AudioTracker

8 Track Harddiskrecording System

Manual

Audio Tracker Version 1.6

For Atari Falcon

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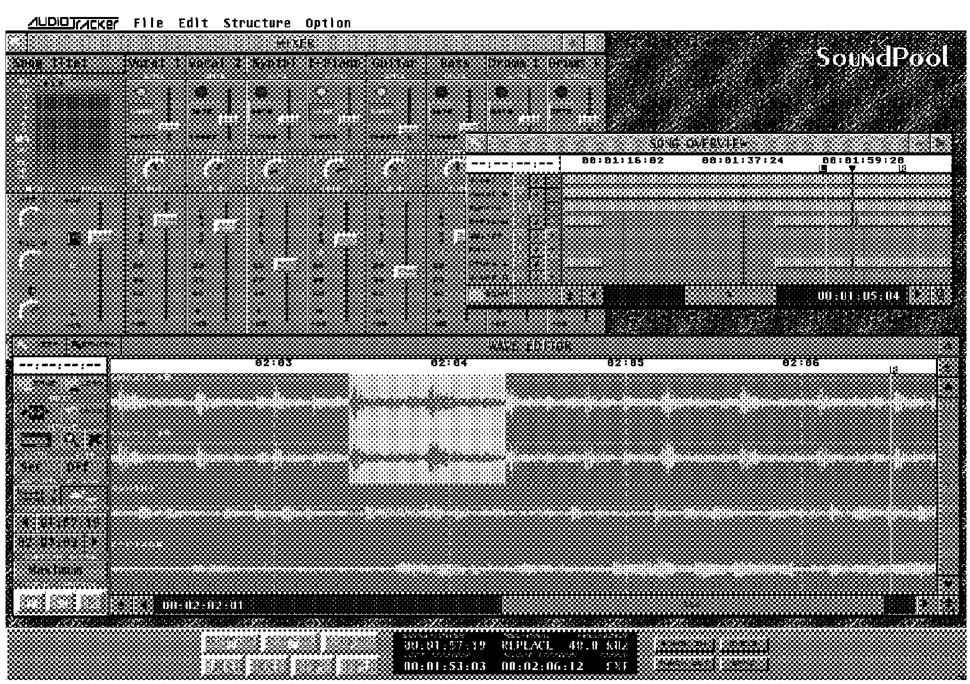

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AudioTracker - Introduction

1.1 Introduction

AudioTracker is a hard disk recording system which has been developed especially for the Atari Falcon 030 computer. It makes full use of the Falcon's audio sub-system and digital signal processor (DSP). The program can run on a Falcon with minimum of additional hardware (an internal hard disk is assumed to be present) - professional work in CD/DAT quality, however, requires that the Falcon's relatively poor AD-DA converter be circumvented by connecting an S/PDIF interface to the DSP port (see the chapter "Installation"). This would prevent loss during data transfer with CD players or DAT recorders and also allows sample frequencies of 44.1 and 48.0 kHz which are standard in high-end equipment.

The hard disk

Hard disk recording demands high performance from the connected hard disk: In order to work comfortably with eight tracks at the highest sampling frequency, the data transfer rate should not fall short of 1.4 MB per second.

Direct to Disk

The program is tape-orientated, meaning that operation is similar to an eight track tape recorder. There is no complicated data handling - one or more partitions which have been declared to be "tape" can be recorded to directly. Unlike real tape recorders, the number of tracks can vary from song to song.

Editing

Apart from standard features such as accurate locator points for Punch In/Out (precision is to the nearest sample!) or record and playback in cycle mode, there are numerous functions used for editing the recorded material afterwards. These can be divided roughly into two types: structural operations for extracting, copying or moving sections of music which manipulate the data on the hard disk directly (destructive editing) and precision work using the wave editor's graphic display, whereby you can decide whether any changes should become permanent at a later point in time (non-destructive editing). Up to four tracks can be edited simultaneously. The editor offers block functions which can even be called during playback and also non-destructive recording in cycle/overdub mode.

Mixing

Productions can be mixed down either by using the integrated mixer or, by way of the Falcon Analog-8 interface ("FA-8") or Adat Interface, using an external mixing desk. AudioTracker can be synchronized to sequencers or other time-dependant equipment using standard MIDI Time Code or MIDI Clock - several Falcons can be linked to form a theoretically unlimited multitrack system via a special internal synchronization protocol.

Backup and archives

Export and import functions allow data to be transferred to and from other audio programs - an integrated streamer function caters for backups and archiving single partitions to DAT.

Installation and Program Start

2.1 Hardware Configuration

Safety copy / Running from hard disk

The AudioTracker floppy disk is not copy-protected - you should make a safety copy of the disk before you do anything else. If you want to start the program from your hard disk you should create a new folder (e.g. A_TRACK) and copy all files into it (from the safety disk of course!).

SCSI driver

Users of SCSI disks should copy the driver "SHDRIVER.SYS" into the root directory of the boot drive (usually C:\). The existing driver will be overwritten (quit the alert box with "Copy"). The original SHDRIVER.SYS may be write protected - in this case select the file and delete its read-only attribute via the menu File/Info first. Only the AudioTracker version of this driver lets you work with eight tracks properly. It will work as a standard AHDI driver for other programs - the special functions are only activated by AudioTracker.

IDE driver

If you work with an IDE drive, the kind of Harddisk driver is not important, because AudioTracker has special code implemented to use IDE-drives in the fastest possible way.

Program font / NVDI

AudioTracker cannot install its own font if you are running NVDI. This is no big problem, but for the sake of completeness you should know how to install the program under NVDI because the font engine (which is not often used by other programs) contains special characters which make the program look a lot tidier.

Copy the font "A_TRACK.FNT" from the folder "A_TRACK.DAT" to the directory which contains the NVDI driver "NVDIDRVx.SYS". As a rule this should be the "GEMSYS" folder. Further action differs according to whether the file "ASSIGN.SYS" is present or not:

a) Installation when "ASSIGN.SYS" is present

Extend the file "ASSIGN.SYS" by writing the name of the new font to it, by using the NVDI tool "ASSIGN.PRG". Any ASCII text editor can be used instead - you should enter "s A_TRACK.FNT" into the list under "05p SCREEN.SYS" and save the file.

b) Installation when "ASSIGN.SYS" is not present

The program disk includes an "ASSIGN.SYS" file which already contains all the necessary information to run AudioTracker, but assumes that the above-mentioned NVDI driver is in the "GEMSYS" folder - if not you should set the path accordingly by loading it into an ASCII editor and changing the "PATH =" entry.

Monitor / screen resolution

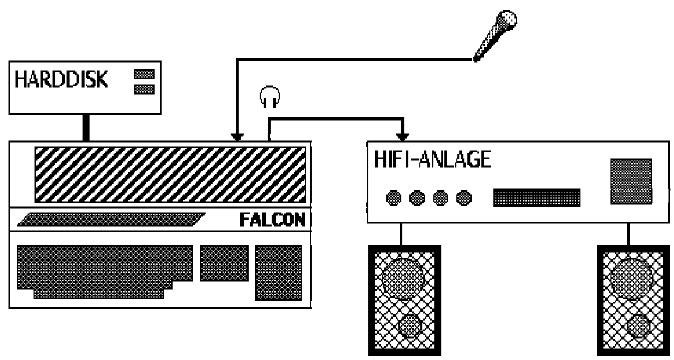
AudioTracker will run at all screen resolutions from 640 x 400 upward except True Color.

Copy Protection - the key

AudioTracker is protected from illegal use with the aid of a hardware key or "dongle", which should be inserted into the ROM port of your Falcon (to the right next to the MIDI sockets) with the marked side visible.

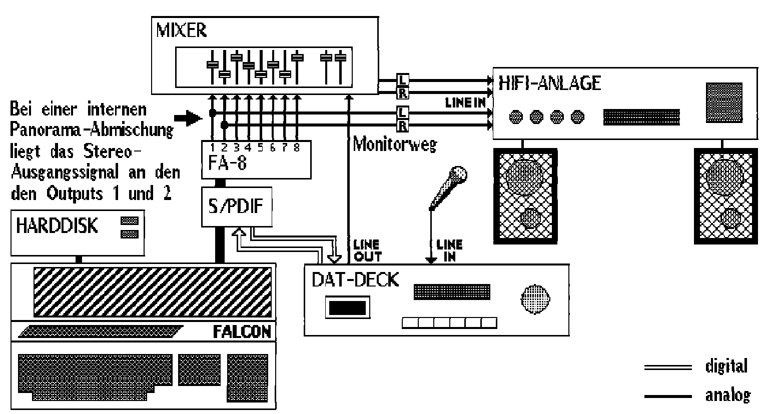
Hardware configuration

Operation without a Digital Interface



If you own the special digital interface (either the Soundpool one from your AudioTracker dealer or an original Steinberg) you can skip this section. Connect the Falcon’s headphone socket to your stereo system - you will usually need a "stereo mini-Jack to double Cinch" cable, or plug headphones directly into the Falcon. If you want to record using a microphone you should connect this directly to the Falcon. Low-impedance analog sources such as tape decks, CD players or most keyboards can be adapted by connecting a 200 Ohm resistor in series with the signal line (in two special Jack leads would be a good place if you are handy with a soldering iron) - ask your dealer if in doubt. Switch on all connected equipment and boot the Falcon.

Operation with S/PDIF and FA-8 interface



As already mentioned in the first chapter, this configuration (or similar) is necessary if you want professional quality sound. The cost of the additional hardware is relatively low if you compare it with the price of dedicated hard disk recording systems. A digital audio tape deck (e.g. DAT recorder) is necessary for the following reasons:

- Analog recordings are converted to digital with the high quality AD converter in the DAT recorder and sent without loss to the Falcon. The sample frequency is dictated by the DAT (usually 48 kHz). You can also record directly with 44.1 or 32.0 kHz using Soundpool's new Sample Rate Converter.

- The finished pieces can be mixed to stereo using the integrated mixer and can then be digitally mastered to DAT tape without any loss of quality whatsoever. Subsequent editing (e.g. with "AudioMaster") will not impair the sound at all as long as the "digital chain" is not interrupted.

- The system can also be used as a comfortable data streamer - digital data is not restricted to audio.

The S/PDIF interface is connected to the ANALOG-8 interface via a short ribbon cable and also to the Falcon's DSP port with a special cord which is included in the package.

The S/PDIF actually has two interfaces, coaxial and optical. Coaxial connections need well shielded Cinch cables - optical connections are made using special cables which are available from better Hi-Fi dealers. The second method is preferred as, apart from better quality sound, loss of data during use as a streamer is avoided. Connect the S/PDIF digital input to the DAT or CD digital output and the S/PDIF digital output to the DAT's digital input. Turn on the power to the FA-8 and connect either all the analog outputs (1-8) to a mixing desk or channels 1 and 2 to the left and right line-in sockets of your Hi-Fi system.

2.2 Starting the Program

Before you start the program you should decide on the partition of your hard disk which you want to reserve for audio data. The chosen partition should not be too small - approximately 5 MB is used per minute for each track sampled at 44.1 kHz. A ten-minute song with eight tracks sampled at 44.1 kHz requires 400 MB (440 MB at 48 kHz) of valuable hard disk memory!

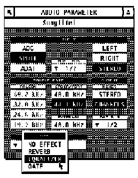
If you own a 1 or 2 Gigabyte hard disk you can start thinking about separating it into two or more partitions for the sake of clarity. Each "tape" can accomodate up to 16 songs. If most of your work is comprised of short jingles, partitioning such a large hard disk can be very sensible.

Start the program by double-clicking on "A_TRACK.PRG". A new desktop with the program logo, menu and tape transport pad will appear after a few moments. The transport pad includes a display showing the song position and locator positions as SMPTE times (hours, minutes, seconds and frames). The current sample frequency and source are displayed to the right. Only the Falcon's internal frequencies are shown (e.g. "49.2 kHz / INT") if the interface is not connected. Otherwise the sample frequency of the external source device (48.0, 44.1 or 32.0 kHz, "EXT") appears. If there is a period in front of the word "EXT" this means that the external device is sending at 32.0 kHz which cannot be generated by the S/PDIF. In this case the program will switch automatically into the (otherwise optional) external clock mode - more on this subject in the next section.

To the left of the display are the tape controls, the function keys "Punch In", "Punch Out", "Cycle" and "Sync". At the far right you will find a clipboard used for temporarily saving any musical snippets.

2.3 Software Adaptation

Audio Parameters Window



Select "Audio Parameters" from the Options menu to adapt the program to your hardware configuration. Most of the parameters are automatically preset to sensible values when the program is loaded - the program recognizes whether the S/PDIF interface is connected and also what frequency the digital input is receiving. A window appears in which the following parameters can be set. These settings are loaded when a new song is created, but can of course be changed at any time. The current settings are saved automatically when the song or the program is removed from memory.

Input

You can only record either via the Falcon’s own AD converter (microphone input) or by way of the digital interface S/PDIF or ADAT. Click on the corresponding button. If the S/PDIF interface is not connected you will not get a choice - "ADC" will be selected by default and the other buttons will be disabled (greyed). If you want to record via the ADAT-8- Interface, you may select either one of four possible stereo pairs or all channels 1-8 by the pop up menu "ADAT-BUS". There are two buttons beside these with which the input channel can be selected. Two neighbouring channels will be recorded to in STEREO mode i.e. tracks 1+2, tracks 3+4 etc. Choose the mono channel to which your instrument or microphone is connected if you wish to record single tracks.

Output

The mixer is able to send the output signal in two basically different ways. STEREO mode lets you to mix all tracks to a stereo sum using the integrated mixer which can then, for instance, be digitally mastered to a DAT recorder. If an FA-8 is connected, the right channel will appear on output 1 and the left channel on output 2. The "CHANNELS" setting causes the output of all tracks to be routed to the corresponding analog outputs of the FA-8 so that they can be mixed using an external mixing console (the panorama control would then be missing from the internal mixer). This mode allows you to assign one stereo pair to the D-A- Converter of the Falcon (headphone socket) by the pop up menu "DAC-BUS". So it is possible to lead out six resp. four single channels without using the FA-8 or ADAT-Interface.

Sample Rate

Either the Falcon’s internal frequencies or those from the S/PDIF (or other connected clock) are available, depending upon the output setting. The S/PDIF offers 44.1 kHz and 48.0 kHz. If you want to synchronize the system to an external clock you should select the button labelled "EXTERNAL CLOCK". The button will always display the most recently measured clock frequency. Measurement occurs when the program starts , after digital recording and also after playback. Digital recordings via S/PDIF will always run at the speed of the source device and will not be influenced by the chosen sample rate. You will find more information on this subject and its inherent problems in the chapter "Digital Recording".

Effects

The DSP (Digital Signal Processor) is an independent CPU that allows to work on the audio signals in realtime. This unit calculates the level and panorama position and is able to process some effects. The current program version includes a reverb module, a Dual 9-Band-EQ and a gate. Select the desired module out of the pop up menu "DSP". One or two of the effect parameters can be controlled in the mixer window - for editing the basic parameters you should use the effect editor window.

The effects can also be processed "offline" (refer chapter 10.2).

The First Recording

3.1 Brief Tutorial

It is a good idea to be introduced to the general flavour of the recording procedure without having to wade through a list of all the available functions - this chapter covers enough ground so that you can make something audible within a few minutes. If you are not in such a hurry and would like to study the concept and operation in depth you can skip this chapter and read the rest of this manual (where all functions are explained more fully).

Installing a "tape"

A "tape" must be installed first. Click on "Install Tape" in the "File" menu and then select the drive/partition you want to use for HD-recording. All registered hard disk partitions are selectable.

The minimum amount of a tape file should be 4000 kB!

Quit the dialog box with [OK]. The [Cancel] button is the default (i.e. can be activated by pressing RETURN or ENTER) for safety’s sake. If the partition already contains data you will be warned with an alert box. Any such data will be erased as soon as you confirm with [OK]. The message "Tape installed" will appear after a few moments: the message box will disappear automatically after a few seconds or when you move the mouse.

The first song

You can set up your first song. Select "New Song" from the "File" menu. A dialog box will open in which you can set the following:

- Song Title: Enter a name for the song and hit ENTER.
- Tracks: Enter the number of tracks
- Time: Enter the length of the song. The maximum length possible is already entered by default. You should reduce this by about 10 seconds to leave enough room in memory for the clipboard functions. Quit the dialog either by hitting RETURN or by clicking on [OK].

The Mixer window will be opened. To the left is the master section, to the right the mixer module. You can now start the recording.

Recording

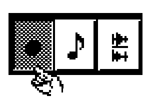
Each track corresponds to a module in the mixer. Select a track to be recorded to by clicking on one of the buttons below the faders. Click on the record button in the tape transport pad. Play something on the connected instrument or start the tape/CD you want to record. The peak meter should average about -12 dB. Purely digital recording will not allow you to manipulate the level - if an microphone/instrument is connected to the analog input of a DAT recorder you can control the level there (DAT in RECORD/PAUSE mode). The default value is "MONO/Left".

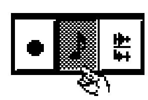
If you want to record in stereo or just the righthand channel, click on the stop key (to the left). Press Control + A on the keyboard or select "Audio Parameters" from the Options menu. A window will open in which you can select the required input mode. You can now select the track(s) you want to record to (stereo mode always records to pairs of adjacent tracks). Start the recording by clicking on the Record button or by hitting the "*" key on the numeric pad of your keyboard.

Set the faders in the mixer and the master fader to the required levels. If you are recording stereo you should set the pan controls to the extreme right and left. To finish recording click on the Stop button (pressing the spacebar or 0 on the numeric keypad will have the same effect). If you don't actively stop the recording it will terminate automatically at the end of the period set in the dialog box you opened with "New Song".

Playback

Using the Play buttons, select the track(s) you have just recorded and rewind to the beginning by clicking on the Rewind button or by pressing Contol + (on the numeric keypad. Click on the Play button or press ENTER to start playback.





Record-Button and Play-Button

Musical Structures

4.1 A Few Definitions

Tape

A "tape" is the basic structure and corresponds to a tape file on the hard disk. The decision of whether you want to work with a single large partition or several smaller ones is left up to you. Tapes are installed by selecting "Install Tape" from the File menu and is the first action to be taken when setting up the system.

Song

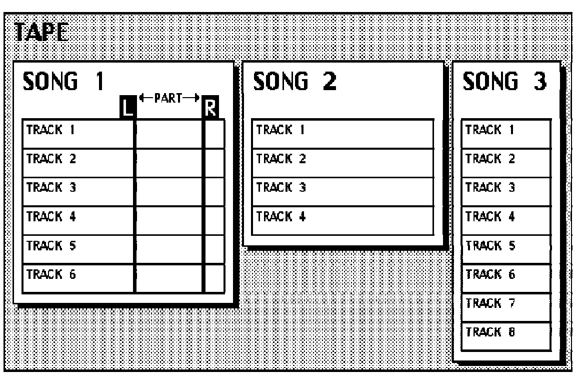
A tape can accomodate a maximum of 16 "songs". New songs are are declared via "New Song" in the File menu - you will need to set the length and number of tracks. "Open Song" activates a previously recorded song for further editing. Songs are handled in a similar way to standard files although they are not GEMDOS structures. Songs cannot be loaded as standard files by other programs and cannot be deleted or copied by normal means. Such actions are carried out by dedicated functions which will be described in later chapters.

Track

A song consists of two to eight "tracks". Each track has a corresponding mixer module in AudioTracker’s main window, which is used for setting volume level, panorama, effect-balance and record/playback. The "Structure" Menu contains functions for copying and deleting single or multiple tracks. The number of tracks in a song can be changed at any time.

Part

A "part" is a musical passage of a song as defined by two locator points - it is not a subdivision of a track but of the entire song across all tracks. Nearly all structural operations such as Delete, Cut, Copy, Paste, Punch-In recording, Cycle-Mode recording manipulate "parts". You can set the tracks to be included in the operation with the Play/Record buttons.



A tape with 3 songs of different length and number of tracks

SMPTE time code

The SMPTE standard (EBU) time code has, in addition to the usual time intervals hours, minutes and seconds, a further subdivision of the second into 25 so-called "Frames". Almost all the editable fields and displays in AudioTracker are concerned with SMTE time.

Samples

Samples are the smallest musical units which make up a sound. The Wave Editor allows access to single samples and lets you position the songpointer or locators to single samples. The number of samples per second is called the sample frequency or rate - a sample frequency of 48 kHz means 48000 samples per second.

4.2 Creating and Manipulating Structures

Installing a tape

AudioTracker is only ready to record for the first time after a "tape file" has been installed on the hard disk. This only has to be done once except if you want to use this function to delete all the songs in the file.

Click on "Install Tape" in the File menu. A dialog box will open which contains a row of buttons from "C" to "P". All drives which are registered in the operating system are selectable, all others are greyed. A drive may already selected, but if you wish to change this you should select the right drive by clicking on the appropriate button.

There is an editable field below the drive letters in which you can write a name - this is usually used to name backups made using the streamer function. Quit the dialog box with [OK] after making sure that you have selected the correct partition. AudioTracker creates a file with the name "A_TRACK.TPE".

Remember: If you attempt to load, move, copy or edit the file "A_TRACK.TPE" in any way from the desktop or from within other programs and you get trouble, use a defragmentation tool on the tapefile.

All the dialog boxes in the program are "flying dialogs" i.e. they can be moved freely across the screen. Click on the blue title bar and drag the dialog to the new position while keeping the mousebutton pressed. The current position of each dialog box is saved when it is closed, to be restored the next time that particular function is called.

New song

Clicking on "New Song" in the File menu will open a dialog box in which you can set a few parameters to define the song:

- Firstly, select the tape using the drive buttons. If you are only working with a single tape this partition will already be selected. Only partitions which have been installed as tape will be selectable - all others are greyed.

- Down left the data transfer rate of your harddisk drive is displayed. For full eight track performance this value should be higher than 1300 KB/s.

Tip: If you need more speed change to the monochrome display mode.

- By default, the text cursor is in the editable field in which you can write the title/name of the song. Enter a descriptive name and hit ENTER.

- Select the number of tracks you will need for the song. The field next to the track buttons shows the approximate free capacity of the tape, displayed as time.

- Enter the length of the song in hours, minutes and seconds (hh:mm:ss.

The [OK] button becomes the default after all the parameters have been entered - hit RETURN again or click on [OK] to quit the dialog.

AudioTracker's main window is opened, containing a mixing console whereby each mixer module represents a track. If your song has eight tracks they can only be displayed simultaneously if the resolution of your monitor is at least 640 pixels wide. The mixer will be described fully in the chapter "The Integrated Mixer".

Opening a song

A song can be opened for editing by clicking on "Open Song" in the File menu. A dialog box similar to the standard file selector will appear. To the right is a row of drive buttons from which you can select the tape. All the names of songs on the selected tape will be listed together with information about the number of tracks, the sample frequency and the size of the file expressed in hours, minutes and seconds. Select the required song and quit the dialog with [OK] - double-clicking the song will have the same effect.

If a song is already active, all its windows will be closed first. All important settings will be saved automatically so that the old song can be restored later. The main window of the currently selected song will then be opened.

Closing a song

Except for one particular reason (see "Deleting a Song") you should never need to close a song - AudioTracker will do this automatically whenever it becomes necessary. Songs are closed when their main windows are closed. Alternatively, hitting ESCAPE will also close the window. Closing the main window of a song will of course close any other windows belonging to that song.

Deleting a song

The function "Delete Song" lets you erase the song from the tape. Not only is the music deleted, but any following songs will be repositioned so that all the free memory on the partition will be in a single block at the end of the tape. This may take a while, depending upon how much data has to be moved. Deleting the last song on tape is immediate. If a song other than the currently active one is to be deleted, you should close the main window before you call this function. The song selector will be opened (as if you had called "Open Song").

You will always be warned before a song is finally deleted so that you have the chance to cancel the action. A horizontal bar similar to that seen while formatting diskettes displays progress so that you can estimate the time required.

Changing song structure

The structure of the currently active song i.e. the length and the number of tracks can be changed at any time using the function "Song Structure" found in the Structure menu. The dialog seen when calling "New Song" is opened in which you can change the parameters. The field showing the length of the song will change according to the number of tracks you enter. When you are satisfied with your entries, quit the dialog box with [OK]. Data may have to be copied/moved according to the changes you have made and depending upon the position of the song on the tape (see above). All windows will then be closed. After all the necessary actions have been carried out, the main window will appear again - if you have changed the number of tracks the number of mixer modules will be changed accordingly.

Musical structure operations on tracks and parts are explained in the chapter "The Structure Menu".

The Tape Transport Pad

5.1 Tape Control

The left side of the pad contains buttons used for controlling tape transport. At the top are STOP, RECORD and PLAY, below these are buttons for fast-rewind and fast-forward. The outer buttons will set the song pointer either to the next locator in that direction or to the beginning or end of the song. The inner buttons move the "tape" either in steps of seconds (with the left mouse button) or frames (with the right mouse button).

Playback / Stop

Playback is started either by clicking on the PLAY button or by hitting ENTER/RETURN on the keyboard. If the song pointer is already at the end of the song it will jump to the start automatically (not during RECORD). Only those tracks whose PLAY buttons are selected will be played back (see "The Integrated Mixer"). Playback is stopped either by clicking on the STOP button or by pressing the spacebar (or 0 on the numeric keypad). The song will also stop when it reaches the end, of course.

Note: If your hard disk is relatively slow , you might be warned of the fact with an alert box ("Maybe HD too slow!"). One reason can be that you have selected too many tracks for playback. If you ignore the message you run the risk of having playback interrupted abruptly in the middle of the song ("HD too slow!").

If one of the above messages appears although you are running a SCSI disk you are either using the wrong HD driver ("SHDRIVER.SYS") or you have chosen a horizontal resolution of more than 640 pixels in 16 colour mode (cf. the chapter "Installation and Program Start").

Song position

There are other ways of changing the song pointer other than with the transport controls:

- Double clicking the song position display opens a small field in which a new position can be entered in SMPTE format. Confirm either by hitting RETURN or by clicking within the field.

- A single click on the display calls a popup list containing up to ten user-definable cue positions. These can also be selected by hitting the function keys together with SHIFT. The current song position can be saved to one of these cues by holding down CONTROL whilst selecting the cue. For instance, you can press CONTROL + SHIFT + F4 to save the current song position to cue number 4. The cue list belongs to the parameters which are saved when the song or the program is terminated (see Appendix).

- When the Wave Editor is open the song pointer can be placed within the sample display by CONTROL + click in the position bar (see the chapter "The Wave Editor: The Song Pointer and Locators").

- Using the keys "[" and "]" the songpointer will be placed to the position of the left resp. right locator.

Left and right locators

The two locators are flexible cues used for delimiting a musical passage. Such a passage is called a "Part" in AudioTracker . There are functions in the Structure menu for copying, moving and deleting parts. The positions of the left and right locators define the start and end of playback/record in Cycle mode and also the Punch-In and Punch-Out positions during normal record. The locators can also be positioned in several different ways:

- By entering the position in SMPTE format after having double-clicked on the corresponding display.
- There are ten user-definable locator-pairs which can be defined and restored in a similar way to the cues (see above) but without pressing SHIFT.
- Locators can be set during playback by hitting the left or right SHIFT key. AudioTracker confirms this with the message "SAVED!" in the locator display. The exact position will be calculated and displayed after playback is stopped.
- The locators can also be positioned in the Wave Editor by clicking on the left or right mouse buttons.

Note: *Playback or record in Cycle mode only works if the locators are at least 2 seconds apart otherwise the cycle button will be disabled. This does not apply to loops in the Wave Editor.*

5.2 The Display

The middle of the tape transport pad is a field showing the song position and locators in SMPTE times (hh:mm:ss:ff).

Next to the song position is a record mode indicator (REC-MODE). "REPLACE" is the default mode. When cycle mode is activated, this will switch to "OVERDUB". More on this subject in the chapter "Further Record and Playback Functions: Cycle Mode"). To the far right is the display for the selected or measured sample rate. Below this is an abbreviation representing the clock: "INT" means the Falcon’s internal clock, "EXT" means the clock in the S/PDIF interface and "*EXT" means another external clock (see the chapters "Installation and Program Start: Software Adaptation" and "Recording: Digital Recording from CD or DAT").

5.3 Function Control

To the right of the display are four buttons used for calling the following functions:

Punch In Record with the left locator as preprogrammed "start recording" switch.

Punch Out Record with the right locator as preprogrammed "stop recording" switch.

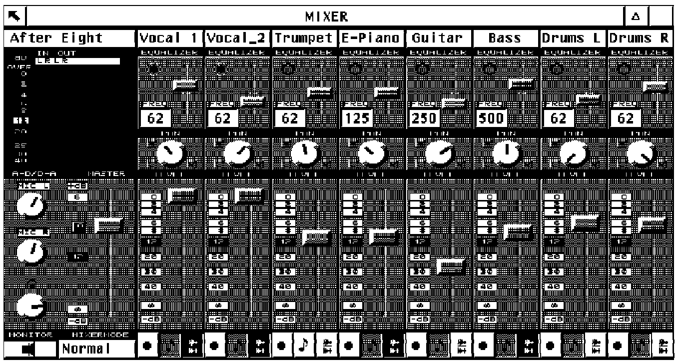
Cycle Record or playback in a loop delimited by the locators.

Sync Switches on the synchronization mode as defined in the corresponding dialog box.

A more comprehensive explanation of these functions can be found in the chapters "Further Record and Playback Functions" and "Synchronization".

The Integrated Mixer

All audio output is routed through the integrated mixer. The mixer (in conjuction with the tape transport controls) allow access to all functions related to recording and playback.



Depending upon the output mode (Stereo/Channels) set in the audio parameter window, the output will either be a stereo sum or individual tracks which will be routed via the FA-8 interface to an external mixer. Alternatively, the mixer can be used as an online effect processor or for "bouncing" tracks together internally (without loss of quality!) to one or more tracks (thus reducing the number of tracks).

The mixer’s user interface opens whenever a song is loaded or a new song is created. The mixer window will remain in the desktop until the song or the program is terminated.

6.1 Window Handling

The mixer window can be moved - like any other window - inside the borders of the desktop (exept the tranport pad) by clicking on the titel bar and dragging by keeping mousebutton pressed. You may choose, whether you want to do this in "full view mode" (left mousebutton) or - as well as known - or in "outline mode" (right button).

- The closebox at the top lefthand will not only close the window, but will terminate the song and close any other window. All song parameters will be saved to harddisk.

Additional, all windows are provided with a button for "iconifying". Clicking on this button, the window will be shrinked up to minimal size and will appear at the buttom of the desktop. A single click on it will resize and place it to the previous position.

6.2 Master Section

The lefthand side of the mixer is the master section. The information bar at the top displays the name of the song and will open a "song info" box when clicked on. On the bottom is an option bar where the audio monitor and peak meter can be enabled/disabled. Switching off the peak meter can be sensible if your hard disk is slow - the system will be relieved of some (not so important) work.

You can see the option bar on the left side.

Mixer mode

The mode button to the right will open a popup menu from which you can choose one of the following:

- Normal: Mode used for normal recording and playback.
- Mixdown: This mode is used for bouncing tracks together to one or more tracks (see the chapter "Playback and Record Functions: Mixdown").
- Online: If online mode is selected, the input signals will be routed directly to the mixer inputs. AudioTracker then works as a stand-alone mixer and/or effects processor.

Peak meter

The input levels are displayed as left and right peak meters (as pseudo LED columns directly below the options bar on the left). Depending upon the output mode, the output levels on the right will either be for the left/right channels (output mode "Stereo") or for each individual track (output mode "Channels"). The peak meter has a "peak freeze" feature so that you can check for maxima. The optimum level should be within the yellow area (-12 dB to 0 dB) - if the top (red) segment lights up the signal level is too high. The peak meter will only be working if the mixer window is active (in the foreground) and the peak button in the options bar is enabled.

Output Control

The master fader sets the output volume up to +6 dB. The 0 dB level can also be set by clicking on to the [0] sign.

A-D/D-A

If you want to record by using the Falcon's A-D-Converter (microphone socket), you may control the input gain with the pots MIC L and MIC R. These pots can be "turned" by clicking on them and moving the mouse in horizontal or vertical direction while keeping the mousebutton pressed.

Below these pots you find a further pot to control the output signal of the headphone socket.

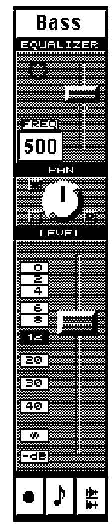
6.3 The Track Modules

Each track in the song has its own mixer module. You can name each track by clicking on the bar at the top of each module. Mixer modules contain some of the following elements, depending upon the Audio Parameters settings:

Reverb

If reverb is active, there is a control for the effect-send of each track. Global parameters are set using the Effect Editor (activated via the Edit menu).

In Channel mode, the effect return channels can be selected with the return button beside the effect send fader. If this button is not activated the channel's output is 'dry'.



EQ

If EQ is active, there is a frequency selector and a gain fader to control the frequencies level. Graphical settings can be made in the EQ Effect Editor

Gate

If Gate is active, the threshold can be controlled by the effect- fader. The button "gate" switches the gate on/off. The other gate parameters can only be edited using the effect editor (refer chapter 10.2)

Panorama

If output is stereo the panorama positions of each track can be set using the pan pots. These pots can be turned by clicking on them and moving the mouse in horizontal or vertical direction while keeping mousebutton pressed (right or downwards: - R; left or upwards: - L). There are three small buttons to set position extremely left, right or center with a single mouse click.

° The pan pots will not appear in "Channel" mode.

° If an FA-8 interface is connected its outputs 1 and 2 correspond to the left and right stereo channels.

° Skillful combinations of reverb and pan can be used to position tracks in virtual space very effectively.

Level

The vertical level faders regulate the relative output levels of each track. The overall level is controlled using the master fader (see above).

In "Channel" mode the DSP module "No Effect" offers a bypass switcher that can be activated separately for each track. The audio signals of these tracks will not be manipulated by the the level faders. If you set the master fader to 0 dB, you will get a 1:1 transmission. Additional, the physical output of each track can be changed. If a double setting is made, you will get a short message when you move the mouse out of the window borders. A quick way to renumber all outputs in increasing sequence is to click on the output box of the track that you want to assign to output 1 while keeping pressed "CONTROL".

Main parts of the mixer can be controlled via MIDI. Refer chapter 11.3 "Synchronization" for details.

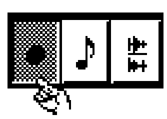
Fader Groups

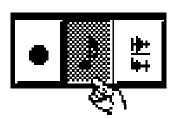
The trackmodule level fader can be combined to fader groups. One group consists of one master and a various number of slaves faders. You can define a master for two groups (A and B) by holding the ALT key (for the A group) and clicking on the desired fader with the left mouse button. On the fader an "A" appears and the fader's color turns to green (in color mode only). In the same way you define some other faders that will now work as slaves. Do the same for the "B" group by holding the CTRL key instead the ALT key. Now you can move the master fader - all slaves will follow but always keep their relative position to the master fader. Move the master with the right mouse button and the slaves will take over the master's position, but the relative position will not be lost. When moving the master fader with the left mouse button again the old positions will be restored. Slaves can be moved separately.

To cancel the fader group "A" hold the ALT key and click on the master "A" fader with the right mouse button. In the same way you can quit a fader's slave status. As long as a group with master fader exists you can define new faders as a slave.

Record / Play/Wave

The record (REC) and playback (PLAY) buttons are situated at the bottom of each module. If the play button is switched off the track will be muted. A track cannot be in both "record" and "play" at the same time if the mixer mode is set to "Normal" i.e the buttons are mutually exclusive. If you are recording in stereo mode, activating the "record" button of a track will also activate the neighbouring track's record button (tracks 1+2, 3+4 etc. are linked). These two buttons are also used for selecting the source and destination tracks for structural operations (copying, deleting etc.) and for audio import/export.





Selecting a "record" track and a "playback" track.

The wave button selects all tracks who are used in the wave editor.

Recording

7.1 Preparation

The following instructions have been written assuming that you have already opened a song and have set all the necessary software adaptation (in the "Audio Parameters" window).

Rewind the "tape" to the required song position and switch on the "REC" button of the track you want to record to. Remember that stereo mode cause adjacent tracks to be activated.

Please secure that the mixer is in normal mode.

Analog recording via DAT recorder

Connect the instrument or microphone to the DAT machine's analog input and set the DAT to RECORD/PAUSE (record mode = analog). Click on the record button in AudioTracker's tape transport pad - the program will also switch to RECORD/PAUSE. Set the input level on the DAT machine - the DAT and AudioTracker levels should be identical or similar. Make sure that the levels are between -12 dB and 0 dB (within the yellow range) when you are playing the instrument.

Note: *Unlike normal analog recording systems, 0 dB should NEVER be exceeded, otherwise the signal will be digitally distorted!*

Digital recording from CD or DAT tape

DAT tapes and CDs will usually be recorded in stereo mode. You may have to check the input mode setting in the Audio Parameters window. Digital to digital recording does not require levels to be set.

Recording from the microphone socket

Connect the instrument/microphone to the Falcon's microphone socket. Click on Record (transport pad) and set the input level using the faders from the audio parameter window.

7.2 Starting Recording

Recording the first track

If AudioTracker is still in RECORD/PAUSE, simply click on the PLAY button to start the "tape". Otherwise you can start recording directly by either clicking on RECORD with the right mouse button or by hitting the "*" key in the numeric keypad of your keyboard. You can listen to the output of the integrated mixer if the audio monitor is switched on (options bar), whereby level, effect send and panorama can be changed without affecting the recording. Finish recording with either the STOP button in the tape transport pad, the spacebar or 0 (numeric keypad). You can then rewind, switching on the "play" buttons of the appropriate tracks and click on PLAY to listen to the recording.

Recording further tracks

Recording further tracks whilst listening to tracks you have already recorded follows the same principle - select the "play" buttons of the tracks you want to hear and the "record" buttons of the new tracks you want to record on.

Note: *If your hard disk is relatively slow you could get the message "Maybe HD too slow!" when you start recording. One reason could be that you have selected too many playback tracks. If you ignore this message you run the risk of the recording being aborted with the alert "HD too slow!". If one or both of the above messages appear although you have a SCSI disk you have either not replace the original HD driver ("SHDRIVER.SYS") with the one on the AudioTracker floppy disk or you have selected a horizontal screen resolution above 640 pixels (16 colour mode) - see the chapter "Installation and Program Start".*

Do not assume that, for instance, "my disk can play five tracks back so I should be able to listen to three tracks while recording in stereo" - write operations to disk (record) are usually much slower than read operations (playback)!

Monitoring

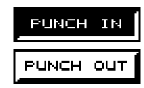
If you are recording/playing back via the Falcon's internal ADC/DAC and the audio monitor is on (mixer window), the signal being recorded will be mixed with the output signal to the headphones. Recording via external ADC (e.g. DAT machine) and S/PDIF is a bit more complicated: the current Falcon's hardware does not allow playback signals and record signals to be routed through the integrated mixer at the same time. Therefore you will have to use an external monitor line: connecting the LINE OUT of the DAT machine (which carries the record signal) and the analog outputs of the FA-8 to an external mixing desk lets you create an individual "monitor-mix" for the instrument player's headphones (see the diagram "Professional configuration" in the chapter "Installation and Program Start").

If you don't have access to a mixing desk you could conceivably make a simple Y-lead to connect the outputs to the amplifier, but audio engineers will probably tell you that this is not good practice!

Further Record and Playback Functions

8.1 Punch In / Punch Out

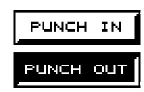
Punch in recording



These functions are used to switch in and out of record mode automatically - the locators are used to delimit that part of the song which you want to record.

Select the Punch In button in the tape transport pad by either clicking on it with the mouse or by hitting the "I" key and rewind to a position before the left locator. Choose a track to be recorded to and start recording (see previous chapter). The RECORD button will be switched off automatically as soon as the "tape" is running and will remain so until the position of the left locator is reached. At this point the RECORD button will switch itself on and the track will be recorded to. You can terminate recording either manually or by using the right locator as Punch Out.

Punch out recording



Select the Punch Out button by either clicking on it with the mouse or by hitting the "O" key. Start recording from a position to the left of the right locator (either manually or punch in). The program will switch to playback mode automatically as soon as the right locator is reached.

Combining punch in and punch out is very useful for improving faulty sections in the middle of recordings which are otherwise good. The section to be re-recorded can be defined very precisely using the Wave Editor and its "snap" functions (see the chapter "The Wave Editor").

8.2 Cycle Mode

A part of the song can run as an "endless loop" in both record or playback modes. The part is, again, delimited by the left and right locators.

Cycle mode is activated by clicking on the Cycle button in the tape transport pad or by hitting "C" on the computer keyboard. The button is disabled (greyed) if the locators are too close to each other. The minimum distace between the locators is about 11 seconds. Shorter loops can be defined within the Wave Editor.

Recording in cycle mode

Recordings in cycle mode can only be started at the left locator - the song position will jump to the left locator if necessary. If you decide later that you want to record normally again you should switch off cycle mode first.

Cyclic recordings are always overdubs ("replace" mode would not be sensible here) - the record mode display will be changed accordingly. Overdubs will not erase any previous material on "tape" but will be added to it. You can start recording as usual and add to the existing part again and again, thus building up complex rhythms, for instance. Other tracks in playback mode can still be listened to while you are recording in cycle mode.

Note: *Excessive use of this feature is not recommended if you are using the Falcon's internal ADC because this converter is rather noisy - each cycle will add noise to your recording which can quickly become intolerable. The same applies to noisy audio sources - you should keep the number of cycles to an absolute minimum if you are concerned with the quality of your recordings.*

Cycle mode and punch in/out are, by nature, mutually exclusive. Recording is carried out in volatile memory and is "nondestructive".

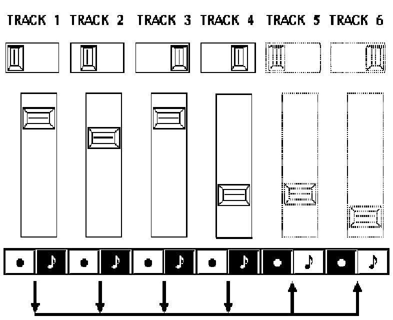
Playback in cycle mode

Playback can be started at any position. The song position pointer will jump back to the left locator as soon as the right locator is reached. This feature can be used for fading "repeats" at the end of a song. Cyclic playback is also helpful for rehearsing a passage you want to record or for testing various mixes.

8.3 Mixdown

You can "bounce" several tracks together to one or two free tracks so that you can delete the originals (thus saving tracks) and/or make any internal effects "permanent". The mixer output in mixdown mode serves as a digital source and therefore you cannot record (normally) at the same time.

Example:



You have created a song with six tracks. Tracks 1 to 4 contain various drums which you want to mix to a stereo sum. Click on the mode button in the master section of the mixer window and select "Mixdown" from the popup list. If cycle or punch functions are on, switch them off and rewind the "tape" to the beginning of the song.

Select tracks 1 to 4 for playback and tracks 5 and 6 for recording (you may have to switch to stereo mode via the Audio Parameters window first). Start playback if you want to do a test run first (and then rewind again).

The fader positions of the record tracks are not important - they are ignored. Start recording as usual. Everything you hear will be recorded to tracks 5 and 6 i.e all mixer parameters and changes to tracks 1 to 4 will be included.

When you are mixing down to a single track, take care to set the pan sliders of the source tracks to the extreme right or extreme left, depending on the input channel as set in the Audio Parameters window.

8.4 8 Channel recording

If you select Adat in the audio parameter window you can record up to 8 tracks simultaneously. The adat bus mode defines which channels will be used as inputs (8 inputs or 2 inputs). If you only want to playback 8 channels to the ADAT interface you don't have to select ADAT. The ADAT interface behaves like FA8.

If you want to mix the 8 channels use the Online function in the mixermode.

Please note: *in 8 channel recording mode you need a fast harddisc. IDE drives will give best performance on AudioTracker.*

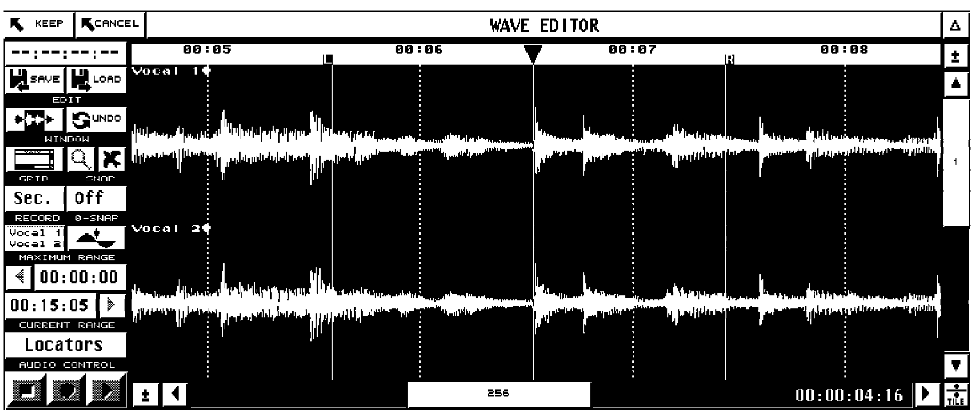
The Wave Editor

AudioTracker's wave editor allows nondestructive editing - all operations are carried out in memory and therefore can be aborted at any time. The display can be scaled and allows access to each sample for smooth cuts and loops. The editor has its own separate audio functions for recording in cycle/overdub mode and for real time acoustic monitoring of the results.

Your computer should be equipped with enough memory to handle the nondestructive editing - 4 MB RAM is minimum. If you have at least 4 MB but still get the message "Not enough memory!" you may have to remove unnecessary resident programs e.g. complex screen saver or large accessories.

9.1 Calling the Wave Editor

Select one, two or four tracks (with the Wave buttons) and set the song pointer to the beginning of the part you wish to edit. Selecting "Wave Editor" from the Edit menu will open a window and the data will be loaded into memory from disk.



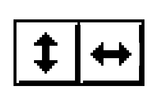
The Wave-Editor-Window

The actual amount of data loaded depends upon the amount of free RAM and the width of your screen. The sample display lets you monitor the audio data visually (at least when it is first loaded) and therefore takes up most of the space in the window. To the left is an area containing all the elements necessary to edit the material i.e. buttons and popup menus.

9.2 The Sample Display

Zoom

The window elements of the wave editor include objects for horizontal and vertical zooming (+/-). Using the left mousebutton the scaling will be decreased, using the right button the scaling will be increased up to 1:1 (every pixel column represents a single sample). the current scaling factor will be shown in the scrolling sliders. Vertical zoom can be defined in a similar way - the quieter passages will be more clearly visible if the display is stretched vertically.



Tile

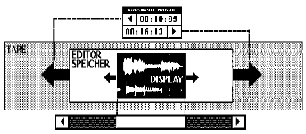
The display can be tiled according to the number of tracks in memory. Possible values are one, two or four tracks simultaneously. Click with the left/right mouse buttons to increase/decrease this value.

Scrolling

There are two different forms of scrolling (see the picture):

Scroll the position of the display within the data already in memory.

Scroll the position within the song i.e. new data has to be loaded from disk.



The former is handled using the window elements (slider, arrows). Clicking on the arrows will scroll in small steps, the grey parts of the slider box let you scroll in steps of one "page" and the slider itself can be "grabbed" and moved directly to a new position. The slider box will show you the position of the display in relation to the song start in SMPTE-time resp. sample number. The vertical slider is used analogeously to scroll through the tracks.

The second method of scrolling (in large steps, from disk) is accessed via the arrow buttons under "Maximum Range" in the control panel to the left of the window. The amount of data updated in memory depends upon the current horizontal zoom - the next/previous part of the song will be loaded and a part of the same size will be "lost" at the other end of the working area.

Note: *If editor's data have been edited and not yet saved, the scroll arrows become red resp. grey. Scrolling from disk will not automatically save these data, you may have to save your work using the save function (see below).*

Window menu

There is actually a further method of positioning the display area within the working area. Clicking on the "Window" button will open a popup menu where you can choose between the following:

- Left margin (of the display) to the current song position
- Left margin to the left locator
- Right margin to the right locator
- Left margin to the start of the block (if one exists)
- Right margin to the end of the block
- If the chosen position is not within the working area the selection will be ignored.
- Call one of nine user-definable positions (including zoom and tile).

The position, zoom and tile values of the display area can be saved to one of nine "window sets" by clicking on a set number while holding down the CONTROL key. Used sets are marked with a dot. Previously stored sets can be called by simply clicking on the appropriate set. Sets are saved automatically as part of the song parameters and will thus be restored the next time the editor is opened. This does not apply if, for instance, you have called the editor with a different number of tracks or have reduced the amount of available memory. The data saved in sets are pointers which are relative to the start of the working area.

9.3 Range (Working Area)

The beginning and end of all sample data in memory are displayed as hourless SMPTE times (mm:ss:ff) under "Maximum Range" in the control panel. These values define the limits of the working area. The position within the song can be scrolled using the arrows as described above or moved directly as follows:

- Clicking on the start time i.e. the upper of the two displays will open the song pointer list - you can move the start to one of ten previously defined points in the song (see the chapter "The Tape Transport Pad: Song Position").

- Double clicking on the display lets you enter the start time directly in SMPTE format.

- A single click while holding down CONTROL sets the editor to the current song pointer.

These actions effectively close the editor and open it at another song position. This means that the buffer (working area) is completely re-read from disk. If the option "Keep/Undo Request" is active you will be warned in case data would be lost in the process.

Current range = Current working area

When mentioning "working area" we have meant the complete memory area which has been allocated to the editor. "Maximum Range" has the same meaning. However, it is sometimes more convenient to limit actions to smaller sections of the working area. Such sections will be called either "current range" or "current working area", depending upon the context. Make sure that you understand these terms as you may become confused if they are not clear!

Under "Current Range" is a button/display which, when clicked on, opens a popup menu containing the following list:

- Maximum: This sets the current range to the complete working area and is therefore the same as the "Maximum Range".

- Locators: The area between the left and right locators. This item is disabled if one or both the locators are not within the working area.

- Window: That part of the working area which is currently visible in the display.

"Locators" is the default when the editor is first started (assuming that they are within the working area).

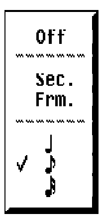
The current range delimits the Keep/undo function, several block functions and defines the start/end of an audio loop (see below).

9.4 Grid/Snap

The song pointer and locators

The graphic display of audio data is well suited for precise positioning of the song pointer and locators. The song pointer (the global one, not the editor's own audio loop pointer) is set with a double click or CONTROL plus single click within the position bar above the sample display. The locators are positioned by single clicks on the left and right mouse buttons. The displayed time in the position bar (mm:ss) are absolute i.e. relative to the start of the song and not to the beginning of the working area. The exact position of the mouse pointer is displayed in the mouse time box in the top lefthand corner of the window.

Grid

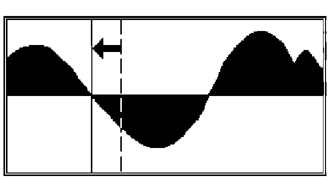


The sample display can be overlayed by a time grid visible as vertical dashed lines. Clicking on the "Grid" button will open a popup menu containing the grid units second, frame, crotchet, quaver and semiquaver. If you have selected one of the last three musical devisions, the time bar will show the number of bars and Çbeats instead of seconds and frames. In this case the grid popup will be expanded by the entry "Options..." that leads you to an input panel. Use keyboard or mousebutton to set sample offset, tempo (bpm) and time signature. The values will change in smaller steps, if you keep the shift key pressed. The sample offset can be used to adjust the grid to the audio data. If you change the tempo, the offset will be set automatically, in order to place one of the grid lines to the beginning of the current working area (e.g. the left locator). In this way it will be very easy to find out the tempo of the area between the locators.

Snap

The positions of songpointer and locators can be quantised to one of the devisions described below. The snap grid has to be selected separately. Clicking on the "Snap" button will open the same popup menu described above. The snap function does not work when you input a position by number or recall from memory (refer chapter 5.1 "The Tape Transport Pad").

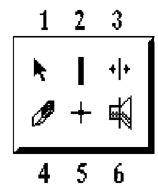
Zero-Snap



The wave editor also allows for so-called "zero snap" which acts on positions in microscopic ranges of the wave and is independant of the above option. Zero snap is also active when positioning the song pointer, locators and block markers. This function looks for the best position for a smooth transition between parts - such positions are (at best) samples of zero amplitude or (otherwise) samples of low amplitude where zero is crossed (see the picture).

Such points can be seen clearly if you set horizontal zoom to maximum (1:1). The samples will, in reality, seldom be exactly "0" - this only happens when a sample is (by chance) taken at the right moment. A tolerance can be set (defined as a percentage of the maximum amplitude) for this reason, below which the program will consider the sample to be "favourable". The default tolerance is 0.5% - this value has been found to be about right for recording via the ADC of a DAT recorder to the SP/DIF. If you are recording using the Falcon's own ADC you should change this value to 1.0%. Zero snap can be toggled on and off by clicking on the button (depicting a wave) directly below the snap button. A double click will open a small editable field in which you can type the tolerance directly.

9.5 The Toolbox



Clicking on the right mouse button within the sample display opens a small toolbox (similar to those found in standard sequencer programs) containing the following tools:

1. Standard mouse pointer: for positioning the song pointer and locators in the position bar etc.
2. Marker: for marking a block
3. Mover: for copying or moving a block
4. Eraser: for deleting single samples directly
5. Fader: for drawing a line which will increase or decrease amplitudes
6. Speaker: for playback a single track in the wave editor

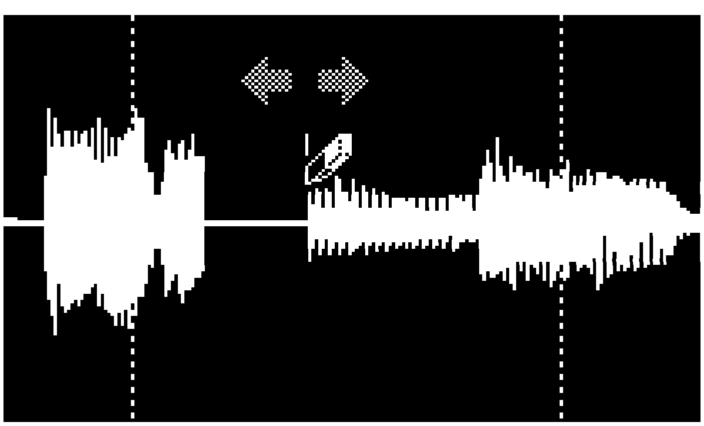
A pencil will replace the speaker tool in the last four scaling factors. With this tool you may "draw" samples, for instance, to close dropouts in the audio data. The standard mouse pointer can be reverted to immediately by clicking on the display with the right mouse button twice in succession.

9.6 Editing Samples

Although single samples can be edited directly, most of the edit functions are block orientated. All edit functions can be called while a loop is being played back - the results of an operation will become audible the next time around.

Erasing samples

The eraser tool can be used to delete one or more samples. Press the righ mouse button to open the toolbox and select the "eraser" icon.



Zoom into the display and erase samples by holding down the left mouse button while moving the mouse horizontally across the display. "Blips" and other unwanted samples can be removed very quickly using this method.

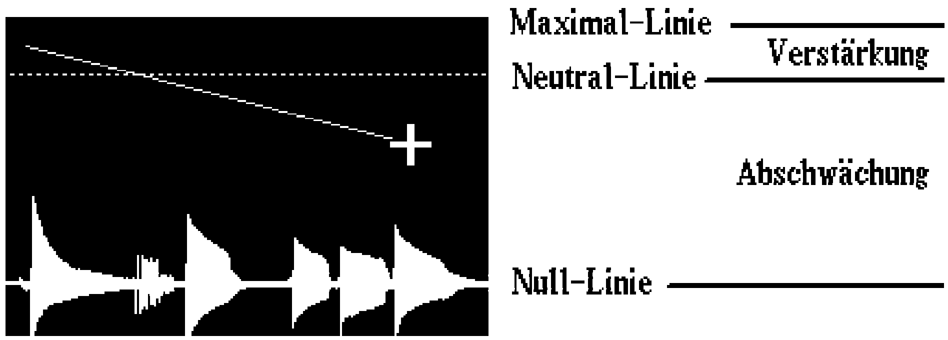
Drawing samples

The pencil tool can be used to "draw" samples. In this way clicks and noises that result out of small gaps in the audio data can be closed.

Changing dynamics

The dynamics of a musical passage can be edited using the "fader" tool. A horizontal line amplifies or attenuates the signal linearly, whereas ascending or descending lines will fade the samples in or out respectively.

Press the right mouse button to open the toolbox and select the "cross" icon. Click on a point in a track where you want to start changing the dynamics - the vertical position defines the starting level and should be above the central "zero amplitude" line. Positions above a "neutral" line (half way up from zero) will amplify the samples, positions below this will attenuate them.



Alter dynamics

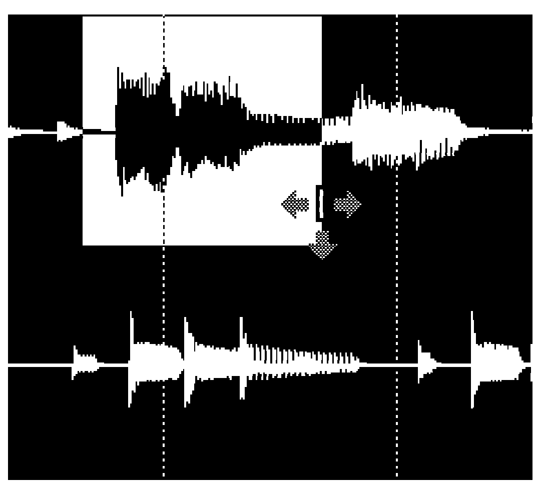
A "rubber band" can then be drawn out to a second position (also above the zero line) which defines the end of the section to be modified its amplification/attenuation. The fader function can be aborted by clicking on the right mouse button as long as this second point remains undefined.

The amplitudes of samples along the length of the line will then be modified by interpolating the vertical positions of the fader line and multiplying each sample with the appropriate factor. An alternative to this function are the "Fade In" and "Fade Out" block functions described in the next section.

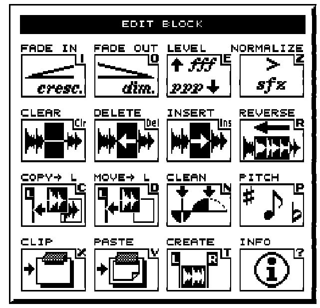
9.7 Block Edit

As already mentioned, most of the edit functions act on "blocks" - these are found in the Block Edit menu. Click on the Block button and select the required function. If a block has not yet been defined, all of the entries except "Create" and possibly "Paste" will be disabled.

Marking blocks



A block defined via the Marker icon



Blocks can be defined in two ways:

- Open the toolbox and select the Marker icon (vertical bar). Mark an area in the sample display by holding down the left mouse button while sweeping across the display. If you want the block to include several tracks you can move the mouse pointer into other tracks before releasing the mouse button. The marked area will be displayed in reverse. The limits of the block may be moved automatically by the program if one of the snap options is active.

- Definition of a block including all samples within the current range can be achieved by double-clicking the Marker icon or by selecting the "Create" function from the Block Edit menu. This feature is very useful e.g. for defining blocks limited by the locators quickly and precisely.

Removing and changing block marks

Block marks are cleared by clicking once inside the sample display using the marker tool. The block marks can be changed via the block info dialog. You may input start and end either in SMPTE time or sample position. Last one enables to create blocks with an exact number of samples.

Fade In

The block will be faded in i.e. volume will increase from zero to the normal volume.

Fade Out

The block will be faded out i.e. volume will decrease down to zero.

Level

The volume of the whole block can also be modified linearly. Clicking on the "Level" button in the Block Edit menu will cause a fader to appear within the display which can then be moved vertically. Releasing the left mouse button will cause the program to process the block accordingly - the right mouse button aborts the function.

Normalize

Another very practical function is "Normalize", which is similar to the "Level" function described above. Normalize should be used sparingly for accentuating particularly important parts of the music (e.g. a drum roll or a bell) because it amplifies the block to the maximum possible volume.

Reverse

The reverse fuction gives you the ability to make an reverse effect.

Pitch

The pitch function enables the tuning of a block about +/- 10 % in a resolution of 120 steps. 10 % corresponds nearly to the interval of one entire tone. Good results can be expected if a relatively short block is edited. In the same way as editing block level, the tuning can be controlled by a fader also while audio loop runs. Quit function by clicking outside of the control panel (left button means "OK", right buttons means "CANCEL").

Clean

This function tries to remove "clicks" and "blips" and smoothes hard cuts between the block limits.

Clear

Clears all samples from the block.

Delete

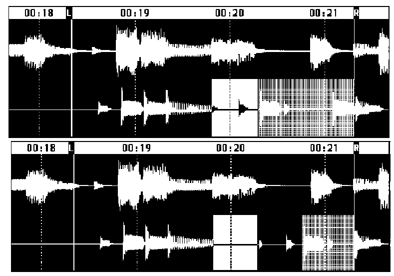
Deletes the entire block. All data within the working area (sic) which follows will be moved to fill the space. Caution: this function can cause the timing of the track(s) to be shifted against each other (see the picture), so you should decide very carefully on the tracks and the area when you define the block! The block marks will remain after the operation but will of course now contain data which has moved to fill the space.

The current working area is "Locators" in the picture. The data which is moved is displayed grey for the sake of clarity.

Insert

This function is the opposite of "Delete". All sample data from the start of the block to the end of the working area will be moved the length of the block "to the right". A space will thus be created whose length is the same as the block.

Note that "Insert", like "Delete" can cause timing problems if the block is not defined carefully. Another problem is that the last part of the working area is lost (see picture).



The block marks will remain after the operation but will contain no data.

The "Insert" function. Note that data from the end of the working area ("Locators" in this example) is lost in the process.

Copy to L

This function is used to copy the contents of a block to the position of the left locator (if the latter is within the maximum working area). The block will be copied to the first track(s) which are visible (sic) in the active display - the required destination track(s) can be set using the tile function and the vertical scroller. Note that "Copy to L" also moves the block markers to the new position.

This funktion moves the whole block including the marks!

Move to L

Similar to "Copy to L", but the block will be moved to the new position i.e the original block will be cleared. ".

Clip

A block can be copied to the clipboard for later insertion. There is only a single clipboard - any data already in the clipboard will be overwritten whenever "Clip" is called (cf. "The Structure Menu: Clip Part"). The block (or better - the "object") can be pasted either as a block using the "Paste" function (see below) or as a "Part" using the "Paste Part" function from the Structure menu.

Paste

If there is an object in the clipboard it can be pasted into the Editor. It doesn't matter whether the object was originally a Part or a Block (cf. "The Structure Menu: Clip Part / Paste Part"). The object will be copied to the position of the left locator if this is within the working area (maximum range) and will appear there as a block. The destination tracks are set as described above ("Copy to L"). If the whole object will not fit (horizontally or vertically) into the working area you will be warned via an alert box - if you are not willing to copy only part of the object you have the chance to abort the function with "Cancel".

Magnify

Magnify will zoom a block to the highest possible size. You can use also the tab key to select magnify. If you want to restore the old view just select shift + tab.

Moving or copying a block

A block can be moved or copied graphically using the Mover tool:

Open the toolbox by clicking in the display with the right mouse button and select the Mover icon (four arrows). Click within the block: a ghost copy of the wave will appear. This can then be moved to any position within the maximum working area. Moving to the edge of the display will scroll in that direction.

Click with the left mouse button to position the block; if you hold down the CONTROL key at the same time, the data will be copied instead of moved. As usual the action can be aborted with the right mouse button.

Switch to overdub mode with the ALTERNATE key an move the block or copy the block with ALT + CTRL.

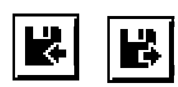
Undo/Redo

Use the Undo function (UNDO key) to undo/redo the last change in the wave editor without leaving the editor window.This function is only available if you have activated the option "Reserve UNDO Buffer" in the "Memory Configuration" dialog.

9.8 Data Transfer with the Hard Disk

Save

You can save the current working area to the song at any time. Only those tracks which are visible in the display will be saved.



If the current range is "Window" the function will save only the data which is visible in the display. Parts of one or more tracks can be saved by setting the left and right locators accordingly, setting current range to "locators" and displaying the correct tracks via "Tile" and the vertical scroller.

Click on the Save button. If the Save/Load Request option (Options menu) is switched on you will be warned first.

Load

"Load" is often used to discard changes. Unlike "Last Version" functions found in other programs, only the current working area will be updated from disk. Whether the original (before calling the editor) data from memory or data from disk will be loaded depends upon your way of working.

9.9 Audio Loops

Play

The current range can be played back as a loop. Almost all the editor functions can be called during playback so you can monitor your work. The locators can also be repositioned during playback - this will take effect the next time around.

Playback is started with the Play button within the editor. The loop will be stopped as soon as you click on the Stop button or close the editor. You can switch back and forth between Cycle mode (tape) and Loop (editor) to compare your edited version with the original.

Record

There is an alternative to recording in cycle mode. A record loop can run within the editor window, the advantage being that this method is non-destructive i.e will not be written automatically to the hard disk. You can also see what you record - the graph will be updated after each cycle. Firstly, select tracks to record on by clicking on the record button to the left of the zero snap button and then select the track from the popup list. Tracks selected in the popup will be marked with a dot. The input mode (STEREO/MONO) as set in the Audio Parameters window is valid for the recording.

A loop within the current range can then be recorded to. This would usually be the area between the left and right locators because they can be set independantly to define a musically sensible loop.

Click on the record button at the bottom of the editor window to start recording. The record area will become red (or inverted in b/w) and you can start playing the instrument.

This method of recording is always in cycle/overdub mode (cf. "Further Record and Playback Functions").

9.10 Closing the Editor

It should be clear that you can call other functions outside of the editor or bring other windows into the foreground at any time. However, the memory allocated to the editor is not freed until the editor is closed so that some functions may not have enough memory to operate. If you consider your work (within the editor) to be finished it is best to close the editor first before attempting to call other functions.

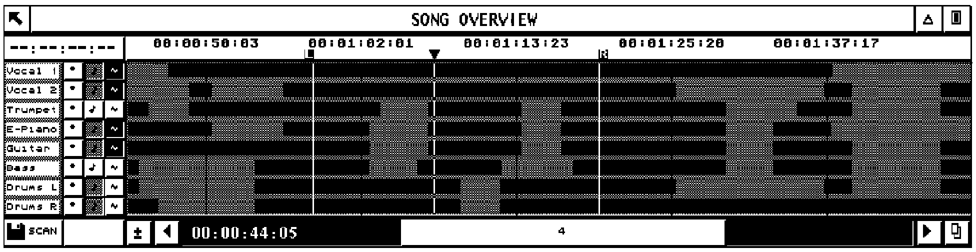
Close the editor with "Keep-Close" if you want to update your song on the hard disk. The working area is always saved. If you only want to save the current working area you should select "Save" first and quit the editor via "Cancel-Close".

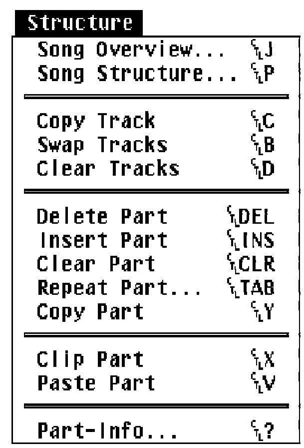
Quitting with "Cancel-Close" means that any work you have done in the editor will be lost unless you "Save" first.

The Menus

10.1 The Structure Menu

Song Overview





The first menu entry will open a window, that displays all sections of each track containing something else than "silence" (i.e. data on disk not equal to zero) as coloured bar graphics. Left hand you will find buttons to select tracks for playback, recording or using in the wave editor. These buttons are switched equal to the buttons of the mixer. In a similar way to the wave editor you may place songpointer and locators by clicking inside of the time bar. If the overview-window is the top window the songpointer will be displayed during playback or recording.

The window is supplied by elements for sizing (fullbox, sizebox), iconifying (refer 6.1 Window Handling), scrolling and scaling in horizontal direction.

Before window can be opened for the first time - especially if it's a song made by an earlier program version - song data have to be scanned from the beginning to the end. This operation will be accompanied by a progress bar and can be interrupted by clicking the right mousebutton. The overview data will be saved automatically when you quit song or program.

Functions that work on audio data (structure functions, recording, importing data, etc.) will cause a rescan of the changed Çsections. If the overview window is not the top window, this scanning will be done later. You may also cause a rescan by clicking on the "Rescan" button.

Song Structure

The structure of the current song, i.e. length and number of tracks can be changed by the dialog "Song Structure" described in chapter 4.2.

Most of the following functions require a source and a destination. The tracks which are to be manipulated are selected by activating either the Play or the Record buttons in the mixer, depending upon the function called. As processing the data can take a while, progress is displayed as a horizontal bar.

Note: *All structure functions work on the harddisk data, so they cannot be undone! If you are not sure you can make file copies using the export function.*

Track operations

Copy Track

This copies a track (selected with "Play") to another (selected with "Record"). Even if more than one track is selected, only the first will be copied (to). The complete track will always be copied i.e. the locator positions are ignored.

Swap Tracks

Swaps two tracks (both selected with "Play"). Only the audio data will be swapped - mixer settings are retained.

Clear Tracks

Clears all tracks selected with "Play". The whole song can be cleared by selecting all tracks and calling this function.

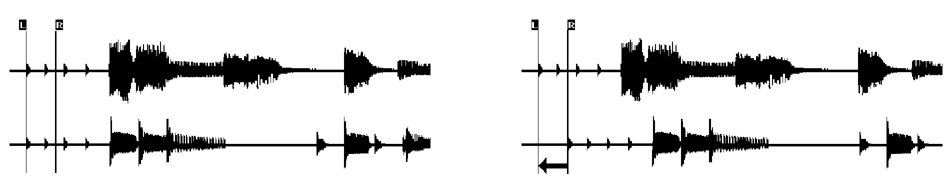
If you only work with snippets of music e.g. to form musical "collages" and there are bits of old recordings on the disk (which would usually be the case) you should "clean up" the track before starting work on a new song. "Clear Tracks" is not called automatically e.g. when a new song is created because the function can take a while (depending upon the number and length of the tracks) and it would be unnecessary for tracks which are going to be completely overwritten.

Part operations

Delete Part

Cuts out the part (section of "multitrack tape") between the two locators and discards it. Do not confuse this function with the "Clear Part" function described below. Unlike cutting a real-world tape, "Delete Part" only cuts out from particular tracks (selected with "Play"). Remember that this can cause tracks to be shifted in time against each other if not all the tracks are selected! This feature can be used to shift tracks which have been recorded out of sync.

The following "trick" can be used to get around a disadvantage of the system (that only two tracks can be recorded at once). You want to record a live performance where you need four separate tracks (e.g. stereo drums, guitar and bass or four acapella singers) so that they can be mixed afterwards; you also have access to two DAT recorders. Record the performance using the two decks. Copy (digitally) both recordings to your hard disk and, using the "count-in", synchronize all tracks using "Delete Part" (see illustration).



Independantly recorded tracks can be synchronized (using the "count-in") via "Delete Part".

Insert Part

This function is the opposite of "Delete Part". An empty part whose length is the same as the difference between the left and right locators is inserted at the position of the left locator. Tracks selected via "Play" will be moved to the right - all other tracks are not affected.

Note: *The total length of the song will not be changed here: data shifted "beyond" the end of the song will be lost!*

Clear Part

Clears selected tracks (with "Play") between the left and right locators only. Not to be confused with "Delete Part".

Repeat Part

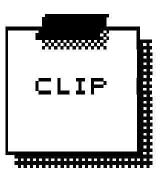
This function is used for creating a row of repeats of the selected part. A dialog box is opened in which you can set the number of repeats. All tracks selected (with "Play") will be included in the operation. Copies are not inserted but will overwrite subsequent data. The song cannot be lengthened using this function - the maximum number of possible repeats is therefore limited by the length of the song.

Copy Part

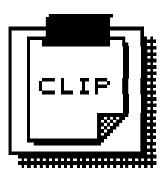
The marked part will be copyed to the current song position only. Just the selected playtracks will be affected.

Clip Part

This is similar to "Copy Part" except that the destination is a clipboard. The clipboard is not in RAM but is part of the partition you have declared as "tape" so you should take care to leave some space on the "tape" when creating a new song (otherwise you will not be able to use the clipboard!). The clipboard can be used to transfer data to or from the Wave Editor or another song on the same "tape". Remember that the clipboard can only hold a single object - previous data in the clipboard is replaced as soon as "Clip Part" is called.



empty Clipboard



used Clipboard

Clicking on this icon will inform you about the number of tracks and the size of the object in the clipboard.

Paste Part

Retrieves an object from the clipboard and copies it to the current position of either the song pointer or the left locator. The "object" can either be a Part sent to the clipboard with "Clip Part" or a block from the Wave Editor (cf. Clip/Paste function in the chapter "The Wave Editor"). The destination tracks are selected with their "Play" buttons. Note the following when copying from the clipboard:

The data is not "inserted" but will overwrite any data in that position. To insert you should make enough space to accomodate the Part first ("Insert Part"). The length of the object (and therefore the extra space you will need) can be determined by clicking on the clipboard icon.

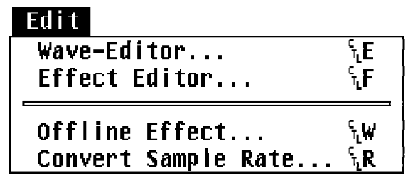
Make sure that the correct destination tracks are selected. If the clipboard contains data from more tracks than the number of selected tracks or there is not enough room left in the song you will be warned with an alert box. Click on "CUT" to accept loss of the overhanging data or "CANCEL" to abort the action.

The object is copied, not moved i.e. it will also remain in the clipboard and can thus be pasted as often as you wish (until it is overwritten with "Clip Part" of course).

Part Info

The part info box shows infos about lenght and activated tracks and can set new values for locator positions.

10.2 The Edit Menu



Wave Editor

This opens the Wave Editor. If the editor window is already open (the entry is ticked) it will be brought to the foreground.

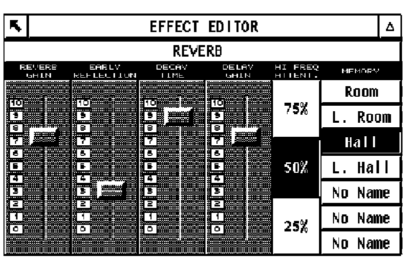
All editor functions are described in the chapter "The Wave Editor".

Effect Windows

If you have selected an effect type (Reverb,Gate or EQ via the Audio Parameters window) a window will open in which the global effect parameters can be set. Audio Tracker lets you configure the effect parameters:

Reverb

- Reverb Gain: Output level of the "wet" signal.
- Early Reflection: Output level of early reflections.
- Decay Time: Length of reverb until it finally disappears.



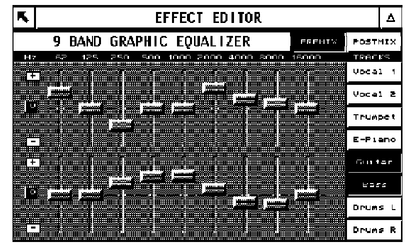
- Delay Gain: Output level of echos.
- High Frequency Attenuation: 25%, 50% and 75%

Play back the recording and experiment a while with these parameters (the editor window must be in the foreground). You should find it more difficult to create natural reverb than synthetic-sounding effects.

On the right of the window are seven memory buttons - the first four already contain editable "presets". Selected buttons will be deselected as soon as you edit the effect. Custom effects are saved to one of the memories by clicking on it while holding down the CONTROL key. The memory button can be given a new name by double clicking on it and entering the name in the editable field.

Dual 9-Band EQ

With the EQ two audio tracks can be controlled independently. First select the position of the EQ: Premix or Postmix.



In Premix mode you can select two of the audio tracks for processing. In Postmix mode the complete stereo-mix signal will be processed. The Postmix mode is not available in Channel mode.

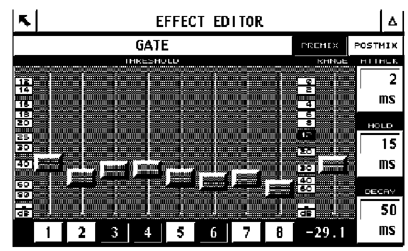
Move the ten gain faders with the left mouse button. With the right mouse button the secon channel will be affected, too. Reset all faders with the ’Zero’ button on the left hand side.

All settings will be displayed on the mixer page. The red LED’s indicates the active EQ channels.

Gate

The main task of a "gate" is to reduce noises but it can be used also to create special sound effects (e.g. hard attack drums using a high "threshold" and a short "decay").

- Threshold (- ... -12 dB) This is the level that has to be exceeded to open the gate (i.e. to let the signal pass). A threshold of - dB is equal to switch off the gate because in this case the gate is always open. You may select different thresholds for each track. This parameter can also be edited in the mixer window but not in the same fine resolution.



- Gate on/off The buttons below the threshold faders resp. the button "GATE" in the mixer window switches the gate on/off.

- Range (- ... 0 dB) This parameter determines the grade of level reducing if gate is active. A value of -40 dB reduces all signals under threshold about 40 dB. Absolute silence will be created by selecting "- dB"; "0 dB" is equal to switch off the gate.

- Attack (0...666 ms) This parameter determines the speed of gate opening when the level of the audio signal exceeds the threshold value.

- Hold (0...666 ms) This parameter determines the time, that the gate will stay open when the level of signal comes under the threshold value.

- Decay (0...9999 ms) This parameter determines the speed of gate closing when the hold time is over. If the signal exceeds threshold during the decay time, the gate will be opened again (in the speed of "attack").

- Premix / Postmix The gate can be switched before or after the level faders ("premix" manipulates the original signal). In the stereo output mode the gate works on the stereo sum, if you have selected "postmix". In this mode the effect editor offers only two threshold faders for the left and right output channel.

Offline Effect

The current effect can be processed "offline", i.e. the audio data on the harddisk will be changed. In this way you are able to use more than one effect in one song or to remove noises in your recording.

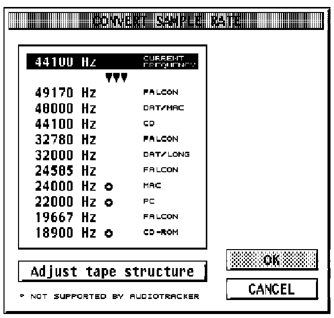
Start playback and edit the parameters using the effect editor. Stop playback. The range of the function can be limited using the locators.

Call "Offline Effect". A dialog will be opened to select the concerning tracks and the desired range ("Song" or "Part"). Quit dialog clicking on "OK" to start the processing.

Note: The effect processing will not be influenced by the setting of the level faders, so the pre/post selection has no effect. Since this is a "destructive" function it may be better to make a safety copy using clipboard or export function.

Convert Sample Rate

The sample rate of a song can be converted using this function. You have a choice of ten of the better-known frequencies between 18900 Hz (CD ROM) and 49170 Hz (Falcon). The current frequency can (at most) be doubled or halved, so some of the entries might be disabled.



Audio Tracker distinguishes between playback frequency and "real" frequency because any arbitrary playback frequency can be chosen in the Audio Parameters window. The real frequency cannot be changed directly - it is determined at record-time or by conversion (see below). Clicking on the title bar of the mixer window gives you information including the real frequency of the song. The "Current Freq." in the Convert Sample Rate dialog box is the same value.

Some of the listed destination frequencies are not supported by AudioTracker i.e. these frequencies cannot be played back within the program. They are included for compatability e.g. for exporting data to Mac or PC.

When sample frequencies are converted the amount of data, and therefore the size of the song on the hard disk, is changed. If the option "Adjust Tape Structure" is switched on, all the following songs are moved i.e. copied to new positions - this can take a while but is the only way to prevent data from being lost or wasting space on the hard disk. If the option is not activated, conversion to a higher frequency would cause data at the end of the song to be lost, conversion to a lower frequency would leave unused space at the end of the song. The difference between the lengths of the original song and the converted one can be calculated using this simple formula:

Difference = song length - (song length * new frequency / old frequency)

Converting between 44.1 Hz and 48.0 Hz will change the song length by about 8%. When the amount of wasted space at the end of the song is not much, it may not be absolutely necessary to switch "Adapt Song Structure" on.

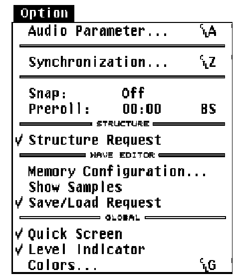
If there is not enough free memory on the disk to allow "Adapt Song Structure" an alert box lets you abort the action via "CANCEL". Clicking on "CONTINUE" causes the song to be converted without the length being adapted!

° *Conversion is "destructive" and therefore cannot be undone! Conversion back to the original frequency will always result in loss of quality.*

° The quality of a recording which has been converted to a higher frequency is always lower than if the material had originally been recorded at this frequency.

10.3 The Options Menu

Audio Parameters



This menu entry has been mentioned quite often in the manual - it opens the "Audio Parameters" window. If a song is already open the current song parameters are adapted and will be saved automatically. If you have not opened a song, the parameters will be based on those used when creating a new song with the function "New Song" (File menu). You should at least enter the required sample frequency so that the correct amount of disk space required by the song can be calculated by the program.

All the audio paramters are explained in the chapter "Installation and Program Start: Software Adaptation".

Synchronization

Selecting this entry opens the synchronization dialog box. This is described fully in the chapter "Synchronization".

Snap

Used for selecting the snap mode from a popup list. Snap will only affect locator positions during playback if it is used outside of the Wave Editor (see the chapter "The Tape Transport Pad": Left and Right Locators". Snap options are described in the chapter "The Wave Editor: Snap".

Preroll

Selecting the preroll button will switch on and off the preroll function. The preroll function is also valid in the punch in mode.

Structure Request

All the structural operations in the Structure menu are destructive i.e. they cannot be undone, and that is why an alert box opens every time you call one of these functions. Experts can switch off this feature, but we recommend that you leave it on (ticked) so that, for instance, hitting the DELETE key by mistake doesn’t destroy two day’s hard work by deleting a part without warning!

Memory Configuration

This menu item opens a dialog for setting up the memory configuration for the wave editor. If editor is opened already this item will be disabled. The first two lines in the dialog show the free memory space, separated into ST-RAM (the original Falcon RAM) and - if available - TT-RAM (alternate RAM). If the alternate RAM should be used as edit memory you have to select the option "Use Alternate RAM". Since the audio functions of the editor play data Çdirectly from the edit memory, you may care that the alternate RAM is able to do this, otherwise you will hear nothing. The undo function of the editor can only be used if you have selected the option "Reserve UNDO buffer". The undo buffer needs the same size of memory than the editor but prefers alternate RAM if available. Normaly, the editor tries to allocate all of the free memory to get a large working area. If this is not desired, you may input the number of bytes the editor has to leave. Less memory space for the editor shortens loading times and may avoid memory deficiency using block or file functions. We recommend to treat a minimum about 20000 to 50000 bytes in any case.

Show SMPTE / Show Samples

If the mouse is inside the sample display, it’s position is displayed either as SMPTE time or as sample number. This menu item switches between the two possibilities.

Save/Load Request

The functions "Save/Load" and "Keep/Undo" are used for data transfer to and from the hard disk by the Wave Editor. There would normally be an alert to warn you that data may be lost in the process - this feature can be switched off (no tick).

Quick Screen

The level indicator and the songpointer display are drawed by certain graphic subroutines to enable a quick refresh also during the harddisk accesses. These subroutines do not work together with some graphic accelerators. In this case you have to switch off this option.

Level indicator

Use this button if you have problems with performance. The peakmeters will not be redrawed.

Colors

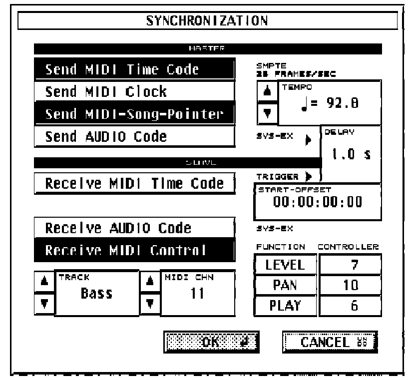
If the program runs in color mode, you can set the colors of the wave editor's display, the overview's bar graphics and the desktop background to your preferences. Instead of the uni-colored desktop, you may load any GEM image (*.IMG) in size of 64 x 64 or 128 x 128 pixels with number of colors not higher than current graphic mode.

Synchronization

AudioTracker supports several methods of synchronizing the program with MIDI recording equipment and/or other hardware (a hard disk recording system based on AudioTracker can have 16 tracks or more).

Although connections in all the modes are via MIDI, only synchronization with sequencers or MIDI recording systems will be called "MIDI Synchronization" - if a second Falcon (and AudioTracker software) is added to the system this is via "Audio Synchronization".

Open the synchronization dialog (Options menu) to select the appropriate mode. AudioTracker will be either the "master" or the "slave" in all modes.



11.1 AudioTracker as Master

AudioTracker should generally be used as master when synchronizing together with MIDI recording systems because any tempo problems the sequencer might have cannot be corrected by the program.

Send MIDI Time Code

AudioTracker sends MIDI Time Code (MTC) in this mode. Set the appropriate synchronization parameter of your MIDI system to "receive" ("external SMPTE", "ext. MTC" or similar). The MIDI system will then start at the correct position when you run AudioTracker and will carry on synchronously because the song position is transferred in SMPTE format.

Send MIDI Clock

This mode should only be used as a last resort if your MIDI system does not support MTC. There are two main reasons for avoiding this mode: Firstly, MIDI Clock is not precise enough for many applications. Secondly it is not a true time code but is dependant upon tempo (hence the name) i.e. AudioTracker determines the tempo of the MIDI system. Set the required tempo in AudioTracker via the keyboard or by clicking on the arrows.

Send MIDI-Song-Pointer

This option should be switched on (inverted) if you are forced (or wish) to synchronize via MIDI Clock. This causes the song position to be updated in the slave system (i.e. MIDI system) - otherwise only start and stop signals would be sent and the song would have to be started from the beginning each time you run it. It is also a nice feature together with MTC as you can follow the song position in the MIDI system while rewinding etc.

Send AUDIO Code

AudioTracker also implements its own system exclusive "Audio Format" for running more than one Falcon simultaneously (both computers must be running AudioTracker). A dedicated format was necessary because the program has to run several initialization routines before record or playback - there is a delay between activating the Start button and playback which can vary according to several factors. To the right of the "Synchronization" dialog box is a box marked "Delay": increasing this value will cause two Falcons to be synchronized more safely (i.e. they will both have enough time to carry out the above mentioned initialization routines). The default value of one second can usually be reduced without problems arising. The current version of the Audio Format includes Start-Play, Start-Record, Stop and a precise Song pointer (to single samples). However, the system is being developed further and future versions will also include locator positions, song selection, sample frequency etc. Practically, this means that the present version allows remote control of "tape" transport for both systems from the master system. If you wish to record on the master and play back from the slave then you should start the slave first (Play) and then the master (Record).

Tempo

The value of "Tempo" controls the speed of the sequencer if MIDI Clock is used. It also affects the musical "snap" quantisation. It therefore makes sense to enter the tempo here even if synchronization is via MTC. This value is otherwise of no importance, but can be helpful for the Wave Editor or while setting locators.

11.2 AudioTracker as Slave

Receive MIDI Time Code

AudioTracker cannot compensate for tempo fluctuations if it is running as slave. This mode is therefore only used as a trigger so that AudioTracker starts as soon as MTC is received. This function was implemented at the request of several users who wanted to synchronize AudioTracker with analog tape machines e.g. to "fly" single short passages from tape into an AudioTracker recording.

AudioTracker is ready to receive MTC whenever record or playback is activated at any position in the song. Reception can be interrupted by clicking on the right mouse button.

More time is required when the song is to be started at the beginning due to the above-mentioned initialization routines and calculation of the correct sample position. The program can only be started with a short delay. The editable delay value makes this effect calculable so that synchronous start is possible - do not set this value too low!

If, for instance, you want a recording or playback to start at the song position 00:01:00:00 you should start the tape machine at 00:00:59:00, assuming a delay of 1.0s. AudioTracker will initialize, wait and start at exactly 00:01:00:00 even if the initialization is finished before this time.

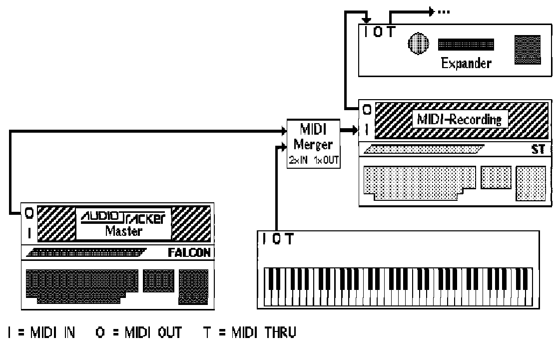
Receive AUDIO Code

This mode only works when the system is connected to a second Falcon with the "Send AUDIO Code" option switched on. As soon as e.g. the "Play" button is activated on the master, the slave will receive a signal for an imminent playback. Unlike MTC you don't have to switch AudioTracker into stand-by yourself. The program initializes playback and waits for the final start signal from the master. Only when this is received will the "tape" start rolling. The slave therefore needs two signals. The delay is set on the master and is not important for the slave. The slave can be forced into the above-mentioned "wait" state by simply clicking on its "Play" button (in our example). The master's first signal would then be ignored. This feature allows recording on the master and playback on the slave (or vice versa).

11.3 Installation

MIDI-Synchronisation

This section illustrates three example applications. The synchronization settings required and any changes to be made to the MIDI system are listed below each diagram. The MIDI system of the first example consists of an Atari ST running MIDI recording software (e.g. Cubase, Notator or similar), a MIDI keyboard (e.g. a master keyboard) and a sound source (e.g. an expander module).



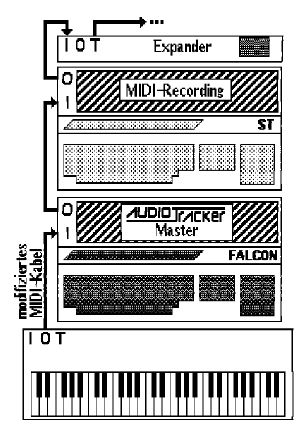
AudioTracker (Master)	MIDI-System (Slave)
a) send MIDI Time Code	Extern SMPTE-Sync,
optional:	Receive MTC
Send MIDI-Songpointer	Set tempo

OR

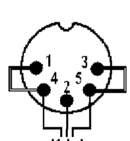
b) Send MIDI Clock	External MIDI-Clock-Sync
Send MIDI-Song Pointer	Set tempo, if this exists
Set tempo	Activate tempo correction
	(e.g. in Cubase: switch Mastertrack on)

Extras: A MIDI Merger is required for the above configuration so that both the master keyboard and the Falcon can be connected to the ST's "MIDI In" port.

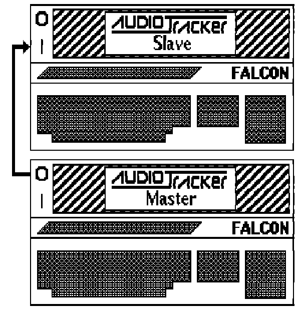
If you want to avoid buying extra harware you can modify a standard MIDI cable. The Falcon's MIDI OUT socket also has a MIDI THRU signal! If you have any experience at all with a soldering iron, this shouldn't be too much of a problem. Open the plug at one end of the MIDI cable and connect pin 1 to pin 4 and pin 3 to pin 5. Pins 1 and 3 are not normally used by MIDI, which means that the cable can be used later for standard connections. You can now connect all the components in series:



A standard MIDI cable can be modified to make use of the Falcon's MIDI THRU signal.



Audio synchronisation



AudioTracker 2 (Slave)

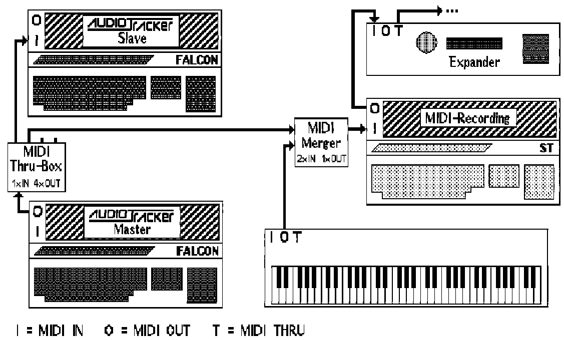
Receive AUDIO Code Delay at least 0.5 s

AudioTracker 1 (Master)

Send AUDIO Code

Both AUDIO and MIDI

AudioTracker 1 (Master)	**AudioTracker 2 (Slave)**
a) Send MIDI-Time Code	Receive AUDIO Code
optional:	



Send MIDI-Song Pointer	
Send AUDIO Code	MIDI-System
Delay at least 0.5 s	Extern SMPTE-Sync,
	Receive MTC , set Tempo
OR	
b) Send MIDI Clock	Extern MIDI-Clock-Sync
Send MIDI-Song Pointer	set tempo, if
Send AUDIO Code	this exists,
Tempo einstellen	Activate tempo correction
Delay mindestens 0.5 s	(e.g. in Cubase: switch Mastertrack on)

Note: The selected synchronization mode is only active if the SYNC button in the tape transport pad is switched on!

MIDI Remote

Level, Panorama, Masterfader and Play/Mute can be remote controlled by MIDI events. In order to do this you can set different controller values for each value.

If mixer window is not active, i.e. is not the top window, MIDI data will be received and processed, but faders will be redrawed when mixer comes to foreground.

Data backup, import and export

12.1 Streamer Function

The streamer function can only be used if there is a DAT recorder connected to the Falcon via an S/PDIF interface. You should reduce the volume of the connected amplifier because data transfer is not very musical to say the least! It is also very loud. Select "Back Up Tape" from the File menu. Select the partition to be backed up in the dialog box. The streamer function is specially tailored for the AudioTracker data format and can only handle files which have been declared to be "tape". Make sure that a start-ID is set on the recorder as you might need to find this data later! The ID can be automized (AUTO) by most DAT recorders, but if there is a longer pause other IDs might also be set (these can safely be deleted). Start recording on the DAT machine. As soon as the tape is running, click on "START!" in the dialog box. The familiar progress-bar will let you know how much data is still to be saved. The partition and the name of the tape (if named) are also displayed.

The display disappears as soon as the bar reaches the end. You can stop the DAT tape and set an end-ID.

If you want to make sure that the DAT machine has recorded all the data correctly you can verify it as follows: Rewind the DAT tape to the start-ID and click on "YES" in the alert. Press the playback button on the DAT recorder. AudioTracker will now compare all the recorded data with data on your hard disk and you will be informed if any errors i.e. differences are found. Faulty backups should be repeated - you may have to change the DAT cassette for a higher quality one.

The backup process can be interrupted at any time by pressing the right mouse button; however, then is no guarantee that the backup data can be restored correctly.

12.2 Restore Tape

This function writes a partition which has been saved with "Back Up Tape" back to the hard disk. The partition does not have to be the same one which was saved, but must have at least the same capacity. The destination partition must also be a "tape" so that AudioTracker can recognize its position and size and thus adapt the transferred data to it. This function is operated in a similar way to the backup function: Select "Restore Tape" from the File menu and choose a destination partition in the dialog. Wind the DAT tape to the required position. Click on "START!" in the dialog box and play back the DAT tape. The name of the tape and the destination drive will be displayed as soon as the program has recognized the tape data and if the destination partition is large enough - the data is restored.

If you have selected the wrong partition by accident you can abort the action by pressing the right mouse button. This is only a "last resort" - if any data has already been transferred you will have overwritten data on another "tape", assuming this "tape" is not empty. However, the directory (which describes all the songs on the "tape") is only updated after all the data has been restored without any errors, so the damage is minimized. All this should make it clear that this function cannot be used to restore only part of the data on tape:

Note: *The streamer functions work with "images" - they are not conceived for saving or restoring anything but complete "tapes". In an emergency i.e. if for some reason the data on a DAT tape is incomplete, transfer should only be interrupted by hitting the PAUSE key on the DAT recorder. The restore function itself must be allowed to finish normally otherwise the restored data will not be accessible ("Open Song" will not work).*

Saving a single song (or parts of a song) can be achieved using the export function described below.

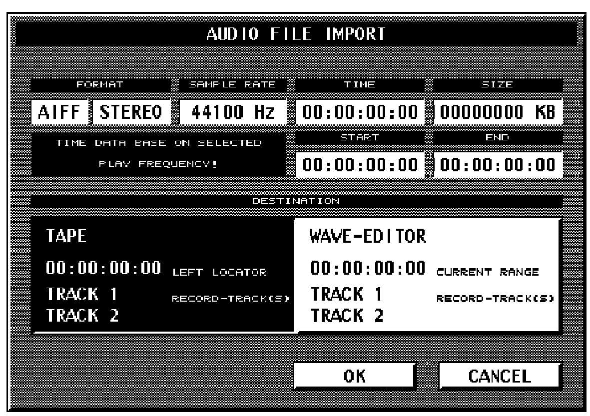
12.3 Audio Import

Sample files in AVR-WAV-DVSM- or AIFF format can be imported into your song. The data can be loaded directly to hard disk or into the Wave Editor. In both cases, a destination position must be defined first. This position is defined by the left locator when importing to hard disk (i.e. directly into the song) and the destination track is selected with the "Record" button in the mixer window. If you are importing into the Wave Editor, the current working area defines the destination. To import the samples to the position of the left locator set "Current Range" to "Locators". The destinatio track is the track selected from the Wave Editor’s "Record" popup list (cf. the chapter "The Wave Editor: Audio Loops").

Call "Audio Import" and select the required file from the standard file selector. The program opens a dialog box displaying the sample format, size in kilobytes and the length as SMPTE time (at the current sample rate of the song).

If you want only a part of the file to be loaded, you may input start and end of this section as SMPTE time.

Below this are two editable fields containing assumed destination parameters. Check the information carefully, especially the track numbers - if the sample file’s format (mono or stereo) does not match the record mode, AudioTracker would record to the next track (stereo mode) or only to the first of two selected tracks (mono mode).



The Audio Import Dialog Box

If the Wave Editor is open the program assumes that this is the destination for the sample data and the corresponding field is preselected.Click on the field with the destination you want .If the destination area is unable to take up data completely, you will get a message with the opportunitiy to exit.

The import function can be used to copy data from audio CD directly into the wave editor or onto the harddisk. A CD ROM drive that supports audio functions is required. In the fileselector you only have to select the drive instead of a filename - a further menu appears if an audio or mixmode CD is inserted. After selecting the desired CD track, the import dialog will be opened and import will be continued as described above.

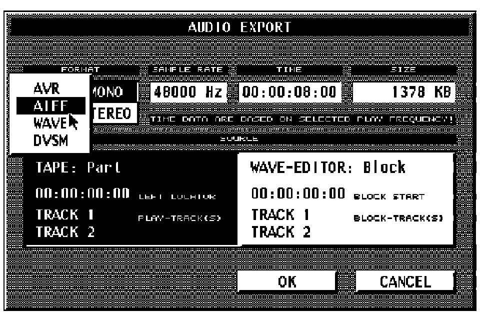
Note: Not all of the CD-ROM-drives supply audio data with a reliability of 100%! An error check tries to locate them, but it may happen that the function must be finished ("Audio CD Read Error!" will occur). A retry may be successful all the same! The progress bar message box shows you the number of data errors, so that you can decide to interrupt or not. Since not all errors can be found, it is recommendable to check the data by ear or using the zoom function of the wave editor.

12.4 Audio Export

Parts of the song can be saved as a sample file in different formats so that the data can, for instance, be edited with another program. The source is defined as either a Part (between the locators) or a block in the Wave Editor. If you want to export a Part you should select the required tracks with their "Play" buttons - you can export any arbitrary tracks, they don't have to be neighbours. Calling the "Audio Export" function will open a dialog box in which you can set the following:

- Format: Choose AVR,WAV,DVSM or AIFF and MONO or STEREO

- Source: If the Wave Editor is open and a block is marked you have a choice between "Tape: Part" and "Wave-Editor: Block". Check the entries for the start position and the tracks because only the first source track (mono) or the first two (stereo) will be exported.



The Audio Export dialog box

All actions within the dialog are interactive i.e. all variable parameters will be updated immediately. If you change (for instance) from MONO to STEREO, the size in KM will double and an extra track will appear in the source fields. If you select a different source the length (SMPTE time) will change accordingly; the mode might also be switched automatically to MONO and the STEREO button will be disabled if the new source only contains a single track. The choice of format depends upon the program which you want to load the data into - most programs can read AVR format. The well-known programs "Logic Audio" and "Cubase Audio" can load AIFF and "Audio Master" is happy with either format.

Click on [OK] when you are satisfied with the settings. A file selector is opened with which you can choose the path and file name. AudioTracker checks first whether there is enough space available on the destination medium and gives you the chance to select a different destination if the first one is too small.

The export function can be used to "stream" a song or individual tracks to DAT. The procedure is as follows: Set the left locator to the beginning of the song and the right locator to the end. The simplest method is to wind to the end of the song using fast forward and then click on the display of the right locator while holding down the CONTROL key. You can then export pairs of tracks in stereo. Stream the files with the program "DataDat" which is part of the S/PDIF package. The files can then be loaded back into AudioTracker at any time in the future without causing synchronization problems between the tracks (they all have definite positions).

12.5 Cubase Import

This function can transfer a complete cubase audio arrangement to an AudioTracker tape file. In order to do that just select all parts in cubase with ctrl-a and save it as "*.PRT". Also save the audio pool ("*.POL"). In AudioTracker select this two files and after that you can make an individual channel routing to AudioTracker tracks.

The flag skip muted tracks/parts will skip muted tracks/parts.

Basic Principles

In order to be able to process audio signals with a digital system like a computer, the voltage of the signal must be sampled (i.e. measured) at very short regular intervals and these values stored as a list of data. This is the job of the ADC (Analog to Digital Converter). The data can be converted back to audio signals using a DAC (Digital to Analog Converter).

Sample Rate

The quality of the results depends to a large extent on the "sample rate". The sample rate (also called "sample frequency") is the number of times per second that the voltage is sampled. The rule of thumb is: the higher the rate, the better the quality. As you need at least two values to describe a wave (think about it!), the sample frequency must be at least double the frequency of the highest frequency in the source. The human ear can detect frequencies up to around 16 kHz (some say 20 kHz) and so a sample rate of 32 kHz is high enough for many purposes. This means 32000 samples per second, per channel!

A few sample rates have become de facto "standard" e.g. 44.1 kHz (CD) and 48.0 kHz (DAT). Some applications also work with 32 kHz e.g. digital radio and long-play DAT.

Most digital audio equipment, including the Falcon, work with 16-bit resolution - the "measurements" are much more precise than 8-bit. Signals with very low harmonic distortion (theoretically about 0.0015%) and high signal-to-noise ratio (usually above 90 dB) can be recreated.

The S/PDIF-Interface

The main advantage of digitalizing audio signals is the fact that digital data can be processed in many different ways by a machine (some kind of computer!) without loss of data i.e. quality. As long as the "sound" stays digital it can be copied, filtered apart, remixed etc. as often as you like without it sounding worse after each generation. Most digital audio equipment is still dedicated hardware (this situation may change in the future - remember hardware sequencers?). They are usually equipped with digital interfaces so that audio data can be transferred without having to convert it to analog signals first.

One interesting effect of digital transfer is that, if you copy a CD to a DAT tape, you may find that the output of the DAT sounds better than the original CD! The sound you hear has only been converted to analog by the DACs in the DAT recorder, and these are often better than those in CD players (DAT recorders are still considered "high-end" and are expensive enough to warrant using quality components).

The Falcon does not have a digital audio interface - the S/PDIF interface has been developed especially for this purpose. The S/PDIF is connected to the Falcon's DSP port and expands the Falcon's arsenal of ports to include both optical and coaxial standard digital interfaces. The combined system is thus able to copy the digital data from a CD to hard disk, process the data and send it to DAT without any loss of quality whatsoever. The advantage of the combination Falcon+S/PDIF+DAT for recording analog signals is that the DAT's higher quality ADC can be used. The S/PDIF interface also offers the standard frequencies 44.1 kHz and 48.0 kHz which are not supported directly by the Falcon.

DAT and CD

The "professional" configuration described in the chapter "Installation and Program Start" makes fully digital productions (DDD) possible; the digital DAT master can be transferred directly to CD. The only necessary conversion to analog takes place in the home of the consumer.

Unfortunately, there is one small problem: the standard sample frequencies of CD and DAT differ. Transfer of digital audio signals is always at the rate of the source. If, for instance, you copy a CD to DAT (digitally), the DAT recorder is forced to record the data at 44.1 kHz although it would normally use 48.0 kHz. If the DAT recorder is used as a DAC (for recording analog signals and converting them to digital) it will record at 48.0 kHz. If you are planning a fully digital production (to CD) you need recordings sampled at 44.1 kHz!

Possible solutions:

- Work with ADCs which can run at 44.1 kHz. Unfortunately, this rate is only available in a few DAT recorders.

- The whole production is at 48.0 kHz. The sample rate is converted before the final mix. The function "Convert Sample Rate" can run without further output in the Edit menu.This can take several minutes because of the huge amount of data involved.

- Use a dedicated hardware Sample Rate Converter, which either converts the signals while recording or the final production to the CD rate in real-time. Your AudioTracker dealer will be able to provide you with "SRC", an extra interface which is connected to the S/PDIF. The SRC provides all the frequencies 32.0 kHz, 44.1 kHz and 48.0 kHz.

Keyboard Commands

There are more functions available directly from the computer keyboard apart from those listed in the menus:

Note: The symbols (,),0,* and / are on the numeric keypad.

Windows (general)

Key	**Function**
RETURN	Close and keep
ESC / UNDO	Close and discard
Cursor keys	Scroll fine
+ SHIFT	Scroll course

The tape transport pad

Key	**Function**
0 or Spacebar	Stop record or playback
.	Stop recording and start again lying
/	Record pause
*	Start recording
ENTER	Start playback
( )	Wind fine (frames)
( ) + SHIFT	Wind course (seconds)
( ) + CONTROL	Song position to left locator/start/end
[{	Songposition to left locator
]}	Songposition to right locator
I / O	Punch In / Punch Out
C	Cycle
S	Sync
F1...F10	Restore locators
+ CONTROL	Save current locators
+ SHIFT	Restore song position
+ SHIFT + CONTROL	Save current song position

The Wave Editor

The keyboard shortcuts in the Wave Editor will only work if the mouse pointer is inside the editor window (except window handling):

Key	**Function**
0 or Spacebar	Stop audio loop
ENTER	Start audio loop (playback)
*	Start audio loop (record)
S	Save current working area
L	Load to current working area
M	Set current range to maximum
;	range to locators
:	range to window
B	Call block edit menu
- / +	Horizontal zoom
shift +/-	Vertical zoom
/	Tile
1...9	Restore window sets
+ CONTROL	Save window sets
TAB	magnify
shift TAB	restore magnify

The block functions can be called directly by keyboard. You find the corresponding key written down in the block dialog box.

Mixer

1..8	play/mute

Definitionen

C.1 Song Parameters

All the parameters listed will be saved for each individual song whenever...

- the song is closed or another one opened
- The Audio Parameters window is closed with "Keep"
- the title, number of tracks or length of the song are changed via the function "Song Structure"
- the program is terminated when a song is still open

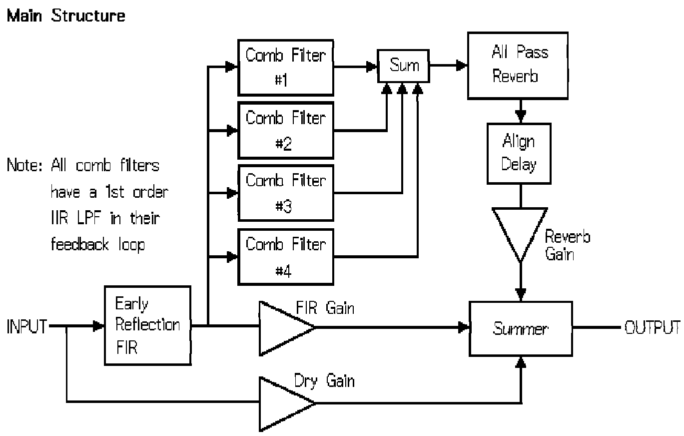
C.2 List of Parameters

Song title, position on the hard disk, length of the song, number of tracks, frequency of the last recording (real), all settings in the Audio Parameters window, all mixer settings (except those of tracks being recorded) including track names, all effect parameters including memory sets and their names, the current and saved locator positions, the saved song positions, all synchronization settings, all options, saved Wave Editor window sets, zero snap tolerance.

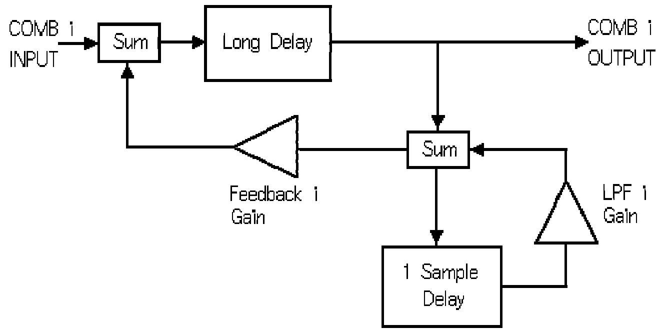
C.3 Reverb

Flow diagram of the implemented reverberation program

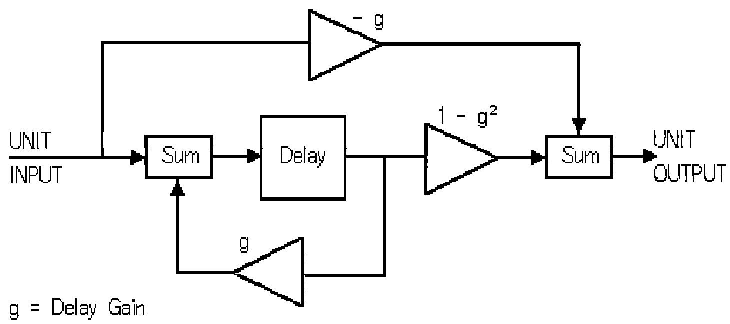
Version 1.1 October 1994



Comb Filter Sub Structure



Unit (All Pass) Reverberator Structure



g = Delay Gain

C.4 Audio Sync Format

Version 1.1 September 1994

AudioTracker has its own synchronization method based upon MIDI system exclusive messages. Each message starts with $F0 ("sysex") according to the MIDI standard and ends with $F7 ("eox"). The Audio format is being developed further for future versions so that complete compatability cannot be guaranteed.

Stop	$FC
Record	$F0,$00,$01,$F7 - Delay - $F0,$00,$10,$F7
Play	$F0,$00,$02,$F7 - Delay - $F0,$00,$10,$F7
Songpointer	$F0,$00,$07,$aa,$bb,$cc,$dd,$F7
	$aabbccdd (LONG) = Sampleposition in Bytes