

**in**

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# Chapter 1

## in

### 1.1 Audio Evolution Documentation

Audio Evolution Documentation  
Version 2.0820, August 1999

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Published by Sultan Systems & Software  
Worldwide distribution by Computer City

Introduction

Requirements

How to start

Usage

The future

Known problems

How to contact us

Acknowledgements

Demo limitations and ordering

### 1.2 introduction

Audio Evolution is an audio harddisk recording system primarily aimed at musicians who want an easy-to-use multitrack recorder for home-studio applications. It's based on the AHI system by Martin Blom, which makes it possible to run Audio Evolution on every modern soundcard designed for Amiga computers. Because of the efficient mixing routines of AHI, it's

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possible to play up to 30 tracks at the same time on expanded systems, even without PowerPC technology. But that's not what makes it a good harddisk recording system: Audio Evolution also offers a versatile mixing desk including individual channel and master level display, mute, solo, panning, subgroups and realtime effects. Besides that, full mixer automation gives you mixing possibilities only limited by your imagination. Also take a look at the timeline display with its marker system for fast positioning in the song, locators and punch in/out features which saves you precious time in your recording session.

#### Features of Audio Evolution:

- Up to 30 (stereo) samples simultaneously.  
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.
    - \* The AHI-mode you have selected. Is it a stereo mode, stereo++ mode with panning, fast mode etcetera?
  - Full duplex recording for simultaneous recording and playback
  - A mixing desk provides you with realtime control over volume, panning, solo and mute per channel. Every channel has a separate volume PPM display. Also three realtime effects can be plugged in and the channel can be assigned to one of four subgroups.  
A mastervolume slider and a stereo master volume indicator are present too.
  - Separate window for 4 subgroups with solo and mute per group.
  - Full mixer-automation based on event-list: channel faders, panning, mutes, master volume, subgroup volumes and subgroup mutes can be automated. Automation events can be edited directly on the timeline.
  - PowerPC realtime effects with the following runtime (near real-time) possibilities:
    - \* Parameter change
    - \* On/off state switch
    - \* Removal of effects
- Also, possible CPU overload is detected, preventing lock-ups during playback.
- A Time Line Display provides you with an overview of the loaded samples and their durations. Samples can be moved along the timeline.
  - A marker mechanism on top of the timeline lets you place locators, punch in/ out markers and the time marker easily, giving quick and accurate access to these items.
  - 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, Parametric EQ ← 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, synchronize to Bars&Pipes or to the camd.library.
- etc.

### 1.3 requirements

\* Minimum hardware requirements:

- 68020 processor
- 0.5 Mb Chip Ram
- 4 Mb Fast Ram + around 256 Kb per sample
- A (fast) harddisk
- A 16-bit sampler supported by the AHI-system (like Prelude, Melody1200 etc.)

Note: recording through the 8-bit Paula modes is NOT supported!

Note2: Delfina Lite and Concierto might give problems. Delfina Lite users should upgrade to the Delfina Plus.

\* Software requirements:

- AHI-system (V4) (from Aminet: ahiusr.lha, or <http://www.lysator.liu.se/~lcs/ahi.html>)
- popupmenu.library v8.0 (included in the Audio Evolution package)
- reqtools.library v38.0
- Phase5 ppc.library for PPC realtime effects

\* Recommended:

- The faster the processor, the better (68060 preferred)
- Fast SCSI harddisk or a modern IDE controller like the PowerFlyer or IDEFix Express (anything but the standard PIO Mode 0 IDE controller!)
- 8 Mb Fast Ram or more
- PPC card for realtime effects
- Graphics card + 17" monitor for high resolution screens

Benchmarks:

On a 68030/25 Mhz with a recent IDE harddrive (2.7 Meg/s), I can get 11 mono channels of playback 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.

On an '060 with SCSI-2 (Prelude or Delfina Lite 'fast' mode) you can play 20 stereo samples at 48 KhZ or 30 mono samples!

A '040/25 Mhz with IDE gives you around 7 to 8 stereo tracks and 12 to 13 mono tracks.

## 1.4 howtostart

You should first install Audio Evolution with the installer that came with the package. When Audio Evolution is installed, you can start it by either clicking on its icon or by entering 'AudioEvolution' in a shell.

Select a mode for the soundcard you are going to use. Preferably a fast stereo++ mode if that's available and preferably no Hifi or non-realtime modes. Delfina users should check if they have the latest versions of the Delfina libraries and shouldn't use the 'AnyFreq' modes because the AHI driver can give some troubles.

Please make sure that you select a 1:1 mode from the screenmode requester, preferably 800x600 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

Marker Window

Control

EffectList

Mixer

and the functionality of the menus:

File

Windows

Sync

Options

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## Tools

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
- An explanation of all the used keys can be found

here

-

Using Audio Evolution with MIDI

- Notes about AHI modes, recording and master volume levels ↔ can be

found

here

.

- Notes on PCG files

## 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.

If the menu option 'Extended Timeline Display' has been selected, the graphical sample data will be displayed in the box too.

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.

The top of the timeline is used for the 'Marker' system: little flags that can be moved along the timeline. Click

here

to get more information

on this topic.

If you press the right mouse button in a sample box, a popup-menu appears with the following items:

- Edit
- Move
- Remove
- Create PCG
- 
- Copy to
- 
- Automation...
- Mute
- Volume
- 

Add workspace

The first four items are explained below because they have the ↔ same

meaning as their corresponding buttons.

You can zoom in on the samples using the zoom-level slider on top of the timeline. The zoomlevel is shown in pixels per second. The usage of

PCG's  
will speed-up the display a lot.

Changing the offset of a sample can be done in two ways: pressing the 'Move' button which opens a window in which you can manually enter the offset, or by clicking on a sample and (while holding down the left mouse button) moving the sample-box to the left or right.

- 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' 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.

- 'Create PCG' creates a '.pcg'-file for the currently selected sample. See the notes on

PCG files  
for more info.

- Pressing 'Edit' will open the  
Sample Edit window

.

- 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.

## 1.7 marker window

The marker window lets you add Locators and gives you the possibility to place punch in/out markers. ↔

Everytime the button 'Add Locator' is pressed, a new Locator is added on top of the timeline at the current time. The Locators are blue and are labelled L1 to L9. To jump to a Locator, you can use the keys 1 to 9 on the keyboard (non-numerical part). You can for example use these Locators to mark different sections in a song.

Activating the 'punch in/out' tickmark will show 2 yellow markers on top of the timeline: 'in' and 'out'. These can be used to record a (small) piece to fix playing mistakes or whatever. Set the 'in' marker to where you want to start the recording and the 'out' marker at the place you want to stop the recording. Next time you will record, only the piece between the 'in' and 'out' markers will be stored.

Note that you can set the 'Time' marker before the 'in' marker to get some time to play along with the previously recorded tracks. The actual playback will not stop after the 'out' marker so you can hear if the previously recorded tracks match the just punched-in track.

Hint: You can use the 'Auto Mute' in the

Record window

to mute a

selected track at the time of punch in/out. This way you won't be disturbed by the previously played mistakes.

The

Marker system

help page gives more general information about markers.

## 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 sampled data 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 project. 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. The 'Time' marker on top of the timeline display will move with the movement of the time-slider and vice versa.

At the bottom, a checkbox and a 'Bars' field are present. This can be used in conjunction 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 with the name of the original sample plus '\_eff' and a number. 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.
  
  - 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 Audio Evolution, 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. The 'Normalize' function searches for the loudest sample value and ← computes the gain factor to optimize the dynamics of the sample. This factor is then used to increase the volume of the sample and after computation the gain factor is shown to the user.
  - 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: One-band equalizer. The 'Center Freq' parameter determines which frequency you wish to boost or cut, the 'Gain' determines how much (in dB) and the Q-factor sets the bandwidth (in octaves). The larger the Q-factor, the bigger the frequency range that is affected.
- Ring modulation: A ring modulator adds a kind of 'metallic' sound to the sample.
- Reverse: Reverses the sample or samplepart.
- Reverb: This is only a simple reverb. Use the 'Density' slider to adjust the amount of reverberation.
- Fade In/Out: Lets you fade in/out a sample. The fade-time can be set by the slider. Note that you can also fade in/out samples using the automation events.

## 1.10 mixer

### The Mixer Window

There are two mixer layouts you can use: advanced or normal. You can switch between these two in the

Preferences  
window.

Both layouts have the following items per channel strip:

- Panning
- Mute (M)
- Solo (S)
- Volume slider
- Realtime effect plug-in (3x) if 'Realtime fx' is selected in the Preferences window. See the section on  
Realtime FX
- Realtime effect on/off (3x) if 'Realtime fx' is selected in the Preferences window.

Besides the better look, the advanced mixer layout has these extra items:

- Subgroup 1 - 4: see the section on  
Subgroups
- Volume indication: this meter goes from -infinity to 0dB at the top. Only if the sample has a  
PCG file  
, the volume will be displayed.

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 setting the master volume too high can result in unwanted clipping.

The two bars at the right or the two led strips next to the master volume slider show the left and right output volume. Their update rate and the update rate of the volume indicators in the advanced mixer can be set in the

Preferences  
window.

The

Auto

button brings you to the mixer automation windows where you can automate the whole mixer setup.

Pressing the 'Init' button tells the automation controller to use the current mixing set-up as the initial set-up at time 0:00.

The two arrow buttons at the top right of the window let you go to the following or previous 10 channels.

## 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 or mixes the selected range to another sample
-

at the same timecode. Click  
here  
for more  
information.

## 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, panning, mute, solo and realtime fx.
    - \* Automation events
    - \* Sync Mode
    - \* MonitorVolume
    - \* Recording input
    - \* Audio output
    - \* Tempo
    - \* Initial mixer settings
    - \* Markers
    - \* Master volume
    - \* Subgroups
- 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 AudioEvolution.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 Audio Evolution again.

- \* Save Preferences: Saves the following items in the 'AudioEvolution.prefs' file:

- \* Screen mode
- \* Audio mode
- \* Time Correction Factor
- \* Samplerate
- \* Recording path
- \* Window positions
- \* Every setting in the Preferences window

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

- \* About: Version and author information
- \* Quit: quits the program

## 1.13 windowsmenu

The options in the 'Windows' menu allow you to open or close ↔ windows that are not always necessary.

- \* Marker Window
- \* Subgroups Window

## 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. The MIDI commands will be send via the 'serial.device'.
- \* 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.  
The MIDI commands be received from the 'serial.device'.

- \* B&P Receive: Waits for a start command from Bars&Pipes. You have to add the 'AudioEvolution.ptool' in B&P to a track first. You will also have to start Audio Evolution first before starting Bars&Pipes.  
First press play or record in Audio Evolution, wait for the harddisk activity to finish and then press play in B&P.
- \* CAMD Receive: If this option is selected, AudioEvolution waits for a MIDI start or continue command via the camd.library. In this way, any program using the camd.library can be synchronised to Audio Evolution.

See the section

Using Audio Evolution with MIDI  
for more details on MIDI.

## 1.15 optionsmenu

Options Menu:

- \* MIDI Machine Control: Sends an MMC-start command when the option 'Send' is selected in the  
Sync  
menu.
- \* Volume Overlay: Toggles the visual display of the volume curve (determined by initial volume and automation events) in the  
Time Line Display  
.
- \* MIDI Bar Display: If this option is switched on, the timeline display will show the time in MIDI bars instead of minutes.
- \* Soundcard In/Out: When this option is selected, the  
Soundcard I/O  
window is opened in which you can select ←  
your input  
and output. Note that this menu item is only available when the selected AHI audio mode supports sampling.
- \* Set SampleRate: This opens a window in which you can change the main samplerate. Note that this doesn't write the samplerate 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

- \* Path Configuration: Opens the 'Path Configuration' window in which you can enter the path where recorded samples are stored.
- \* Preferences: Opens the Preferences 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 soundcard io

Soundcard In/Out Window

- \* 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.
- \* Listen: This checkbox allows you to listen to the incoming signal of the selected 'Input'. 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.

## 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 the new Amiga (MCC)
- Porting the program to QNX / Amirage
- New sample edit window

Minor things I have planned in the near future:

- Timeline features like vertical zoom and editing on the timeline
- etc.

## 1.20 problems

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.
- Audio Evolution 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.  
Explanation from Martin Blom:

"There is no difference between sound cards, as you say, but there is a difference between different modes: The fast modes use large tables, which are recalculated on each master volume change. That's why those modes take more time to change the master volume."

Solution: Use a non-fast mode!

- Delfina Lite has an incorrect sync with the 'any freq' modes

Solution: Use another mode for now, the bug is again in the AHI-driver. It's being fixed by Petsoff.

- "I'm recording with a '060 and a Delfina Lite and I can't get many tracks!"
-

Solution: Upgrade to the Delfina Plus, the Delfina Lite has a slow bus interface.

## 1.21 contact

If you have any questions, suggestions, criticism, etc., please feel free to contact me at the following address:

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If you would like to order the full version of Audio Evolution, please contact Computer City at [info@compcity.nl](mailto:info@compcity.nl) .

## 1.22 acknowledgements

Many thanks to :

- Misja van Waterschoot for testing Audio Evolution and giving tips!
- Martin Blom for his wonderful AHI-package.
- Roger Angus for his many suggestions and testing on Audio Evolution.
- Niko Tomatsidis for his patience, testing and support.
- Computer City for offering to distribute Audio Evolution.
  
- And last but not least:

Thanks to all the people that responded to the demo-releases of Audio Evolution and even more thanks to the people who bought Audio Evolution! You people are wonderful! :-)

## 1.23 demo limitations

### DEMO LIMITATIONS

The demo version is limited to only 3 (stereo) tracks and 60 seconds of playback- and record time. The full version has an unlimited number of tracks and unlimited playback- and recordtime. For obtaining the full version, see below.

### PRICING

The price is around 189 Dutch guilders, which is around 91 USD or 86 Euro. For exact prices, please

contact Computer City

.

#### ORDERING

Please check that the demo-version is fully working on the system and soundcard that you're going to run it on. If you're satisfied with it and you want to order, you can

contact Computer City

.

They are distributing Audio Evolution now and are handling all orders. Audio Evolution now also comes on CD instead of being send by email.

## 1.24 recordwindow

### The Record Window

In this window you can start the recording or verify your incoming signal. With the gain-control and peak meters you can get the best signal-to-noise ratio without clipping.

- \* Record button: If you press this button, the program will start recording from the input you have selected at the SoundCard In/Out window. If you have selected a sync receive option in the Sync Menu, the program will first wait for a (MIDI) start command before starting the recording.
  - \* Stop button: Stops the recording.
  - \* HD-light: To check whether the audio data is really written to disk, a harddisk-led is present here. When recording, it flashes red when something is written to disk. If you pressed 'Record' and nothing happens within a few seconds, something probably went wrong. You should stop the recording then and try again.
  - \* Mon: (Monitor) This determines how loud you will hear the incoming signal.
  - \* Gain: Sets the input level in dB. This option only works when your sampler has variable input gain.
  - \* Output: Adjusts the overall output level. This option only works when your sampler has a variable output volume.
  - \* Listen (at the left of the record level led bars): This checkbox allows you to listen to the incoming signal before and while recording. Adjusting the potmeters is only possible before recording. If this gadget is active, the two led bars will show the recording level. The highest incoming level (peak) will be kept and displayed at the right of the led bars. Make sure that your loudest recording signal stays below 0 dB in order to prevent ugly digital distortion.
  - \* Peak reset: Resets the peak indicators to -96 dB.
-

- \* Auto Mute: Enabling this option mutes the selected track in the timeline window at the time of punch in/out. This way you won't be disturbed by the possibly previously played mistakes.

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.

Selecting an existing sample will pop up a requester asking you if you want to replace the sample.

You can also choose to create a

PCG file

or choose to only keep the left

or right channel.

## 1.25 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.

Hint: You can also edit automation events directly on the timeline.

For more information click

here

.

## 1.26 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.

## 1.27 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.
- \* Get tempo from range: This button calculates the tempo from a range you selected in the Sample Edit window : select a range in a sample that is 4 beats long, go to the MIDI Settings Window and ↔ press this button. You now got the tempo from the sample! Note: this only works for 4/4 at the moment.
- \* Time Correction Factor: For some reason, timing in MIDI sequencers is not 100% ↔ correct. So if you use Audio Evolution 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 vice versa. Repeat this for 2,4,8 and 12 minutes.

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

## 1.28 using ae with midi

Audio Evolution 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 conjunction 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 occur in the middle of a recording.

## 1.29 notes about ahi modes

### Notes about AHI modes

It's important to use an AHI 'Fast' and 'realtime' mode for recording, for example 'Fast 16 bit stereo++ realtime'. A non-fast mode can overload the cpu too much, causing missed sample blocks. The AHI-mode has no influence on the sound quality of the recorded sample: whether the mode is a Hifi or a Fast mode, the same samples get written to disk.

It's also important to use a realtime mode, otherwise the played and recorded tracks can be out of sync. This is also necessary for using Audio Evolution with MIDI synchronisation.

Using an AHI stereo mode without '++' causes the odd channel numbers to be played on the left side and the even channels on the right side. There's no panning available in this mode.

### 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 sample values 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.

Audio Evolution 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. 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.

## 1.30 pcg notes

Notes on PCG (Pre-Computed Graphics) files

PCG's are little files that hold graphical information on how a sample looks like. They are computed and stored at a given zoomfactor, for example 256 pixels per second. In this way, a sample can be shown quickly at different zoom levels using the PCG, because the graphical data is already in memory and doesn't have to be computed from the actual sample each time you zoom in or out. The size of these files are around 1/650th of the real samplesize for stereo samples at 44.1 Khz.

The PCG's are currently only used for the timeline display, but will also be used in the sample edit window in the future.

Samples with and without PCG's can be used together.

## 1.31 copy to track

With 'Copy To Track' you can copy a sample or a sample-range to another sample.

Choose the destination sample first and select one of the following items:

- **Overwrite:** Simply overwrites the destination sample (or range) with the source sample.
  - **Mix:** Here the source and destination samples are mixed with the given percentages.
  - **Cross-fade:** With cross-fade, you can fade the source sample in with the time selected by the 'fade time' slider. If the tickmark for 'Fade End' is selected, the sample will be faded out at the end too. This is great in combination with the 'punch in/out' option to correct for playing mistakes. A good setting for the fade time for correcting mistakes is usually around 150 milliseconds.
-

## 1.32 editing events on timeline

Besides entering automation events with timecodes you can also add and edit automation events on the timeline display itself.

By pressing the right mouse button on a samplebox in the timeline window, a popup-menu appears with an item 'Automation'. This leads to a submenu in which you can choose between 'Mute' and 'Volume'. Selecting one of the two brings you in the automation event 'Add' mode. The corresponding automation events are graphically displayed for this sample.

Pressing the right mouse button again in automation editing mode pops up another menu with the items 'Add', 'Move', 'Delete' and 'Finish'. The first three are self-explanatory. 'Finish' means that you want to leave the automation edit mode.

**Mute:** When in 'Add' mode, click with the left mouse button where you want the mute to start on the sample's samplebox. This will add a mute on and off event, depicted by the speaker symbols and the grey part. After adding, the edit mode automatically jumps to 'Move' mode which allows you to adjust the start- and stoptime of the event. You can do this by clicking with the left mouse button at the start or end of the mute and (while holding down the left mouse button) dragging the black line that appears to the place you want.

**Volume:** Almost the same as 'Mute'. The only difference is that now a red line appears with two handles which depicts the volume-change. The bottom of the sample-box corresponds with a volume of 0%, and the top corresponds with 100%. In this way you can enter either a fade-in or a fade-out.

Don't forget to select 'Finish' if you want to stop editing automation events.

## 1.33 marker system

The Marker system

At the top of the timeline (where you can see the timecodes), different kinds of markers can be edited. Markers are small flags with a specific color which can be dragged along the timeline.

The following markers exist:

- Locators - Blue: Locators can mark a specific location in a song or are handy when you want to record or play from specific locations. More information on the Locators can be found

here

- Time marker - Green: The Time marker is used to set the current time. The timeslider in the Control window will move with the adjustment of this marker and vice versa.
- Punch In/Out - Yellow: Allows you to set the punch in/out locations for recording. See the Marker Window help for more information on this subject.

Using the markers:

You can move a marker by clicking on it with the left mouse button and while holding it down, move it with your mousepointer. A black vertical line will appear under it so you can accurately position the marker. This is of course only useful when in 'Extended Timeline' mode.

When double-clicking the marker, a window will pop up showing the exact timecode of this marker and whether it's locked or not. You can change the timecode if you want or alter the 'locked' state. When a marker has been locked, four white dots will appear at the corners of the marker. In this state, the marker can't be dragged anymore. This is useful if you wish to prevent accidental movement of a marker.

In this window you can also delete the marker. Note that at the moment you delete a Locator marker, the locator numbers are rearranged.

## 1.34 preferences window

The Preferences Window

The Preferences window is subdivided into three classes:

- Advanced: Contains several advanced options:

- \* Soundcard Delay Correction: Here you can enter the specific hardware delay of your soundcard. If you feel that recorded samples are a bit out of sync, than you can try to fix this by finding this delay correction time.

You can find the value for your soundcard/system by connecting your output directly to your input and measuring the delay by doing the following:

Load a sample with a samplerate you're most likely to use. You need a sample with a little piece of silence at the beginning. Set the monitor volume completely to the left in the Record window (like -96dB) to prevent loops. Now start recording for a few seconds and stop. Open the first (original) sample in the Edit window and zoom in a lot. Now try

to find the start of where sound really gets played (that's why we had to pick a sample with a bit of silence at the start...) and write this timecode down.

Do the same with the second (recorded) sample and determine the difference. This will be likely anywhere between 0 and 30 msec. You can enter this value here as the Soundcard Delay Correction time.

- \* Volume Indicator Update Speed: This slider controls the speed at which the volume displays in the mixer window will be updated. Faster settings will cost more cpu-time.
- \* Record Indicator Update Speed: This slider controls the speed at which the record level displays in the Record window will be updated. Faster settings will cost more cpu-time.

- Mixer: Contains mixer window related items:

- \* Advanced Mixer: Switches between a nice gui for the mixer window with volume display per channel and subgroups or the old simple look.
- \* Realtime Fx: Turns the usage of realtime effects on or off. This also makes the mixer window larger or smaller by removing or adding gadgets for the realtime effects.
- \* Master Volume Display: Turns the master volume indicator in the mixer window on or off.
- \* Channel Volume Display: Turns the channel volume indicators in the mixer window on or off.
- \* Automation Display: Turns the visual display of the automation events in the mixer window on or off.

- Timeline: Contains timeline window related items:

- \* 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.

## 1.35 add workspace

Add Workspace Window

When applying effects such as delay, it may be needed to add some extra workspace to a sample in order to prevent the delay being cut off. In this window you can specify how much workspace you want to add at the end of the sample you have selected. After pressing 'Use', the sample length will be updated and a pcg-file will possibly be created again.

## 1.36 realtime fx

### Realtime Effects

As realtime effects require a lot of cpu time, a PowerPC processor is highly recommended. If you don't have a PowerPC processor, at least a '040 is required. In order to prevent audio hick-ups, Audio Evolution will warn you when a cpu overload occurs and will stop playback or recording.

In order to use realtime effects, enable the option 'Realtime Fx' in the

#### Preferences

window first. Effects can only be plugged in into a channel which has a sample loaded.

At the top of the mixer window, there are three buttons and three leds per channel. The buttons depict the effect that are plugged in and the leds are on/off switches for the effect.

If a button is empty/grey, pressing the button will pop up a window in which you can select the effect you want to plug in. Effects are grouped per kind of effect. After selecting an effect, it's corresponding color and abbreviation will be displayed in the button. The red led right next to the button will light up, which means its state is 'on'. You can toggle the effect on or off by clicking on the led, even while playing.

When the effect is plugged in, selecting the button again will open the parameter window in which you can set the parameters for the chosen effect. The triangular buttons let you change the parameters with great accuracy, while the slider gives you quick access to the parameter which button is selected under the parameter display box. Here you can also remove or change the effect if you like.

The following runtime possibilities are present while playing. As the effects are processed in blocks, the changes are not performed real-time, but with a small delay:

- \* Parameter change: you can open the parameter window while playing and change the parameters on the fly. Parameters which are grey during playback can only be changed before playing because they use look-up tables.
- \* On/Off state switch: pressing the red led will turn the corresponding effect on or off while playing.
- \* Removal of effects

Hint: PPC realtime effects can be downloaded for free from my website.

## 1.37 subgroups

Subgroups window

The subgroups window lets you create a sub-mix. This is very handy for dealing with drums etcetera: if you have for example 6 channels for drums (bassdrum, snare, hihat, etc.) and you want the drums to be louder in total (all their channels), you can assign those channel to the same subgroup and adjust their overall volume with the corresponding subgroup slider. In this way it's also easy to hear the drums alone (by selecting solo above the subgroup slider) or mute them.

You can assign a channel to a subgroup by selecting one of the four blue buttons at the right side of a channel strip (you can only assign a channel to one subgroup). By selecting the same group on other channels, these channels will belong to the same group and can be controlled by the corresponding subgroup channel strip in the subgroups window.

Subgroup volumes and mutes can also be automated.

## 1.38 keys

The following keys are recognized and can be used in most windows.

p : Starts playback

s : Stops playback

b : Resets time to 0:00

q : Quits the program

Up arrow : Selects the previous item in the sample list in the timeline window

Down arrow : Selects the next item in the sample list in the timeline window

l - 0 (non numerical) : Jumps to the corresponding locator

F1 : Moves timeline window to front

F2 : Moves mixer window to front

F3 : Moves control window to front

F4 : Moves effectlist window to front

F5 : Opens marker window if necessary and moves it to the front

F6 : Opens subgroups window if necessary and moves it to the front

Keys on the keypad:

4 : Forward one second

5 : Start or stop playback

6 : Backward one second

7 : Resets time to 0:00

Enter : Opens the timecode window

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