going to record something, you should check your input gain in the
Record Options window
. Make sure you have a good dynamic
range and peaks near the -1 or 0 dB for loud pieces, but make sure you don't set it too high, because recording too loud on digital equipment sounds really bad!

These two things will most likely reduce volume problems and of course you can always set your mastervolume yourself!

Note: With the 'Fast' AHI-modes the volume will be crudely rounded
to levels like 100%, 50%, 25% and so on. This of course will limit your
level adjustments but takes far less CPU-time.