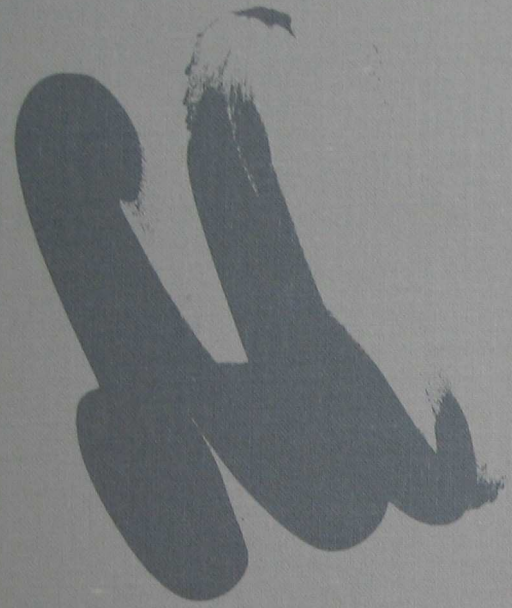
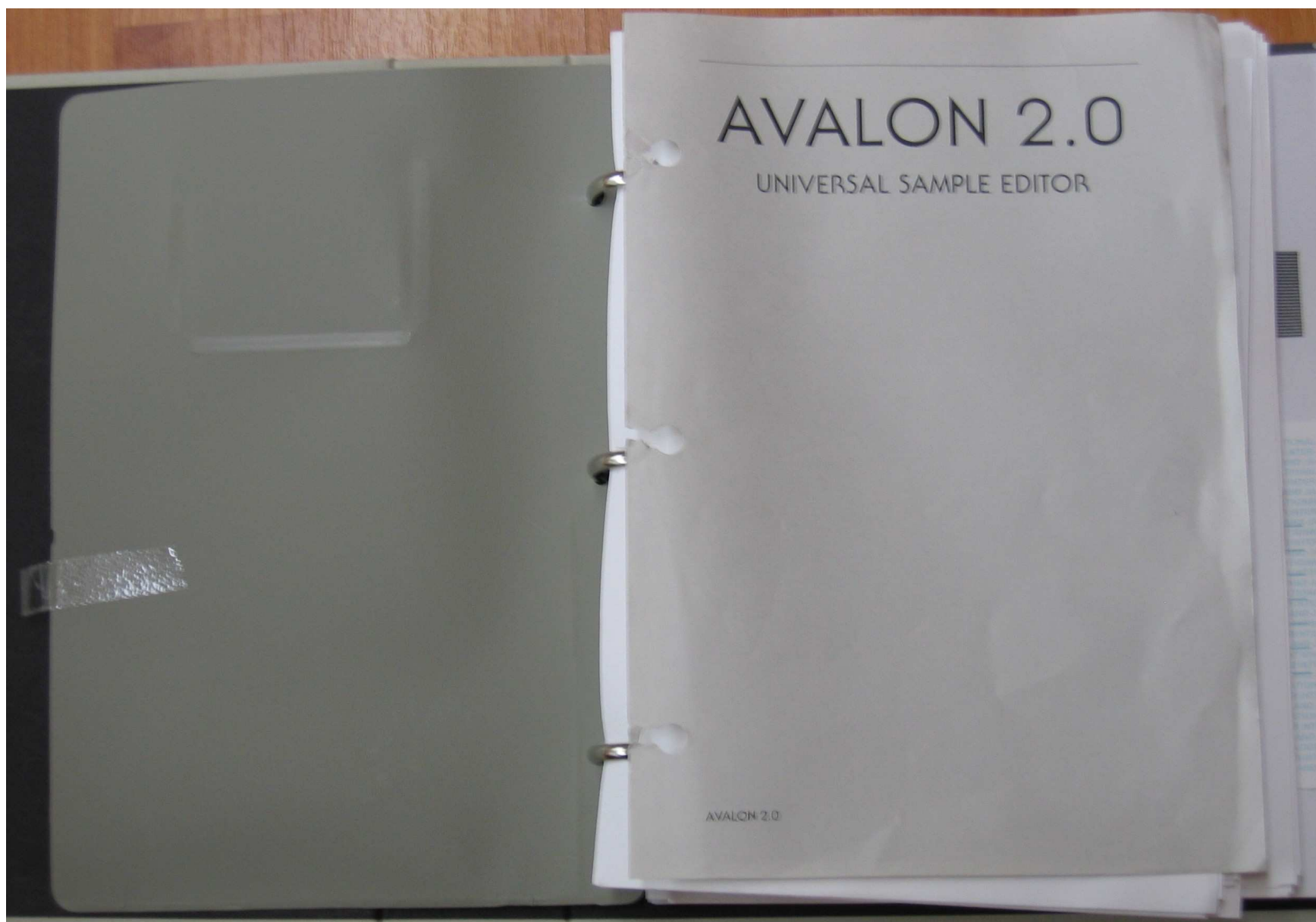




Steinberg



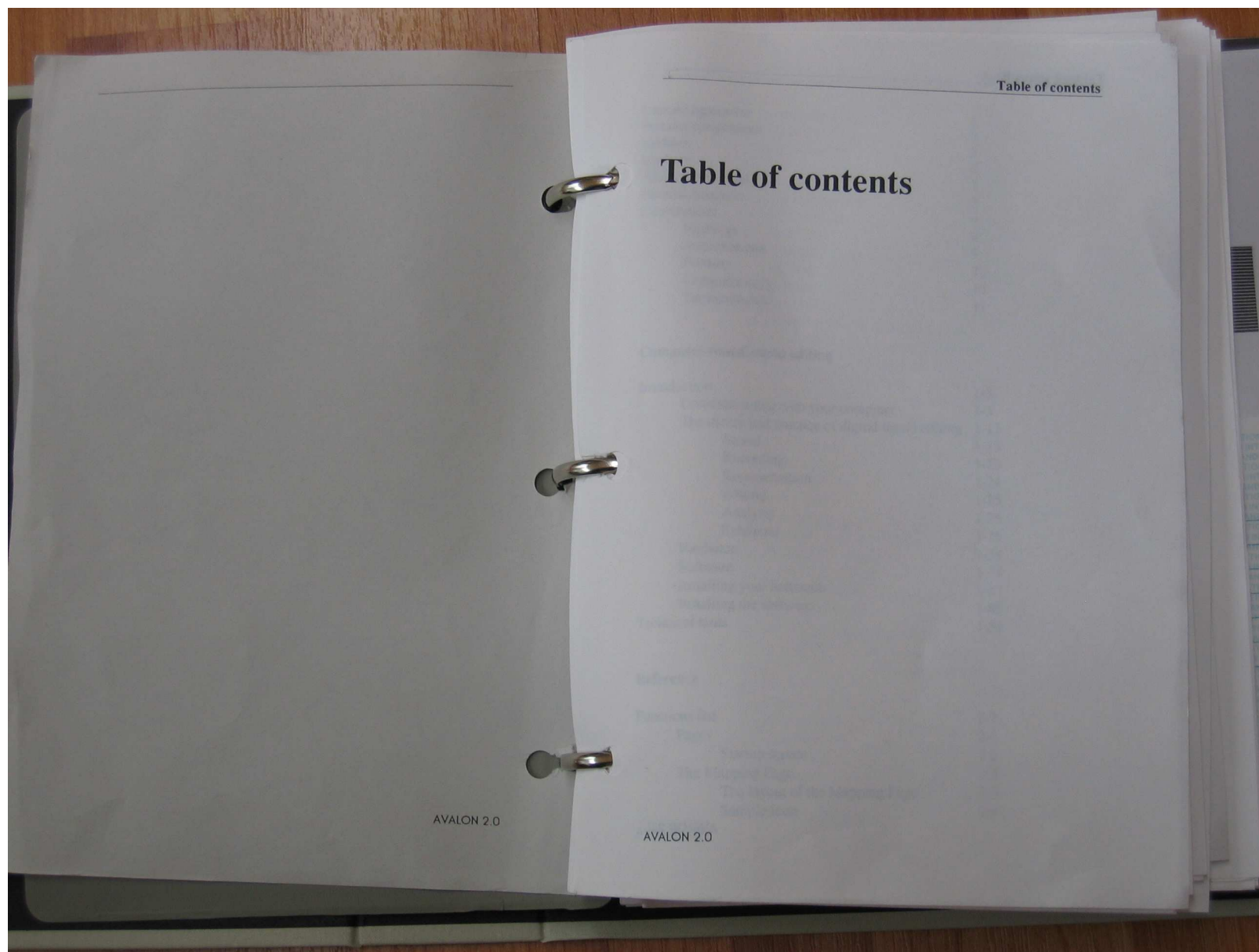


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Updates

Software

Christian Assall
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Michael Schindler
Jürgen Wehnert

Manual

Martin Post

Manual graphics

Peter Post

Steinberg would like to thank all those who have been directly or indirectly involved in the development of this system.

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Steinberg Soft- & Hardware GmbH, Germany

License agreement

The sale of software and software-controlled systems is subject to special licensing conditions which form an integral part of the contract of sale.

Product components

The product you have acquired includes the following:

- Program Disks I & II
- Hardware key
- Manual
- Registration card

Updates

As Steinberg products are continually being improved and developed we would like to be able to inform you when a new version of an existing product becomes available.

In order to benefit from this service, you should return the attached Update Service Form to our distributor immediately after purchasing one of our products. You will then be informed when a new version is released.

Only registered users can take advantage of the update service.

Backups

This program is protected by a hardware key that has to be inserted into the ROM port on the lefthand side of your computer. It will only function if the key is installed.

As a result, making backup copies of the program disk presents no problems. In fact, we would strongly recommend that you copy the program immediately after purchase - either to another disk or to a hard disk - and then use this as your working copy.

You will find a description of how to copy disks in your computer handbook. You can, however, also make use of one of the many copying programs that are available commercially or on a public domain basis.

Copy all files onto your working disk or hard disk. Keep the original disk in a safe place and only work with the copy.

About this manual

Let us take the opportunity of thanking and congratulating you on purchasing AVALON. With it, you have acquired one of the most powerful software packages available to create, analyze and edit sounds. We hope that it will facilitate your work as a musician, producer, sound engineer, sound designer, music scholar or teacher and also present you with many new musical opportunities that were previously closed to you. First of all, of course, you have to get to know the program and that's what this manual is all about.

This manual aims to help you understand and fully exploit the many and sometimes highly complex functions that AVALON offers.

If you feel that important information is missing or a function has not been properly explained, please let us know. We will make every effort to remove such errors in the course of the next update.

Manual structure

This manual is intended for use by both the absolute beginner who has never had anything to do with digital sound editing and the professional sound designer who already has a great deal of knowledge and experience of this field.

In order for it to fulfil this purpose, the manual has been structured in accordance with very strict guidelines. This was also necessary to enable us to accommodate all the relevant information within a single handbook.

As a result, hardly any information is repeated. It is thus essential that you read this manual very carefully and attempt to assimilate what you have read.

What follows is an overview of the different sections of the manual as well as an explanation of the conventions used within it. Please don't skip this section - it contains essential information!

Clearly, this manual consists of several parts. The first of these is the chapter you are now reading "About this manual". It contains basic information and represents essential reading for every AVALON user.

The next chapter deals with the subject of "Computer-based sound editing". It represents a general introduction to the subject and deals with the following areas: the theory and practice of digital sound editing, hardware, software and how to install your system. It contains a great deal of general information that will not only help you when using AVALON.

The chapter's section on "The theory and practice of digital sound editing" will be of particular interest to those of you who know little or nothing about the subject but want to use your samplers to do more than simply playback recorded sounds.

It is followed by sections that describe the hardware AVALON uses, how it should be installed and how to set up your system. These sections are an absolute must for anyone who is new to computers or MIDI.

The largest chapter is the "Reference" section. It is also the most important if you wish to make full use of AVALON.

The "Reference" section does not intend to explain fundamental theoretical principles or present general information. Instead you will find concise descriptions of all AVALON's subprograms and the functions they provide.

At the end of the manual you'll find the "Index". If you don't want to read through the whole manual, but need to find a specific piece of information, the index enables you to quickly find the place where a term is mentioned for the first time and explained in detail.

The "Appendix" contains information on individual samplers and their installation.

Conventions

Headings

Please note that throughout this manual references to manual headings and subheadings always appear in double quotation marks, whilst the functions to which they refer do not. For example:

'...This is described in the "XFade Mode" section', but
'...This is the XFade Mode referred to above'.

Abbreviations

Frequently used terms are only written out in full the first time they appear. Afterwards they are usually abbreviated. The most important abbreviations are:

MP	Mapping Page
TDE	Time Domain Editor
FDE	Frequency Domain Editor
SP	Synthesis Page
TC	Time Correction Module
FA	Fourier Analysis
FS	Fourier Synthesis
FM	Frequency Modulation
AM	Amplitude Modulation
WS	Wavetable Synthesis
RM	Ring Modulation
TVW	Time Variant Waveshaping
PD	Phase Distortion
KSS	Karplus/Strong Synthesis
ST	Atari ST computer
STE	Atari STE computer
GEM	Graphic Environment Manager
M•ROS	MIDI Realtime Operating System

Hz	Hertz
kHz	kilohertz
MHz	megahertz
ms	milliseconds
sec	seconds
KB	kilobyte
MB	megabyte
GB	gigabyte

Formats

Unless otherwise indicated, details of memory requirements are always given in bytes.

Unless otherwise indicated, details about sound data are given in the following formats:

Sampling rate	Hertz
Sample length	sample words and/or milliseconds at a sampling rate of 44.1 kHz

Computer keys

Although the program is largely mouse-oriented, many of its functions can also be controlled using the computer's alphanumeric keyboard. When we refer to the use of the computer keyboard the appropriate letters and numbers appear in square brackets.

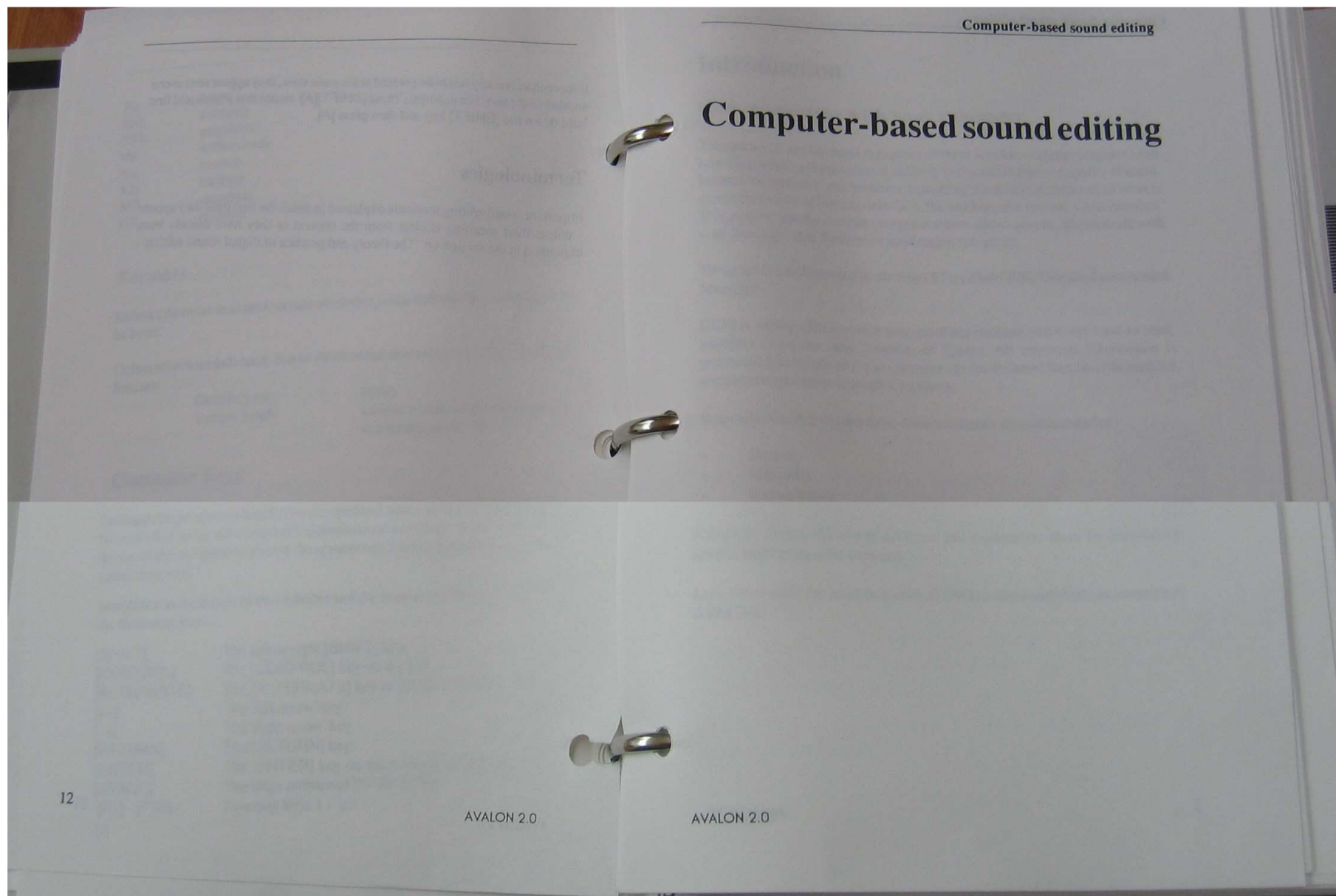
In addition to the letters of the alphabet and the numbers 0 - 9, we also refer to the following keys:

[SHIFT]	The left or right [SHIFT] key
[CONTROL]	The [CONTROL] key on the left
[ALTERNATE]	The [ALTERNATE] key in the lower lefthand corner
[←]	The 'left arrow' key
[→]	The 'right arrow' key
[RETURN]	The [RETURN] key
[ENTER]	The [ENTER] key on the numeric keypad on the right
[SPACE]	The large unmarked [SPACE] bar
[F1] - [F10]	Function keys 1 - 10

If more than one key has to be pressed at the same time, they appear next to one another in the text. For example, 'Press [SHIFT][A]' means that you should first hold down the [SHIFT] key and then press [A].

Terminologies

Important sound editing terms are explained in detail the first time they appear - unless their meaning is clear from the context or they have already been explained in the section on "The theory and practice of digital sound editing".



Introduction

Communicating with your computer

This section is not intended to explain all there is to know about computers and how they work - a subject that is unlikely to interest the great majority of users. Instead we wish to concentrate on describing the different elements of what is generally known as the user-interface, the link between you and your computer. This graphic interface on the computer screen allows you to communicate with your hardware and the current application (program).

The graphic interface used by the Atari ST is called GEM (Graphic Environment Manager).

GEM is highly effective as a user-interface because you never have to read, interpret and enter large columns of figures. All important information is presented graphically and you can carry out the different functions through the simple manipulation of graphic symbols.

Basically, GEM provides three different means of communication:

- **Menus**
- **Windows**
- **Dialog boxes**

AVALON makes full use of all these and expands on them by introducing several interesting new elements.

Let's move on to the most important GEM functions and their counterparts in AVALON.

Pointer



The pointer (also sometimes known as the mouse cursor) on the monitor screen represents your hand within the graphic environment. The mouse or trackball connected to your computer is used to position the pointer and manipulate objects on the screen.

Clicking

Position the pointer at an object and briefly press the left mouse button. This process is known as 'clicking'.

Select / Deselect



When an object is clicked using the mouse, it is then either activated (selected) or deactivated (deselected).

An active **window** (see below) can be recognized by the fact that it has a grey Move Bar (the horizontal bar at the top of the window). The Move Bar of an inactive window is white.

A selected **icon** (see below) is displayed in inverse form (ie black becomes white and vice versa).

In order to select more than one icon with the mouse, hold down the computer's [SHIFT] key and click on each icon, one after the other.

Double-clicking

A double-click involves briefly clicking on an object twice.

A double-click enables you to execute a function or call a window or dialog box.

Dragging



Position the pointer at an object, hold the mouse button down, reposition the pointer and then release the mouse button.

This enables you to move the pointer around the screen and 'drag' an outline of the object behind it. As soon as you release the mouse button, the object will appear at the new pointer position.

Clicking with the right mouse button

This technique is rarely used under GEM. In AVALON it can be employed to call additional functions.

Icons

Icons represent 'things', in the widest sense of the word. They can represent a set of data stored in the computer or a device connected to it. The Disk Icon, for example, represents a disk drive.

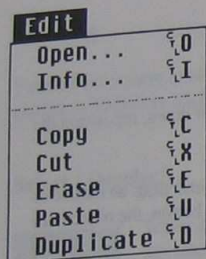
In AVALON, some icons can be freely moved about on the screen, others are 'installed' at a fixed position.

Computer-based sound editing

There are different ways of manipulating icons. You can execute certain functions with a click or double-click on the mouse button. Many icons can also be clicked on using the right mouse button in order to execute other functions or call a dialog box.

Certain functions can also be executed by dragging a moveable icon to another icon on the screen.

Menus



It should be fairly self-evident that a menu allows you to make a selection.

Menus are groups of functions and settings which are found under a common menu heading. If you position the mouse on the menu bar, the menu automatically unfolds and you are then able to select the desired item with the mouse.

There are different types of menu items:

- Functions which are immediately carried out when you click on the mouse at their position (eg Quit).
- Flags which can be directly (de)activated by clicking. An active flag is marked with a 'check'.
- Menu items that branch to another part of the program or a dialog box. These items can be recognized by three dots at the end of the line ('Preferences...').

A lot of menu functions can also be selected using specific key combinations on the computer keyboard. These are given at the end of line of the relevant menu item. The three keys used to switch to the different keyboard levels are represented as follows:

[CONTROL] [^] or [ctrl]
[ALTERNATE] [⌘] or [alt]

Computer-based sound editing

Thus, for example, the menu item 'Open... ^O' means that you can also select the menu function 'Open' using the key combination [CONTROL][O].

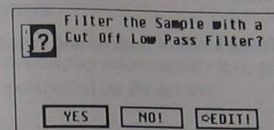
Dialog boxes

As its name says, the moment such a box appears you have entered into a dialog with your computer.

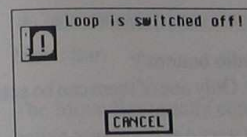
In concrete terms, what this means is that it will not do anything else until you have first made some sort of decision - even if this only involves telling it that it's 'OK' to go ahead with a specific function.

You have to close an open dialog box before you can continue working on the desktop.

There are different kinds of dialog boxes:



- Decision dialog boxes appear when the computer expects you to make a decision, eg when you have to choose between two variants of the same function or a critical decision requires confirmation.



- Information dialog boxes provide you with specific data. You have to acknowledge such messages by clicking on the box's OK button.

Computer-based sound editing

- Settings dialog boxes contain entry buttons and/or switches that affect the system as a whole or data/devices.

In turn, settings dialog boxes contain different kinds of buttons and switches.

- ☒ Draw all Samples
- ☐ Default Overview

- Normal switches:
These can be (de)activated by clicking - with no effect whatsoever on other switches.
☐ Low ☐ Mid ☒ High
- Interdependent switches (also called 'radio buttons'):
At least two switches are linked together. Only one of them can be active at any time.
☒ Start ☒ End Point
- Selection switches:
Here, too, at least two switches are linked. At least one switch must be active, but it can also be more.

Computer-based sound editing

Compared with normal GEM dialog boxes, AVALON's settings dialog boxes also offer an additional feature: so-called 'Flydials'. These can be freely positioned around the screen.

To move a flydial, position the pointer at the circle in its top righthand corner. Drag the flydial to the desired position with the mouse button pressed and then release it.

Most settings dialog boxes can be exited in two different ways:

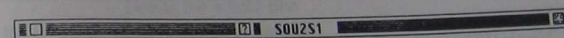
- By clicking on the OK button if you wish to accept the modifications that have been made.
- By clicking on the CANCEL button if you wish to retain the previous settings (and thus cancel any changes).

Windows

Windows display information - text, graphics or both - and can be more or less freely positioned on the screen.

Several windows can be open at the same time, but only one window - the 'top' one - is active. To select a window, click on it.

Windows contain the following elements:



- Move Bar.

The Move Bar usually contains the name of the data it displays. An active window has a grey Move Bar. The Move Bar of an inactive window is white.

You can reposition the window on the screen by positioning the pointer on its Move Bar and then dragging it to the desired position - with the mouse button pressed. When you release the mouse button, the window will reappear at its new destination.

Computer-based sound editing



Close Box.

Clicking on the box in the top lefthand corner closes the window.



Zoom Box.

If you click the box in the top righthand corner, the window is enlarged so that it takes up the whole screen. Clicking on the box a second time returns the window to its previous size.



Resize Box.

Clicking on the box in the bottom righthand corner and holding down the mouse button allows you to resize the window with the mouse. When you release the mouse button, the window takes on its new size.



Vertical/Horizontal Scroll Bar.

These bars allow you to change the area being displayed in the window. Clicking the arrows at the end of the bars moves the display area in small steps. If you keep the mouse button pressed, the display area continues to be moved until it is released. The display area can be moved in larger steps by clicking on the grey sections of the scroll bars instead of the arrows. It is also possible to move the Scroll Box (the white area of the scroll bar) directly. Position the pointer on the scroll box, hold down the left mouse button and release it when the desired position has been reached. If the scroll bar is completely white, there is nothing more to see; all data is already displayed.

Computer-based sound editing

In some windows the right mouse button can be used to modify the zoom factor. Click on the right or top arrow to magnify the display and the left or bottom arrow to reduce it.

The Sample Windows in AVALON also have additional functions. These are described in the "Windows" section.

Entering text

Basically, entering text via the computer keyboard is as simple as using a typewriter.

Click on the place where you want to enter your text. A text cursor will appear and you can begin entering characters.

In addition to the normal character keys used to enter the text itself, the following keys can also be used to correct or modify what you have already entered:

[←]	One character left
[→]	One character right
[↑]	One line up
[↓]	One line down
[BACKSPACE]	Delete character to the left of the cursor
[DELETE]	Delete character at the cursor
[ESC]	Delete the whole line
[RETURN]	Accept the text entered

When typing in a file name you do not need to include its file extension ('.SMP', '.BNK', '.SPC'), it is automatically added on saving or loading; you only need to provide the (up to) eight letters of the file name itself (eg BASSDRUM, not BASSDRUM.SMP).

Value Slider and Keypad

AVALON also provides two other useful methods of entering data, that you will not otherwise find in 'normal' GEM programs.

The first is the Pop Up Slider. It is useful when you want to quickly modify numeric values within a larger range.



Where a decimal number appears between two arrow boxes you can increase or decrease it by simply clicking on the appropriate arrow.

However, clicking on the value itself with the **left** mouse button calls a small window that contains a grey shaded column. This column can be pushed up or down with the mouse like a fader, causing the value of the number displayed to change. When you have set the value you want, you simply let go of the mouse button and the slider disappears.

When dealing with certain values (eg those which have more than four digits and are thus difficult to change accurately), clicking on the number with the **right** mouse button enables you to activate a slider that allows you to make finer adjustments. As a result, however, the range it covers is also smaller.

1	2	3
4	5	6
7	8	9
0	.	C

As an alternative to the slider, you can also decide to enter numeric values via a so-called Keypad, which appears beside the value that has to be changed.

Generally you should be able to set any decimal value very quickly using a combination of these methods.

When you have selected one of these two techniques (via the appropriate mode setting), you don't have to completely abandon the use of the other. You can activate it by clicking on the value that you want to alter whilst holding down the [SHIFT] key.

The theory and practice of digital sound editing

The process of sound creation and editing with digital systems involves the following areas:

- sound events, and their
- recording
- representation
- editing
- analysis
- synthesis

Sound

In order to be able to really understand what a filter, a reverb unit or a complex sound editing system do, you first need to know what sound is and how you can modify it. Without this knowledge, operating a sound editing system is almost like trying to drive a car with your eyes closed.

In this chapter we are thus going to have a closer look at acoustics as well as analysis and resynthesis techniques.

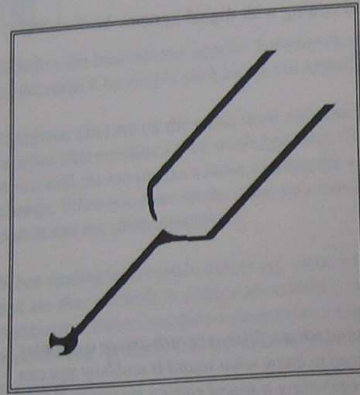
What is a sound?

Initially let's simply call any acoustic phenomenon with a beginning and an end a sound - whether it is a glass breaking, the sound of a tuning fork or a guitar sound.

This brings us to our first question - how does sound reach our ears?

Acoustic events

In the dark ages of the musical past, when quartz-stabilized oscillators did not yet exist and most instruments did not yet automatically tune themselves when they were switched on - in fact, most instruments did not even have switches to be turned on or displays or MIDI ports - a lot of musicians used to carry a rather small insignificant-looking metallic object around with them which was known as a tuning fork.



This acoustically rather modest little instrument has now almost been forgotten in an age of VLSIs, FM, LA and PCM, but we will now get it out of mothballs once again in order to explain some fundamental acoustic principles.

In order to get a sound out of a tuning fork we have to grasp it at its lower end and gently strike it against another object. What we then hear is a not particularly loud but pure sound which eventually fades away.

It may be hard to imagine that such a simple object can serve to illustrate the most complex acoustic processes, but it will soon become clear that this is exactly what it can do.

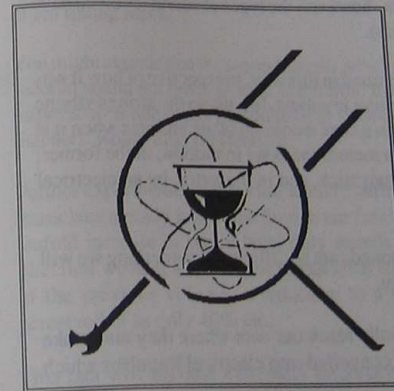
Sound waves

If we had the inclination and the financial resources to undertake a flight into outer space with our tuning fork, we would discover that our experiment cannot be repeated in the vacuum of space; the tuning fork would remain silent.

Clearly there is a direct relationship between the production of sound and the medium in which it occurs. No air means no sound!

But what actually happens when we hear a sound? What factors influence how loud, how high, how 'clear' or 'faint', or how short or long it appears?

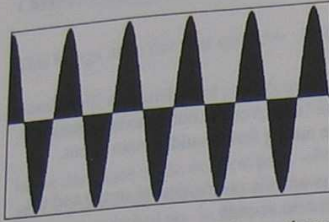
To answer all these questions, let's examine our tuning fork more closely under a magnifying glass - under a very special magnifying glass which is not only able to show us what happens to individual molecules of air but is also capable of slowing down time.



When we strike our tuning fork, the energy released makes it vibrate. Its two metal arms begin to move back and forth; they do this so fast that we are normally barely able to see this taking place.

However, with the aid of our special magnifying glass we can see how the vibrating pieces of metal also set the surrounding air molecules in motion. The molecules of air begin to vibrate in the same way as the arms of the tuning fork move to and fro.

If we were to position ourselves not far away from our tuning fork and record the changes in air pressure while it produced a sound, the resultant diagram would look very much like this:



This graph might say something to those of you who managed to sit through those physics lessons at school without falling asleep: it is a sine wave. In fact, this simple, periodic wave is the basic component of all acoustic processes, however complex they are.

The vibration of the air molecules spreads; starting from the position of our tuning fork, sound waves travel through the room - they are not substantially different from the waves produced in a lake or pond when you throw a stone into the water (incidentally, water is also capable of transmitting sound waves, as is metal, helium and other substances - however, the types of sound involved have completely different characteristics).

Any acoustic phenomenon is transmitted in this way, irrespective of how it was produced and the transmission medium involved. In a piano the strings vibrate when they are struck by hammers. In a flute a column of air vibrates when it is blown. In a drum or a loudspeaker a membrane is set in motion, in the former, as the result of being struck by a drum stick, and in the latter, by an electrical current.

The reason for the fact that all these sounds are so different is something we will examine more closely a little later on.

In all events the sound waves eventually reach our ears where they also make our ear drums vibrate. Here they are converted into electrical impulses which are passed on to the brain.

What actually happens there is something we do not need to consider in greater detail here. It should be sufficient to say that we hear the sound of our tuning fork and are able to recognize it as such.

Amplitude

We now know that the vibrations of our tuning fork can be represented as a sine curve on a graph. When it is depicted in this way, the (vertical) Y-axis represents the sound pressure at a specific point in time. This value is known - not only in the field of acoustics - as the amplitude of the signal.

An increase in the amplitude - logically - also results in an increase in volume.

In order to make a tuning fork sound louder or quieter we have to strike it against another object with more or less force. (Or as our old physics teacher might say: the more energy we input into the system, the greater the amplitude of the resulting oscillation...)

Although the dynamic range of a tuning fork may not appear particularly great, we would nevertheless be able to make an interesting discovery if we were first to listen to one tuning fork and then follow this with the sound of an 'orchestra' of ten tuning forks.

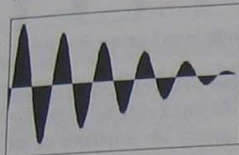
You might assume that the second signal - which involves a tenfold amplification - would sound ten times as loud as the first. In reality, however, the subjective impression is one of only a doubling in volume. This is a fact which other 'test listeners' would also substantiate.

Further experiments of this kind would confirm what mathematically inclined musicians already know: the human ear functions on a logarithmic basis, i.e. a tenfold increase in signal level only sounds two times as loud. In the other direction we hear a halving of signal level as a reduction to only around 70% of the previous volume, a reduction to a quarter of the previous level is perceived of as only 40% etc.

This fact will become extremely important later when we begin to modify sounds.

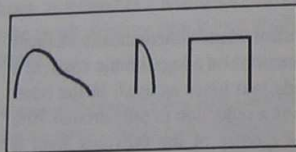
Dynamic range & envelopes

With time the volume of a sound made with our tuning fork gradually fades; the energy is no longer converted into motion but into an insignificant amount of heat and the two arms of the tuning fork eventually come to a standstill. If we were to depict this process in a diagram, it would look something like this:



Clearly the amplitude of the sound has continuously fallen from its maximum level at the start to zero at the end. This is known as the sound's dynamic range. The line which represents this phenomenon (ie the way the volume of a sound fluctuates) could be seen as an outer 'covering' curve which surrounds an inner one (ie the curve representing amplitude) and for this reason it is known as an ENVELOPE.

Naturally curves other than the one depicted here are possible. Clearly, the envelope of the sound produced by our tuning fork offers few possibilities for modification, so here are some examples of the envelopes of other sounds.



Dynamic range is a decisive factor in the evaluation and creation of a sound as it determines its length and thus also its vividness and complexity.

Frequency

With the aid of our special magnifying glass we are not only able to determine that our tuning fork vibrates, but also how often it does so.

We can thus ascertain that it moves 440 times a second, first in one direction and then in the other. This is known as its frequency, which is measured in Hz (hertz = vibrations per second).

What happens if we change this frequency?

The first question this raises is how can this be done. Our old physics teacher would explain here that you only need to reduce the mass to be vibrated while maintaining the same level of energy input. So what do we do? Using a hacksaw we shorten the arms of our tuning fork to half their original length.

We then discover two things:

- the tuning fork now vibrates 880 times a second, and
- the pitch has doubled. Instead of the original A, it now produces a note which is one octave higher.

Clearly frequency and pitch are directly related to one another on a linear basis. A doubling of frequency doubles the pitch.

The relationships between different frequencies

If we were blessed with great patience and an inexhaustible supply of tuning forks, we would now be in a position to create every possible pitch at any level of volume simply by varying the amplitude (through energy input) and frequency (by decreasing or increasing the length of our tuning forks).

Further experimentation would demonstrate that there are interesting relationships between different frequencies. We would discover, for example, that all the frequencies which are integrally related to one another (such as 440 and 880, 100 and 300 Hz) sound as if they belong together and are harmonious. Two frequencies which are fairly close together (such as 440 and 443 Hz) sound 'mellow'. On the other hand, two frequencies which have no clear mathematical relationship to one another (eg 440 and 1370 Hz) sound dissonant.

Computer-based sound editing

The fascinating thing about these otherwise rather boring sine waves with their different frequencies, amplitudes and envelopes is that by combining them in very specific ways we are able to form almost any complex sound we choose. They can be used as 'building blocks' to represent any acoustic phenomenon. They can also - something which will be of extreme importance to us later on - any natural sound can also be reconstructed with the aid of a specific number of sine waves.

Fortunately in an age of manned space travel it is possible to assign the task of analyzing and creating frequencies and amplitudes to a microchip so that we do not need to worry about obtaining huge supplies of tuning forks. Instead we can concentrate our efforts on the fascinating field of digital sound editing.>

Recording

In order to be able to manipulate sound data with the aid of a computer we first need to convert it into a 'machine-readable' form. The problem here is that any sound involves a progressive (analog) signal, whereas a computer can only process discrete numeric values. We thus need to convert our analog signal into data which the computer can understand. To achieve this we use an ANALOG/DIGITAL converter.

Sampling Frequency

The A/D converter takes regular samples from the analog signal input into it and stores them.

In this process it is the speed with which these samples are taken (the sampling rate) which determines the maximum frequency range which can be recorded. As a general rule the sampling rate must always amount to double the frequency range to be recorded. Thus if the highest frequency to be recorded lies in the region of 20,000 Hz, the sampling rate has to be 40,000 Hz.

At the present time the 44,100 Hz used in CDs has become accepted as the standard sampling rate, although, of course, a lower or higher rate can also be used (with a correspondingly smaller or larger frequency range).

Computer-based sound editing

Resolution

Each sample is coded as a number. It should be clear that the gradation between the individual values will become finer - and the authenticity of the digitalized sound greater - the larger the range of numeric values available.

Nowadays most professional samplers encode each sample value in what is known as a 16-bit word - ie in a sequence of 16 zeros and ones.

As this provides 2 to the power of 16 (ie 65,536) possible values, adequate resolution can be achieved; it is almost impossible to distinguish between the original sound and a digital copy.

All samplers and music computers work on these principles. The only things which vary - often quite considerably - are their quality, speed, additional functions and naturally their price.

Technical problems

However straightforward the theory of sampling and digital sound editing may seem, in practice difficult problems can arise. Some of these are outlined below in order to help you avoid them.

Memory

Imagine for a moment that we want to create a really authentic grand piano sound with our sampler. We thus decide to sample each of the instrument's individual notes in CD quality - and allocate 6 seconds for each note (which naturally has to fade away completely); of course, this all has to be done in stereo.

Let's start...

As explained above, every second our sampler takes 44,100 samples from the sound. Each of these values contains 16 bits, ie 2 bytes, so we need 88,200 bytes per second.

Each note lasts 6 seconds, which means we need 528,600 bytes for each note.

Computer-based sound editing

For stereo recording the amount of memory required is doubled, ie 1,057,200 bytes.

Any grand piano worthy of the name has a total range of 7 octaves ($7 \times 12 = 72$ notes); we have now reached a total of 76,118,400 bytes and have only digitalized one 'slice' of the piano's sound spectrum; clearly its tonal structure is greatly influenced by the way in which the keys are struck and to take this into account we really ought to sample the complete keyboard using four different velocities...

At a total figure of 304,473,600 bytes we now come to a halt only to be told by a well-informed friend that this amount of data is the equivalent of 130,000 densely printed pages of type - undoubtedly much more than we ever imagined we would need for our (now admittedly excellent) grand piano sound.

What can we do in order to reduce the memory capacity required?

Whatever we do, we lose data. The question (as so often) is which data is superfluous.

Resampling

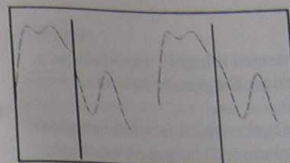
With the appropriate software we could, for example, create a new sample from one which is already available, however, this time with a lower sampling rate - possibly of only a half. We would thus lose half of the frequency range, but in relation to many low frequency sounds (eg bass drum, percussion) this would be acceptable.

Looping

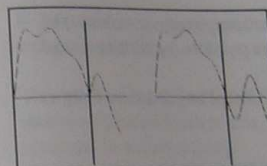
This is the most widely known technique for reducing memory requirements. It allows us to repeat a freely definable section of a sound as often as we choose in order to give the impression of a longer sound. Such a loop could, for example, be repeated three times - or also for as long as we keep the appropriate key pressed.

In defining such a loop, several important points have to be borne in mind.

Computer-based sound editing



- **Transition points:** Loop start and end points ought to have the same values in order to avoid clicks. Generally it is advisable to set loop points at a zero transition point as these are the easiest to deal with.



- **Phase & gradient:** A smooth, continuous transition should always be maintained between the two loop points. This means that a 'wave valley' has to follow a 'wave hill' (correct phase), and the gradient at both ends of the loop should be almost the same.
- **Length:** The optimum length for a loop depends on the sound material involved. Simple, periodic sounds (sine waves, eg a flute) can be recorded very well using short loops. More complex, aperiodic sounds such as the human voice, on the other hand, require a correspondingly long loop. Long loops of this kind are more difficult to find, especially as volume and tone often change between the loop start and end points. In such cases it is advisable to work with automatic loop find and loop adjust functions.

Representation

The way in which digitalized sounds are represented is highly important, as a clear and understandable display is the prerequisite for precise editing.

Presenting sound data as a long series of numbers - which is what samples actually are - would be the worst conceivable solution. Clearly it makes sense here to exploit the possibilities offered by a graphic user interface.

The following means of representing sound data have proved themselves in practice:

- **Time/amplitude matrix**
Time is represented by the horizontal axis and amplitude (ie volume) by the vertical axis. Various forms of scaling are possible (eg milliseconds/ per cent or sample words/sample values).
This form of representation is used in AVALON's Time Domain Editor.
- **Time/amplitude/frequency matrix**
After a Fourier analysis (see below) it is possible to display the individual frequency bands that make up a sound.
Here, time is represented by one axis, frequency bands by the second and their amplitudes by the third. This gives rise to the characteristic 'wave mountains' of spectrum lines that - in contrast to a time/amplitude grid - provide more precise and clear information on the structure of the sound.
The time/amplitude/frequency matrix can be displayed in two ways:
 - as a 3D graphic In this case you can examine the sound spectrum from a freely definable viewpoint outside the 'wave mountain'. This form of representation is implemented in AVALON's Frequency Domain Editor as 3D Graphic Mode.
 - as a spectrogram In this case the amplitude axis is represented by variation in display density.
This form of representation is implemented in AVALON's Frequency Domain Editor as Spectrogram Mode.

Editing

A computer can be used to filter, distort, fade in and fade out sounds. In the final analysis, however, all these functions only really involve specific algorithms (mathematical formulas) being applied to huge sets of data. The computer doesn't 'know' what it is doing with the sample at all, and it is thus up to the user to make sure that these algorithms are applied in a sensible way. Theoretical knowledge and practical experience are equally important here.

Technical Problems

In theory everything appears fairly straightforward, but unfortunately there are technical problems.

The problems caused by a lack of memory capacity have already been mentioned. In addition to this, there is also the problem of the processing times involved in sound editing.

Many of the complex computer operations which make digital sound editing attractive in the first place inherently involve a great deal of processing and thus considerable processing times. Until recently, for example, a Fourier analysis of a sound lasting one second still required over an hour of processing on a 16-bit computer. However, a Fourier analysis of this kind is the prerequisite for the selective editing of individual frequencies within a sound (please note: only the prerequisite; all the additional calculations also require a considerable amount of processing time).

Through efficient programming it has now become possible to work with tolerable processing times using 'ordinary' 16-bit computers. Fourier analyses, resynthesis and complex macros can, however, not be carried out in realtime (ie without processing times being noticed as such) without additional hardware enhancement.

It is thus advisable to continue only utilizing time-consuming functions on those sounds and sound sections which really demand them (attack phase, particularly critical points within a longer sample) and to work in a carefully planned way - ie by making sure in advance that you have thought out what you want to do and how you want to do it.

Analysis

As we already pointed out earlier, the creation and editing of digital sounds using a computer is really a rather simple matter. Columns of numbers are merely copied or moved, multiplied by specific values and merged with other columns, etc. The computer 'knows' absolutely nothing about sound and its structure.

Here, computer-based analysis goes an important step further: it simulates what our brains do with sounds. It splits a sound up into its constituent parts - the individual frequency bands - and evaluates the results.

This form of analysis becomes very interesting when used in combination with the appropriate editing functions. Individual frequency bands can be moved up or down, deleted or shifted to another position - and all this with little or no effect on the other frequencies around them. Functions like these can also be combined to create macros that represent extremely powerful tools.

The fast Fourier analysis and resynthesis of a sample are the most complex data processing operations carried out by AVALON. You should expect the analysis of sound material to take at least five times as long as the original sound length. It is thus advisable to proceed as economically as possible - ie carefully select only the section of the sound that you really want to work on before you subject it to analysis prior to editing in the Frequency Domain Editor.

Synthesis

The music press and musicians themselves still tend to somewhat pedantically differentiate between samplers and synthesizers, although both types of equipment create sounds using digital processors and the appropriate algorithms.

The reason that is often given for this differentiation is the fact that samplers basically work with recorded natural sounds whilst synthesizers 'create' their own.

In fact, however, it is now the case that many synthesizers are equipped with a basic repertoire of natural sounds or sound fragments, and every manufacturer provides an editing system of some kind (filters, envelopes, etc).

So, why shouldn't a sampler be equipped with synthesized sounds? After all, it is no great problem for a computer to calculate a waveform, it is only a question of whether it has access to the appropriate algorithms.

AVALON provides you with a variety of synthesis techniques. They are described on the following pages.

General

Although the synthesis techniques described here may be rather different, they do have certain things in common.

Almost all forms of synthesis involve sound-creating and sound-modifying elements. In synthesis, a sound-creating component is known as a generator and a sound-modifying component is known as a modulator.

However, this difference is purely functional. A sine wave can, for example, be used both as a generator (as a sine generator in the reconstruction of a soft natural sound) or as a modulator (to influence the harmonic content of another wave).

Fourier synthesis

According to the theorem developed by the French mathematician Jean Baptiste Fourier, any complex waveform can be represented as a combination of different sine waves.

This is the principle that underlies the highly complex Fourier analysis used in AVALON's Frequency Domain Editor. Its counterpart here - as a synthesis technique - is represented by the Fourier generator. It provides a series of sine generators whose phase positions and frequency and amplitude relationships can be modified. What is involved here is the digital equivalent of the church organ principle.

Wavetable synthesis

This technique is based on the crossfading of waveforms.

Wavetable synthesis is always based on at least two different waveforms.

One period of each waveform is recorded in a table. This table is then read out in a cyclical fashion. In this process

- the read-out rate determines the frequency of the output signal, ie the pitch, and
- the shape of the curve in the table determines the spectrum and thus the resultant sound.

In wavetable synthesis it is possible to arrange as many of these wave tables as desired along the time axis. The tables are faded into one another in chronological order, which results in a dynamic sound structure.

Frequency modulation

In this form of synthesis the frequency of one waveform (the modulator) influences the frequency of the other (the generator).

The following applies:

If the frequency of the modulator is below 30 Hz, the modulation is perceived as vibrato (similar to the tremolo effect achieved by amplitude modulation).

However, if the frequency of the modulator is close to that of the carrier frequency, a rich sound spectrum is produced. The characteristics of this spectrum are influenced by

- the frequency relationship between the two signals. Non-integer relationships (eg 1:1.3) lead to disharmonic frequency spectrums, and
- the amplitude of the modulator. The higher it is, the richer the resulting sound.

In AVALON, the frequency modulation module provides a fixed sine generator as its generator. It only requires an input signal as modulator.

Amplitude modulation

In this form of synthesis the amplitude of a waveform (the modulator) affects the amplitude of another waveform (the carrier).

The following applies:

If the frequency of the modulator is under 30 Hz, the modulation is perceived as tremolo; the volume of the resultant audible signal is affected by the amplitude of the modulator.

However, if the frequency of the modulator is close to that of the carrier frequency, modulation affects the audible spectrum much more strongly. The reason for this is the fact that the volume of the audible signal is influenced by a frequency that is already audible itself.

The characteristics of this spectrum are influenced by the frequency relationship between the two waveforms and also by the amplitude of the modulator.

Ring modulation

Ring modulation multiplies two input signals. This means that relatively simple waveforms can be combined to create a complex output signal.

This form of synthesis seriously alters the spectral content of the two sounds. As a result, it is possible to create disharmonic sounds, eg as required for bell sounds etc, in a simple way.

Avalon's ring modulator carries out multiplication in a mathematically correct fashion. This means that each sample value of the input signal is combined with the corresponding sample value of the other signal in accordance with the rules of multiplication.

Digital ring modulation of this kind allows very much more subtle sound modification than the analog methods that were previously available.

Time variant waveshaping

The waveshaping technique involves passing a signal through a non-linear curve. The original signal is then distorted - in a way that depends on the properties of the curve selected.

This technique can result in a sound spectrum being very dramatically changed; a large number of partial tones are added to the original signal. This phenomenon is widely known as distortion (eg in specific types of amplifier).

It is very difficult to predict how the form of the curve will change the input signal. For this reason, useful results are rarely achieved by creating the curve by hand - as is also the case when you attempt to 'draw' a waveform. This is why previous attempts at implementing waveshaping have not been very successful.

The curves used in AVALON's waveshaping module have thus been created on the basis of specific mathematical algorithms.

When a signal is passed through such a curve, new frequencies are added to the original sound.

The amplitude of the original signal determines whether it passes through the whole curve or only parts of it. If the amplitude of the input signal becomes smaller, it only passes through parts of the curve.

This effect can be used to create sound spectrums.

If several curves are arranged along the time axis and faded into one another with the aid of wavetable synthesis, the opportunities for dynamic sound shaping are increased on a cumulative basis. This technique is referred to as time variant waveshaping.

Phase distortion

To a certain extent, this involves a combination of the waveshaping and frequency modulation techniques described above.

Phase distortion is based on the same algorithm as frequency modulation. Here, however, the frequency of the carrier is missing.

This has the following consequences:

An FM module without a modulator only creates a sine curve.

A PD module without a modulator produces no output signal - as the carrier frequency is zero. Thus, in phase distortion, the frequency of the output signal always follows the frequency of the modulator.

As a result, the modulator determines the frequency of the output signal. A sine wave is used as the waveform for the generator - as is the case in frequency modulation.

Actually, PD is thus only a special case or a slimmed down form of FM synthesis. Disharmonic spectrums based on non-integer frequency relationships, for example, are completely impossible as the frequency relationship is always the same as the modulator frequency.

The affinity between phase distortion and waveshaping is due to the fact that the processes are mathematically very similar; the difference is that in PD the algorithm affects frequency whilst in waveshaping it influences amplitude.

Karplus-Strong synthesis

In Karplus-Strong synthesis, noise runs through a delay circuit. At each step, the current value of the internal noise generator is averaged with the neighbouring value and written in its place. The output signal is produced by this averaging process.

This process has an effect similar to a lowpass filter. Thus, the original signal is continually dampened in the delay circuit so that it gradually fades away. The length of the decay period depends on how many values are contained in the delay circuit.

Computer-based sound editing

The delay factor used is thus also the only sound-modifying factor. At the same time it also provides you with a rough idea of the period - and thus the basic frequency - of the output signal.

Despite the simple arithmetic basis of Karplus-Strong synthesis, the sounds produced are rather interesting and have their own characteristic quality. It is similar to the type of sound we associate with percussive or plucked instruments. The results of KSS are restricted to sounds of this type, however.

Fractal synthesis

Of all the synthesis techniques introduced here, fractal synthesis is probably the most recent. It is based on the results of chaos theory, an area of study that is becoming increasingly important in many different fields.

What follows is a short introduction to the theory of fractals to help you understand what this technique involves.

Fractality is one of the most important properties of the physical universe. It is to be found, for example, in the structures of leaves, trees, mountain chains and complete landscapes.

Recently, more and more fractals are being created in computers; usually in graphic form. The most well-known of these 'synthetic' fractals is the figure found by Benoit B. Mandelbrot. It has now become a kind of mascot for chaos research.

An important aspect of fractals is their recursiveness. This means that a large structure (eg a fern) has substructures which are similar to the whole. In turn, these substructures also have their own substructures, and so on. Interrelationships of this kind produce extremely attractive structures.

This can also be seen in another way - one that is more useful in the kind of synthesis we are looking at here. All fractal constructs are based on one - rather simple - microstructure that replicates itself again and again and eventually grows into a complex whole.

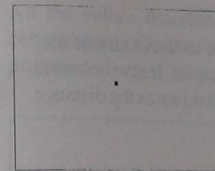
Computer-based sound editing

This enables computer graphics systems to calculate extremely complex and natural-looking structures from very simple geometric shapes. Clearly, this is also a form of synthesis - and there is absolutely no reason why it shouldn't be applied to graphics **and** music.

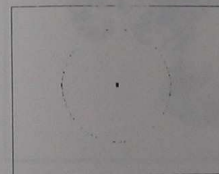
We will now describe how fractal synthesis is implemented in AVALON. It is only one of many possible approaches, but one that is highly promising.

It is rather difficult to explain how fractal systems function in words alone. We would thus encourage you to experiment with fractal synthesis - in this case, using your intuition is superior to a purely theoretical approach to the subject.

The starting point for this synthesis technique is a matrix. Within this matrix we want to create circles that should in turn create copies of themselves, that in turn ... Well, the principle should be clear.



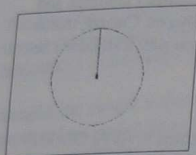
We start off with a single point. This is the centre of the first circle.



We then determine the radius of the circle with a second point.

The radius is 'drawn'. As a result, a vector is created, which is referred to as the base vector. Where this should occur is something we can determine more or less at will. We are also able to turn the circle at a later point in time and thus modify the direction of the base vector.

We have now determined the properties of a so-called 'mother' circle.

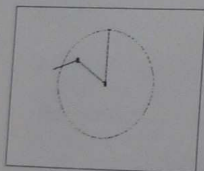


We now fix another point. It serves as the basis for the creation of a 'daughter' circle.

The relationship between the mother and daughter circles can be described as follows:

- The new point is the centre of the daughter circle.
- The relationship between
 - the radius of the mother circle and
 - the distance from the centre of the mother circle to the centre of the daughter circle defines the radius of the daughter circle. It also becomes a hereditary property of the daughter circle. This is known as the distance distribution relationship.

In simple terms: The nearer the daughter is to the mother, the smaller it is, and the further away the daughter is, the larger it is. This also automatically applies to the 'grandchildren' - they are accordingly even smaller or even larger.

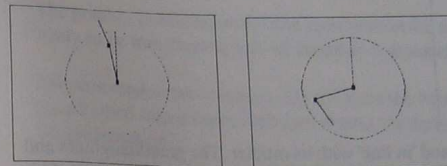


- The angle produced between
 - the base vector and
 - the new vector joining the centre of the mother circle with the centre of the daughter circle becomes a hereditary property of the daughter circle. This is known as the reference angle.

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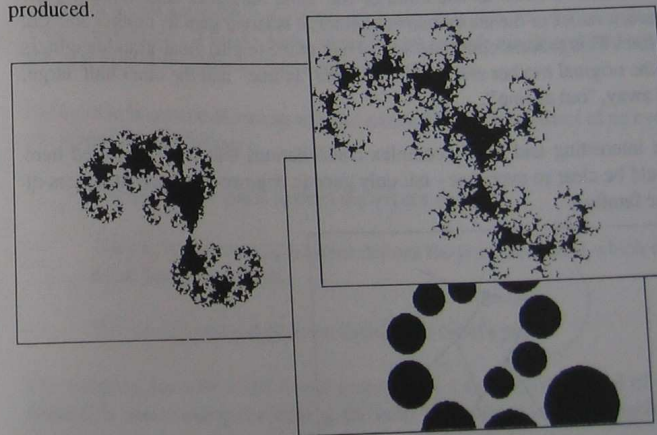
AVALON 2.0

In simple terms: The larger the angle between the 'mother vector' and the 'daughter vector', the greater the angle between the 'daughter vector' and the 'granddaughter vector'. If, however, the daughter is 'in line' with its mother, this will also apply to the granddaughter.



These relationships form the basis for the mathematical formula that is used to create the fractal. When it comes down to it, the process is like a genetic family tree of circles - the only thing that has to be determined is the number of generations.

All the values described above can be varied - sometimes with dramatic consequences for the family history. Spirals, coils or groups of circles are produced.



But just like in the field of genetics, this process only becomes really interesting when a mother has several daughters. The important property of all daughter circles is then not only that they carry their own relationship to their mother in their 'genes', but also details of the relationship of all the other daughters.

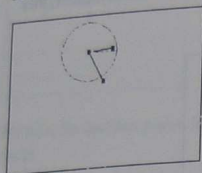
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Computer-based sound editing

An example:

A mother has two daughters.

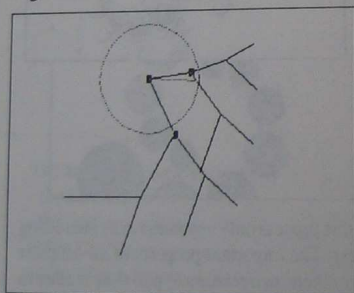


One of them is very close and 'in line' with its mother. The granddaughters and great-granddaughters should thus be increasingly smaller and closer to their mothers - and always 'in line'.

The second daughter, however, stands at a sharp angle to its mother and is very far away. As a result, its 'genes' carry the information 'larger' and 'further away from the original line'.

This situation leads to one child of the 'close' daughter also inheriting the characteristics of the unorthodox daughter; it is larger than its mother and 'out of line'. This process continues so that half of the (eight) great-granddaughters of the original mother are 'small, close and "in line"' and the other half 'large, far away, "out of line"'.

The interesting and highly complex constellations that can be created here should be clear to everyone - not only genetic engineers and the members of large families.



Computer-based sound editing

When producing fractals with AVALON, you are provided with a 'mother' and three 'daughter circles'. You will soon discover that using these relatively few variables allows you to create extremely complex structures. You will find a series of very attractive fractals on the disk of examples.

The graphic 'family histories' that you create in the way outlined above are then ready to be transformed into an acoustic or musical model.

This conversion process basically involves the two-dimensional matrix being transformed into a time/amplitude matrix, as described in the section headed "Representation".

In order to do this, the first thing required is the scaling. You have to decide how the fractal graphics are converted into frequencies and time values.

The scaling of the horizontal axis is determined by the length of the sample to be created.

The vertical axis is always fixed as the complete range of audible frequencies (which is scaled logarithmically here). However, it is also possible to limit the 'conversion range'. If this is done, all the circles outside the determined range are faded out; the frequency band created is narrower.

Each circle is converted into an acoustic event. The characteristics of an event are determined as follows:

- The size of the circle defines the event's volume.
- The circle's horizontal position defines the point in time at which the event becomes audible.
- The circle's vertical position defines the event's pitch.

The simplest form of event would naturally be a sine wave. Instead of this, AVALON uses a triangular wave as the basis for events in order to enable the creation of more complex sounds.

Computer-based sound editing

You can also vary the characteristics of events by determining the following:

- the length of events and their
- attack and release behaviour. This involves a simple attack/release envelope.

If you choose to make the total length of the new sound very long and keep the length of the individual events short, it is also possible to create quasi-musical structures (that could be regarded as 'fractal melodies'). In this process the dividing line between sound creation and composition is very thin.

There are innumerable possibilities for applying this synthesis technique. However, only a small number of the conceivable applications correspond to traditional concepts of music and sound creation. Depending on the complexity, direction and length selected, fractals can sound aggressive or meditative, amusing or threatening, excited and nervous or melodic and harmonious.

Using this feature as a sound generator is only one conceivable application. Used as a modulator, complex fractal structures can add a touch of vitality to static basic sounds.

Subtractive synthesis

Subtractive synthesis can now be regarded as an almost classic technique. For many years it was the only technically feasible method of creating sounds electronically.

Subtractive synthesis involves passing a waveform through a filter. The gradient and quality of the filter have a decisive influence on the quality of the sound.

Lowpass filters are generally used. Naturally, however, high pass filters or a combination of filters can be applied.

Computer-based sound editing

Enveloping

What this involves is not a synthesis technique in the strictest sense, but a means of modulating signals.

Envelope generators create relatively slow curves which are inaudible themselves, but can control changes of volume or frequency content.

The best-known form of envelope generator is the four-part ADSR envelope (attack/decay/sustain/release). However, a digital envelope generator does not have to be limited to a modest four or eight variables - it enables you to create curves using almost any number of salient points. The values between the points are then interpolated into a curve.

Hardware

You need the following hardware to run AVALON:

Computer

At present AVALON can be run on the following ATARI ST series computers:

- 520 ST+
- 1040 ST
- 1040 STE
- 1040 STF
- MEGA ST 1
- MEGA ST 2
- MEGA ST 4
- MEGA ST STE (only with the cache switched off)
- STACY

Monitor

AVALON will operate with monochrome monitors with a resolution of at least 640 x 400 pixels. It has been tested on the following monitors:

- ATARI SM 124
- ATARI SM 194
- MATRIX 110

Its use with other monochrome monitors should not present any problems.

If you make extensive use of the Time Domain Editor and the 3D functions of the Frequency Domain Editor - particularly to display and edit very long samples - we recommend that you use a large screen monitor.

Colour monitors are not supported.

Mass storage

AVALON supports all the mass storage units that can be operated by your computer.

Here you can choose between

- 3.5" disks (floppies).
They represent a convenient, safe storage medium. However, in the light of the enormous amounts of data which rapidly accumulate during sound editing you should definitely consider acquiring a 60 or 100 megabyte
- hard disk
that is also very much faster than a floppy disk drive - or a
- removable disk system, which pays for itself once you have over approximately 120 MB of data.

AVALON itself is supplied on a double-sided disk as the computers mentioned above all have disk drives capable of reading and writing to both sides of a disk.

Samplers

The samplers that AVALON supports are listed in the "Appendix" at the end of this manual.

Cables

In addition to the relevant power cables, you will also need the following:

- MIDI cables (2 per sampler),
- to operate an SCSI controller: a 25/50 pin ribbon cable, and
- audio cables.

Although not absolutely essential to run AVALON, you can purchase additional equipment that will make your work with the program much easier. This includes

16-bit D/A Board

The 16-bit D/A Board enables you to playback your samples - over the full frequency range with no distortion - without first having to send them back to the samplers via MIDI. This means that you can immediately check the results of your work.

If you use a D/A Board to monitor your work in this way, you naturally need to have an amplifier available. A normal hi-fi system or an instrument amplifier can be used.

M•ROS MIDI processors and expanders

The MIDI Realtime Operating System (M•ROS) is a computer operating system extension developed by STEINBERG.

STEINBERG offers various MIDI expanders that are supported by all M•ROS programs.

Computer-based sound editing

Large amounts of data have to be transferred when samples are transmitted via MIDI. As a result, transmission errors often occur when several samplers are connected in series. Thus, the most sensible application for MIDI expanders is in the simultaneous operation of several samplers.

In the 'MROS' folder on the program disk you will find a so-called driver (with the extension '.DRV') for each MIDI expander supported by M.ROS. These drivers are part of M-ROS and should not be confused with AVALON's sampler drivers (see below).

Currently the following equipment with additional MIDI interfaces is supported:

- SMP 24
- MIDEX
- MIDEX+
- UNITOR

If you use one of these machines, copy the appropriate driver from the 'MROS' folder of the relevant Program Disk to the folder/disk where your working copy is located. You should only install the drivers for the equipment you actually have.

Some basic points about MIDI

In 1983 MIDI was agreed by the world's leading instrument manufacturers as the standard for the transmission of musical data. This occurred at a time when practically nobody thought about data exchange between computers and samplers as both affordable samplers and the powerful computers required did not exist. This was rather unfortunate as the transmission rate agreed, which was then fully adequate, has in the meantime proved to be far too low when the transfer of samples is involved.

MIDI is thus - for our purposes - frustratingly slow; it can take up to 12 minutes to transfer a 10 second stereo sample from a sampler to a computer (compared with only 8 seconds from a hard disk).

Computer-based sound editing

We would thus advise you

- only to transfer samples via MIDI when it cannot be avoided; in other words, when you want to transfer a 'freshly' recorded sound to the computer or to send the finished, optimized sound back to the sampler.
- only to transfer what you really need. In order to avoid longer than necessary waiting times, before you transfer a sound, use TRUNCATE (if your sampler offers this function) to cut out the parts you do not need.

SCSI controller

This controller can be connected to the DMA port of the computer and makes it possible to transfer data to a sampler at much higher speeds.

In the case of an AKAI S1000, for example, the speed of data transmission is increased by a factor of approximately 250 compared with a normal MIDI sample dump. This means that a sample that takes more than four minutes to transfer via MIDI can be transferred in only one second using the SCSI interface.

For professional users, eg in the post-production field, this represents an immense time-saving and is well worth the investment.

Key expander

A key expander enables you to connect several keys to the ROM port of your computer at the same time. You then no longer have to switch off the computer and change keys when you want to load another hardware-protected program.

Software

Program files

AVALON consists of the following files:

Folder	File(s)	Type
-	AVALON.PRG	main program
AVALON	AVALON.IMG	program file
AVALON	AVALON.PAC	program file
AVALON	AVALON.SET	basic settings
DRIVERS	(various).DRI	sampler drivers
MROS	MROS x_xx	MROS version x_xx
MROS	(various).DRV	MROS device drivers
OVERLAY	FREQEDIT.OVL	Frequency Domain Editor (FDE)
OVERLAY	SYNTHESE.OVL	Synthesis Page (SP)
OVERLAY	TIMECORR.OVL	Time Correction Module (TC)
WAVES	(various).WAV	Preset waveforms for SP
SHAPES	(various).TVW	Preset shapes for SP
CONNECT	(various).CON	Example connections for SP
RUBBERS	(various).RUB	Example rubberbands for FDE
SAMPLES	(various).SMP	Example samples
SPECTRAL	(various).SPC	Example spectrals for FDE

Program structure

AVALON consists of various modules. This means that you are not dealing with a single program but with several subprograms that are able to exchange their data and thus complement one another.

The advantage of this modular concept is that the less frequently used sections of the program only need to be loaded into memory when required and can also

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be removed again. This enables the limited memory available to be used in an effective way.

The various modules are arranged on several different levels. Their complexity and power increases, the 'higher' you go into the program.

Level 1: Mapping Page

Level 1 is the Mapping Page. It contains all the functions that transfer sample data - whether from or to floppy disk, hard disk and sampler or within the computer's memory. From this screen, sample data can be moved, copied, erased and transferred, but not edited.

Level 2: Time Domain Editor and Synthesis Page

Level 2 consists of two subprograms: the Time Domain Editor and the Synthesis Page.

In the Time Domain Editor, you are able to edit sounds, which are represented on a two-dimensional basis (x-axis: time, y-axis: amplitude, ie volume). Sections of a sound (blocks), individual sounds and stereo sounds can be displayed, edited and copied in a variety of ways.

All modifications are carried out directly on the sound being edited and are thus transferred back to Level 1 (Mapping Page). Please note that unlike a synthesizer editor there is no edit buffer available. It is only possible to cancel the last modification.

The Mapping Page and the Time Domain Editor form AVALON's central core. These modules are resident, ie they cannot be removed from memory.

The Synthesis Page provides you with numerous sound synthesis and editing functions. They can be combined as desired in order to emulate popular types of synthesizers or to produce completely new sounds.

Sounds created on the Synthesis Page can then be transferred to the Time Domain Editor for further editing - or back to the Mapping Page.

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Computer-based sound editing

If the Synthesis Page is not currently in memory, it is automatically loaded when you click on the Synthesis Icon on the Mapping Page.

Level 3: Frequency Domain Editor and Time Correction

Level 3 consists of the Frequency Domain Editor and the Time Correction module. Both these modules can only be called from the Time Domain Editor.

The Frequency Domain Editor splits a complex sound into its component parts (sine waves) on the basis of a Fourier analysis and then displays it in three-dimensional form (x-axis: frequency, y-axis: amplitude, z-axis: time).

In the Frequency Domain Editor modifications can be made on three levels:

- in the pixel range: modification of the volume of an individual frequency within the smallest possible time segment.
- in the micro range: modification of an individual frequency along the complete length of the time axis or modification of all frequencies within the smallest possible time segment with the aid of what we call 'rubber bands'.
- in the macro range: modification of the complete frequency/amplitude/time system using 3D macros.

In order to reconvert a frequency range back into an audible sound it has to be resynthesized. Afterwards it is to be found in the two-dimensional time/amplitude system of the Time Domain Editor (Level 2) and can there be edited further or alternatively archived or transferred to a sampler via the Mapping Page (Level 1).

If the Frequency Domain Editor is not currently located in memory, you can load it using the Call FD-Editor option in the File Menu of the Time Domain Editor.

The Time Correction module enables you to shorten or lengthen a sample located in the Time Domain Editor without changing its pitch. The algorithms used result in a quality and processing speed that equals those of very much larger systems and enable you to carry out both very subtle and dramatic modifications.

Computer-based sound editing

The module always shortens or lengthens the sample in the active window of the Time Domain Editor.

The Time Correction function can also be applied to a specific section of a sample using the Block Time Correction option. This can be very helpful, for example, in the correction of a particular section of a long voice passage.

If the Time Correction module is not currently in memory, it is automatically loaded when you call the Time Correction option in the File Menu of the Time Domain Editor.

Together, these modules form a highly effective system for storing, transferring, analyzing and synthesizing sounds, and they thus fulfil practically all the demands of modern digital sound editing.

Installing your hardware

What follows is a description of how to set up a fully equipped system (computer, monitor, hard disk, MIDI expanders, key expander, samplers, D/A Board). Simply ignore the description of the equipment that is not part of your system.

- Connect all your equipment to the mains electricity supply.
- Connect the computer to the monitor with the monitor cable.
- Connect the hard disk to the computer.
- If you are using a MIDI expander, connect it to the computer. This process is explained in the expander's instruction booklet.
- Connect
 - the computer's MIDI OUT socket to the sampler's MIDI IN socket, and
 - the sampler's MIDI OUT socket to the computer's MIDI IN socket.

- If you are using a MIDI expander, connect
 - the expander's MIDI OUT socket to the sampler's MIDI IN socket, and
 - the sampler's MIDI OUT socket to the expander's MIDI IN socket.
- If you are using a D/A Board, insert the AVALON key (the small plastic box) into the port on the lefthand side of the D/A Board. Ensure that the 'THIS SIDE UP' marking is facing upwards.
- If you are using a key expander, insert the key in a free port. If you are also using a D/A Board, insert its ribbon cable into the port marked 'Identical to ROM Port'.
- Insert the key/key expander or the D/A Board's ribbon cable
 - in the ROM port on the lefthand side of the computer housing, or
 - in the port of the MIDI expander that is identical to the ROM port (this applies for the MIDEK and MIDEK+).
- Connect the samplers and/or D/A Board to your amplifier. Turn the amplifier's volume control right down to avoid any initial damage caused by excessive signal levels.
- Turn the MIDI expander, hard disk, monitor, computer, samplers and amplifier on - in this order. (The D/A Board, MIDEK, MIDEK+ and UNITOR do not have to be switched on.)

Installing the software

Installation to floppy disk

- The first thing you should do is create working copies of the original disks, as described under the heading "Backups" at the beginning of this manual.
- Insert the working copy of Program Disk I into the disk drive.
- Open the disk's directory with a double-click on the Disk A Icon.
- Find the file entry 'AVALON.PRG' in the directory and start the program with a double-click on the file name.
- If you want to load one of the external program modules (overlay modules), you will be asked to change the disk in the disk drive. When this occurs, insert your working copy of Program Disk II in the drive (see the section headed "Pages").

Installation to hard disk

- Insert Program Disk I in the computer's disk drive.
- Open the disk's directory with a double-click on the Disk A Icon.
- Open the directory of the hard disk with a double-click on the relevant icon (eg Disk C).
- Create a folder for AVALON on the hard disk using the New Folder function in the File Menu.
- Copy all the files in the drive A directory to the new folder on the hard disk.
- When all the files have been copied, insert Program Disk II in the computer's disk drive and press the [ESC] key on the computer keyboard. The directory of the second program disk will appear.

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- Copy the 'OVERLAY' folder in drive A to the new folder on the hard disk. The other files do not need to be copied or can be copied to another location on your hard disk; they are not directly needed by the program itself.

It is important that

- the file 'AVALON.PRG' and
- the folders 'AVALON', 'DRIVERS', 'MROS' and 'OVERLAY'

are always located at the same level, ie either in the root directory or in the same folder.

Registering AVALON as an application

Depending on the operating system you use, it is possible to instruct your computer to treat AVALON as a standard application. This means that each time you open an '.SMP' file (in other words, a sample) with a double-click on the mouse or by selecting the appropriate menu function, the following will occur:

- AVALON will automatically be started,
- the selected sample loaded and then
- transferred to the Time Domain Editor, so that you can immediately begin editing it.

At present, registering AVALON as an application is possible under

- TOS 030 (the operating system of the MEGA STE and the TT)
- Gemini
- Neodesk

Computer-based sound editing

The following steps have to be taken in order to register AVALON as an application:

- Select 'AVALON.PRG'.
- Call the menu option 'Application'.
- A dialog box will appear, in which you inform the operating system that AVALON.PRG is responsible for file names ending with the extension '.SMP'.
- Save the new settings by calling the appropriate menu option.
- Press the reset button at the back of your computer. AVALON is now registered as a standard application.

Technical hints

Hardware

Please take the usual safety precautions involved in operating electrical equipment. Avoid high humidity, powerful magnetic fields, direct sunlight, extreme temperatures and strong vibrations.

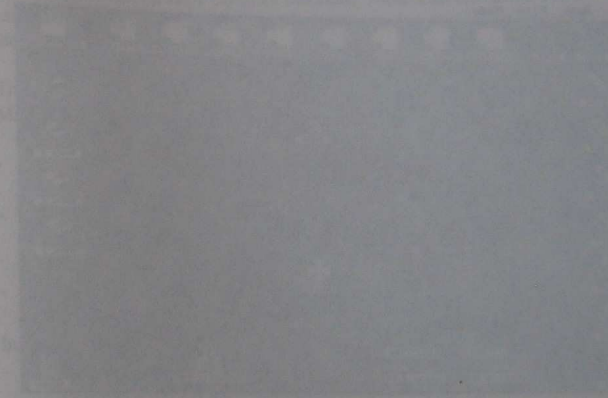
If possible, connect the sampler, D/A Board and amplifier to the same electricity circuit to help prevent interference. Only use special-purpose, screened audio or MIDI cables.

Storage media

Treat disks and the cartridges for removable disk systems with the same care that you would treat audio or video cassettes. They must not be placed on monitors, loudspeakers or other equipment that produce powerful magnetic fields. When not in place inside a disk drive, disks and cartridges should always be protected from dust, damp and heat.

In addition to this, it is highly advisable to carefully label disks and cartridges, so that you know what data they hold.

Reference



Functions list

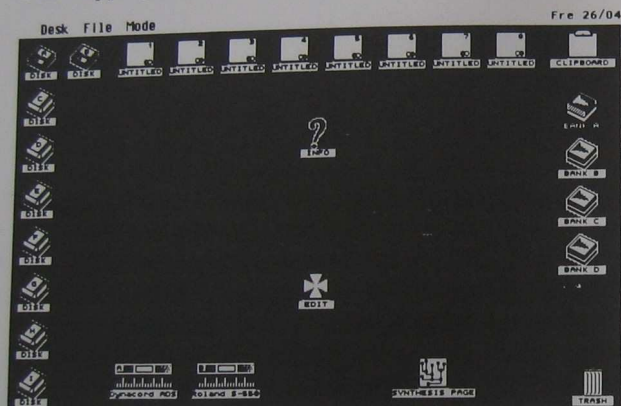
Pages

Startup screen

The program takes about 100 seconds to load from a floppy disk and around 20 seconds from a hard disk.

As soon as you see movement on the title screen you know that the program is loaded and you can start working with AVALON by moving the mouse.

The Mapping Page



The Mapping Page is the first thing you will see when you enter the AVALON system. This program level is the system's control centre and the starting point for all further work. All functions involving the transfer of sample data are carried out from this page.

- A transfer is a process in which sample data is copied or moved, but not altered. This includes saving and loading to and from floppy disk and hard disk as well as copying operations within memory or the transmission of data between a sampler and the computer via MIDI.

Of course, AVALON can do very much more than simply transfer, copy and erase samples, but before moving on to more 'exciting' activities you should become so used to performing these basic functions that you don't have to think about them. Then you will be much better prepared for working within the different editors, the data-modifying modules of the program.

The layout of the Mapping Page

The Mapping Page is a mouse-oriented system, like the GEM desktop, but it provides you with far more complex functions. Almost all operations can be carried out using the icons found here by either moving or clicking them with the mouse.

The following information is listed below for each of the icons found on the Mapping Page:

- what it represents,
- which functions it calls when clicked, double-clicked or clicked with the right mouse button, and
- which functions it calls when dragged to another icon.

In this manual the symbol '>' is used to indicate dragging, eg

[ICON] > [DESTINATION ICON] : [FUNCTION]

Mouse operations that have no function are not listed.

Sample Icon



Each sample icon represents one of the 32 memory locations available for sample data.

There are always 8 sample icons displayed on the screen. You can position and allocate them as you choose. These 8 samples form a bank.

You can switch between the total of 4 banks using the 4 bank icons A - D.

The sample icons are used for practically all the program's important functions.

An unused memory location is displayed as an empty white box; one which is being used contains a waveform symbol.

The following details can be saved with each sample:

- the sampling rate (in kHz)
- the start and end points of the sample itself as well as up to 8 loops (in sample words)
- loop status (on/off)
- reverse (on/off)

Dependent on the sampler used, this parameter allows you to activate loop playback either in reverse (end to beginning) or in an alternating mode (forward>reverse and reverse>forward).

- 8 markers
- its name
- a brief comment (5 lines of 30 characters)
- pitch (in standard MIDI format)

AVALON allocates the available memory dynamically; a sample location is thus able to contain only one individual sample value or 40 seconds of music.

Clicking a Sample Icon

This operation enables you to link two consecutive samples (1&2, 3&4, 5&6 and 7&8) together to form a stereo sample.

Stereo samples can be modified, copied, saved and transmitted to a sampler together.

Sample icons that are linked in this way are displayed in inverse video (ie black).

To split a stereo sample back into two mono samples simply click on the sample icon again.

The same result can be achieved using the Mode Menu options 'Sample 1&2: Mono/Stereo' to 'Sample 7&8: Mono/Stereo'.

The small symbol in the lower righthand corner of a sample icon indicates whether it is the left or right element in a stereo pair.

Double-clicking a Sample Icon

Call the Time Domain Editor.

The selected sample is transferred to the TDE and is displayed there in a window of maximum possible size.

Reference

Clicking a Sample Icon with the right mouse button

Playback sample.

The sample can be played back via

- the computer's internal monitor,
- the 16-bit D/A Board, or
- the sound chip of the STE.

The desired playback medium can be selected via the Preferences option in the File Menu.

Please note that you can also playback a specific sample by pressing the appropriate function key, ie pressing function key [F5] plays back sample 5, and so on.

Stereo samples are played back in stereo.

- During playback all other computer functions are blocked and signals from the mouse are ignored. There is thus no need to panic if the pointer disappears and there is no response to keyboard entries while a long sample is being played; this is perfectly normal.

The MIDI processor SMP 24 carries out a reset during sample playback. There are technical reasons for this and unfortunately it cannot be avoided.

Sample > Edit

Call the Time Domain Editor.

The selected sample is transferred to the TDE and displayed there in a window of maximum possible size.

Reference

Sample > Disk

Save the sample to the relevant disk drive.

A File Selector appears. The sample's name is entered automatically, but naturally you can give it another name if you wish.

If the sample is part of a stereo sample, the File Selector appears again to allow you to save the second sample.

We would advise you to save the individual parts of a stereo sound under names which include a 'channel code' as their last letter (eg 'SNARE_L' and 'SNARE_R').

When working with longer sounds, which often take up considerable amounts of memory, the speed at which data can be loaded or saved becomes very noticeable. It would seem that due to the apparently interminable waiting times involved in using a floppy disk drive many computer users who do not have a hard disk often do not take the trouble to regularly save their work. They simply make do by copying samples within memory or sending them to a sampler via MIDI.

However, you should always bear in mind that computer and sampler memory is volatile, ie all the data stored in it is lost when the machine is switched off or a system crash or power cut occurs (things which actually do happen every now and then!).

For this reason regularly backup your data on a 'physical' disk drive, even if this takes some time. When it comes down to it, investing in a hard disk is not only worth it because of its speed and capacity, but also because of the security this medium provides.

Sample > Sampler

Transfer a sample to a sampler.

The Sample Transfer Dialog Box appears - the parameters and settings for specific samplers are described in the "Appendix".

Click on the OK button to begin the transfer.

Reference

Whilst the sample is being transferred a bar appears on the screen which indicates how long this procedure will take.

If Stereo Mode is active then two consecutive samples (1/2, 3/4, 5/6, 7/8) are sent to the sampler. If the equipment allows, AVALON can even create pseudo-stereo sounds on mono samplers.

Sample > Sample

Copy a sample to another sample memory location.

If another sample is already located there, it will be overwritten, ie erased.

If you drag a stereo sample (ie a sample icon that forms part of a stereo sample) to another stereo location, both sample components (eg 'BD_LEFT' and 'BD_RIGHT') are copied.

If you drag one of the parts of a stereo sample to a 'normal' (mono) sample, then only that component will be copied (eg 'BD_LEFT').

Sample > Clipboard

Copy a sample to the clipboard.

If another sample is already located there, it will be erased.

If you want to copy a stereo sample to the clipboard, the clipboard must be in Stereo Mode.

Sample > Bank

Copy a sample to the first free sample memory location in a(nother) bank. An error message will appear if there are no sample locations free in the bank.

Reference

Sample > Info

Sample Information
CANCEL OK

Name : TRIANGLE

Length : 20290 Samples
= 468 ms

S.-Rate : 44100 Hz

Pitch : 62

Format : 12 Bit

Modified: ☐

PPP Standard Triangle

S1000 Sample (05/05/91)

by Pete

Use: 'Ratio', Track 1

START POINT END POINT

Sample : 0 20290

Marker #1 :	0	☐
Marker #2 :	0	☐
Marker #3 :	0	☐
Marker #4 :	0	☐
Marker #5 :	0	☐
Marker #6 :	0	☐
Marker #7 :	0	☐
Marker #8 :	0	☐

Call the Sample Info Dialog Box.

The Sample Info Dialog Box displays the following information on a sample:

- its name
- a short comment (5 lines of 30 characters)
- pitch (in standard MIDI format)
- the length of the sample (in sample words and milliseconds)
- the sampling rate (in kHz)
- the start and end points of the sample itself as well as up to 8 loops (in sample words)
- loop status (on/off)
Only the start and end points of active loops are displayed. This improves the clarity of the display.

Reference

- reverse (on/off)
- 8 markers
- a Modified flag that indicates whether the sample has been edited since it was loaded/saved.

When you save a bank (see 'Bank Icon'), the only samples saved are those whose Modified flags are active.

You can modify all the data in this dialog box with the exception of the sample's length.

Sample > Trash

Erase sample.

Once executed, this function cannot be undone.

Sample > Synthesis

No function.

Clipboard Icon



The clipboard is the 'thirty-third sample location' and - as its name suggests - it is used to temporarily store data which is to be processed later. You can for example, copy sample 3 in bank A to sample location 7 in bank C by using the clipboard as a temporary transit point.

Reference

The Cut Block and Copy Block functions in the Time Domain Editor also use the clipboard as a temporary store. If you wish to work with these functions, it is vital that you do not leave important samples on the clipboard before doing so as they would be erased.

Otherwise, the clipboard can be used, copied, erased and switched to Stereo Mode in exactly the same way as a 'normal' sample location.

The clipboard is not allocated to a particular bank; it is a fixed, additional memory location.

Clicking the Clipboard Icon

Switch the clipboard into Stereo Mode.

The operation has the same effect as clicking on a sample icon (see above).

Double-clicking the Clipboard Icon

Call the Clipboard Info Dialog Box.

The Clipboard Info Dialog Box displays the same information as the Sample Info Dialog Box.

You can modify all entries except for the sample's length.

Clicking the Clipboard Icon with the right mouse button

Playback the sample on the clipboard.

Playback is carried out via

- the computer's internal monitor,
- the 16-bit D/A Board, or
- the sound chip of the STE.

Reference

The desired playback medium can be selected via the Preferences option in the File Menu.

The clipboard sample can also be played back by pressing function keys [F9] or [F10].

- * During playback all other computer functions are blocked and signals from the mouse are ignored.

The MIDI processor SMP 24 carries out a reset during sample playback. There are technical reasons for this and unfortunately it cannot be avoided.

Clipboard > Edit

No function.

Clipboard > Disk

Save the clipboard sample to the relevant disk drive.

A File Selector appears. The name of the clipboard sample is entered automatically, but you can also enter another name if you wish.

If the clipboard contains a stereo sample, the File Selector appears again to allow you to save the second sample.

Clipboard > Sampler

Transfer the clipboard sample to a sampler.

The Sample Transfer Dialog Box appears - the parameters and settings for specific samplers are described in the "Appendix".

Click on the OK button to begin the transfer.

Reference

Whilst the sample is being transferred a bar appears on the screen which indicates how long this procedure will take.

Clipboard > Sample

Copy the clipboard sample to a sample memory location.

If another sample is already located there, it will be erased.

If you copy a stereo sample from the clipboard to another stereo sample location, both sample components are copied (eg 'BD_LEFT' and 'BD_RIGHT').

If you drag a stereo sample from the clipboard to a 'normal' (mono) sample location, only the first component is copied (eg 'BD_LEFT').

Clipboard > Bank

No function.

Clipboard > Info

Call the Clipboard Info Dialog Box.

The Clipboard Info Dialog Box displays the same information as the Sample Info Dialog Box.

You can modify all the data displayed in this dialog box except for the sample's length.

Clipboard > Trash

Erase clipboard sample.

Once executed, this function cannot be undone.

Reference

Clipboard > Synthesis

No function.

Bank Icon



The bank icons represent the four banks A - D, which each consist of 8 sample memory locations. In addition to their main function, ie switching between the four banks, these icons also allow you to load, copy and erase complete banks of samples. This is particularly useful if you do not only work with individual samples, but also with larger sets (eg 8 different drum sounds) or multisamples.

Clicking a Bank Icon

Switch to this bank.

In order to change to another bank simply click it on briefly with the mouse.

The active bank is displayed in inverse form (black). When you first load the program this will always be bank A.

Double-clicking a Bank Icon

Call the Bank Info Dialog Box.

This dialog box displays the following information:

- Name
The name of the relevant sample.

Reference

- Modified
If active, the relevant sample has been modified since it was loaded.
- Size
The length of the sample.
- Total Size
The size of the bank.

Bank > Edit

No function.

Bank > Disk

Save bank.

You do not need to enter a name before saving; the samples in the bank are saved under the names displayed at the individual sample icons. Clearly, unoccupied sample locations ('UNTITLED') are not saved.

Bank > Sampler

No function.

Bank > Sample

No function.

Bank > Bank

Copy a bank to another bank.

If there are any samples in the destination bank, they are overwritten, ie erased.

Reference

Bank > Info

Call the Bank Info Dialog Box.

This dialog box displays the following information:

- **Name**
The name of the relevant sample.
- **Modified**
If active, the relevant sample has been modified since it was loaded.
- **Size**
The length of the sample.
- **Total Size**
The size of the bank.

Bank > Trash

Erase bank.

All the samples in this bank are erased.

Once executed, this function cannot be undone.

Bank > Synthesis

No function.

Reference

Sampler Icon

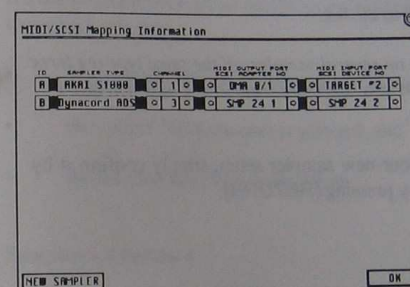


The sampler icons represent the samplers and sample players connected to the computer. You are able to operate up to 10 such machines at the same time.

Clicking a Sampler Icon

No function.

Double-clicking a Sampler Icon



Call the Sampler Setup Dialog Box.

This dialog box indicates the following on each registered sampler:

- a code letter ('A' to 'J'),
- the name of the sampler, and
- the MIDI channel and MIDI port to which it is connected or (when you are using the Steinberg SCSI controller)

Reference

- the SCSI adapter No./ Device No.
For more information about transferring samples via SCSI, please refer to the SCSI controller's manual.

The last two parameters can be changed directly inside this dialog box.

In addition to the computer's MIDI IN and OUT port, you can also use the ports of a MIDI expander. The expander must be correctly connected to your computer, switched on and installed - using the appropriate driver in the program's 'MROS' folder.

If you now want to log on a new sampler, click on the New Sampler button in the lower left-hand corner of the dialog box. An additional dialog box then appears which lists all the samplers which can be installed within the AVALON system. You simply select the desired sampler with a click of the mouse and it is added to those already selected.

If you have already selected ten samplers, you must first delete one of them before a new one can be added to the list.

Note: It is also possible to log on several samplers of the same type (eg three AKAI S1000s) as long as you configure them to use different MIDI channels or MIDI ports.

When you are satisfied with your new sampler setup, simply confirm it by clicking on the Exit button or by pressing [RETURN].

Sampler > Edit

No function.

Sampler > Disk

No function.

Reference

Sampler > Sampler

No function.

Sampler > Sample

Transfer a sample from a sampler and store it at the relevant sample location.

The Sample Transfer Dialog Box appears - the parameters and settings for specific samplers are described in the "Appendix".

Click on the OK button to begin the transfer.

Whilst the sample is being transferred a bar appears on the screen which indicates how long this procedure will take.

If a transmission error occurs, the program will inform you of this. You should then check whether

- the sampler is properly connected to the computer or MIDI expander,
- the correct MIDI channel is selected, and
- the selected sample actually exists.

Sampler > Clipboard

Transfer a sample from the sampler and store it on the clipboard.

This function is similar to Sampler > Sample (see above).

Sampler > Bank

No function.

Reference

Sampler > Info

Call the Sampler Setup Dialog Box.

This dialog box is described above in the section "Double-clicking a Sampler Icon".

Sampler > Trash

Remove a sampler from the sampler setup.

Sampler > Synthesis

No function.

Disk Icon



The disk icons represent the drives which are available for saving and loading sound data.

Each time you start work with AVALON the program automatically determines how many disk drives your computer system has and displays an appropriate icon for each one of them.

Drives A: and B: are always floppy disk drives.

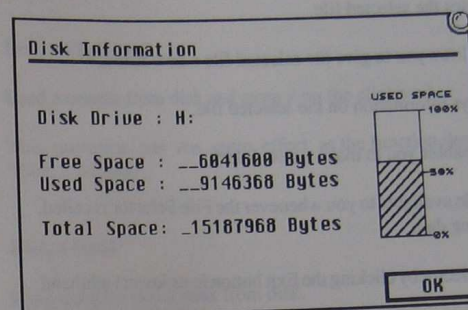
Drives C:, D:, E:, F:.. to P: are - dependent on what you installed before starting the program - either RAM disks or hard disks.

Reference

Clicking a Disk Icon

No function.

Double-clicking a Disk Icon



Call the Disk Info Dialog Box.

The following information is displayed:

- Total Space
Storage capacity of this disk.
(Values: 358008 bytes for a single-sided disk,
726016 bytes for a double-sided disk.)
- Used Space
Disk space already occupied.
- Free Space
Remaining disk space available.

Reference

Disk > Edit

Disk Edit functions.

The AVALON File Selector appears. It offers additional functions to those provided by the standard GEM File Selector.

- Erase File deletes the selected file.
- Rename File allows you to give the selected file a new name.
- File Info displays information on the selected file.
- Create Folder enables you to make a new folder.

These functions are made available to you whenever the File Selector is called, ie when loading or saving data.

You can leave the File Selector by clicking the Exit button in its lower righthand corner.

Disk > Disk

No function.

A dialog box informs you that disk copies cannot be made from within the program.

Disk > Sampler

No function.

Reference

Disk > Sample

Load a sample from disk and store it at the relevant sample memory location.

The File Selector appears (the directory of the selected disk drive) and you can select and load the desired sample.>

Disk > Clipboard

Load a sample from disk and store it on the clipboard.

This operation has the same effect as the function described above under "Disk > Sample".

Disk > Bank

Load samples into a bank from disk.

The File Selector appears for the relevant disk drive - as when loading individual samples.

However, in this case you can only determine from which drive and/or from which folder you want to load samples. As soon as you confirm with OK the first 8 samples in this drive or folder will be loaded - irrespective of whether and which sample names were previously entered.

Disk Scan Load in the Mode Menu has no effect when loading banks.

Reference

Disk > Info

Call the Disk Info Dialog Box.

The following information is displayed:

- Total Space
Storage capacity of this disk.
(Values: 358008 bytes for a single-sided disk,
726016 bytes for a double-sided disk.)
- Used Space
Disk space already occupied.
- Free Space
Remaining disk space available.

Disk > Trash

Format disk.

This function is only available for floppy disks, not for hard disks or removable disk systems. This means that you can only drag the Disk A and Disk B icons to the Trash Icon.

You are able to choose between single-sided or double-sided disk formatting.

A dialog box will warn you if the disk is already formatted.

Disk > Synthesis

No function.

Reference

Edit Icon



The Edit icon gives you access to the other parts of the program in which - in contrast to the Mapping Page - data can be modified.

The exact function of the Edit icon depends on the type of icon which you move onto it.

Clicking the Edit Icon

No function.

Double-clicking the Edit Icon

Call the Time Domain Editor.

If you have not yet accessed the Time Domain Editor, the window for sample location 1 in the default bank is automatically opened, irrespective of whether this memory location contains sample data or not.

If you have already worked in the Time Domain Editor and call it again by double-clicking the Edit Icon, it reappears in its previous state. This means that all windows are the same size and in the same position as they were when you left the editor.

Edit > Disk

No function.

Reference

Edit > Sampler

No function.

Edit > Sample

No function.

Edit > Clipboard

No function.

Edit > Bank

No function.

Edit > Info

Call the Program Info Dialog Box.

This dialog box displays the following information:

- Version No.
The AVALON version number. The digit before the decimal point indicates the program version, the digit after the decimal point the number of the update.
- Release Date
The date this version was released.
- The program authors
- A copyright notice.

Reference

Edit > Trash

No function.

Info Icon



Like Edit and Trash, Info is also an icon whose function depends on which icon is dragged to it. It can be used to call information on sample data, samplers and disks. You cannot drag the Info icon to another icon.

Clicking the Info Icon

No function.

Double-clicking the Info Icon

Call the Sampler Setup Dialog Box.

This dialog box is described in the section "Double-clicking a Sampler Icon".

Trash Icon



The Trash Icon is used to erase data.

The Trash Icon itself cannot be dragged to another icon. In order to erase a set of data, the appropriate icon has to be dragged to the trash can.

Reference

Synthesis Icon



This Mapping Page icon represents the Synthesis Page, the program module that gives you access to all important sound synthesis techniques.

Clicking the Synthesis Icon

No function.

Double-clicking the Synthesis Icon

Call the Synthesis Page.

Sample memory location 1 in the current bank is reserved as the default location for a synthesized sound. When you want to keep a sound you have created on the Synthesis Page, you have to transfer it using the Copy To MP function. When you do this, the synthesized sound is copied to memory location 1 of the active bank. If this location already contained sample data, it is erased.

Synthesis > Edit

No function.

Synthesis > Disk

No function.

Reference

Synthesis > Sampler

No function.

Synthesis > Sample

Call the Synthesis Page.

The sample memory location to which you drag the Synthesis Icon is reserved as the default location for a synthesized sound. This means that when you want to keep a sound created on the Synthesis Page and transfer it using the Copy to MP function, it is copied to this memory location. If this location already contains sample data, it will be erased.

Synthesis > Bank

No function.

Synthesis > Info

A dialog box appears that informs you whether the SP module is already in memory ('Synthesis Page is resident') or still has to be loaded ('...not resident').

Synthesis > Trash

Remove Synthesis Page from memory.

The SP module is removed from memory. This means that the memory it occupied can again be used to store sample data.

Reference

Steinberg Logo

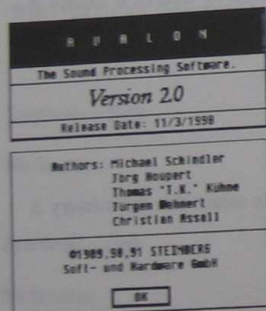


In addition to the 'real' icons listed above, the STEINBERG logo also appears on the Mapping Page. Within the program it has no further function and you can switch it on or off via the Show/Hide STEINBERG option in the Mode Menu

Mapping Page menus and dialog boxes

Desk Menu

About



Call the Program Info Dialog Box.

Reference

This dialog box displays the following information:

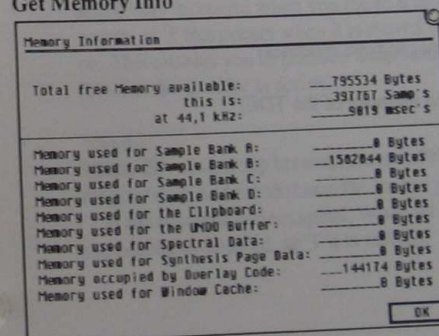
- Version No.
The AVALON version number. The digit before the decimal point indicates the program version, the digit after the decimal point the number of the update.
- Release Date
The date this version was released.
- The program authors
- A copyright notice.

It is also possible to run accessories (eg the control accessory) from the Desk Menu - provided that you loaded AVALON from a drive that contained accessories.

Accessories can only be used on the Mapping Page and should be closed before you enter higher program levels as they could otherwise cause problems by clashing with the program.

File Menu

Get Memory Info



This function gives you an immediate overview of the current state of memory.

Reference

The following data is displayed

- Total Free Memory
Details of available memory
 - in bytes
 - in sample values
(Internally AVALON uses 16 bits for each sample value; a sample value thus consists of 2 bytes)
 - length of the sample
(at an assumed sampling rate of 44.1 kHz)

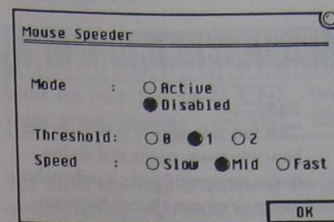
The memory occupied by

- Banks A-D
- Clipboard
- Undo buffer
(memory buffer containing the last modification in the TDE)
- Spectral data (of the sample in the FDE)
- Synthesis Page data
- Overlay Code (external program modules)
- Window Cache (screen memory of the TDE)

These details are given in bytes.

Reference

Mouse Speeder

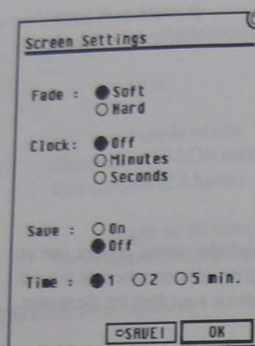


The Mouse Speeder increases the speed at which the mouse pointer can be moved across the screen. This is extremely useful if you want to be able to work very quickly and/or you have little room available on your desk for the mouse.

You can adjust the following parameters:

- Active/Disabled
Switches the Mouse Speeder on and off.
- Threshold
This is the 'threshold value' that determines when the speed should be increased.
If it is set at '0', the pointer always moves at the same speed. At '1' the speed only increases when you move it more than one point on the screen, at '2' this occurs when it is moved more than two points, and so on. This enables you to position the pointer very precisely despite the fact that the speeder is active.
- Speed
This is the relationship between the speed you move the 'physical' mouse on your desk and the distance the mouse pointer moves on the screen. '1' is the standard system speed. When set at '2' the distance covered by the pointer is doubled, at '3' it is trebled.

Screen Settings



This menu option allows you to adjust various basic screen settings.

- **Fade**
This function allows you to decide how windows, pages and dialog boxes are opened and closed on the screen. You can choose between the normal Hard setting - windows are immediately displayed - or Soft - windows are gently faded in/out.
- **Clock**
This allows you to have the time displayed on the screen. The display can either be in hours/minutes/seconds (Sec) or just hours/minutes (Min).

If you have a MEGA ST or an appropriately modified machine, your computer has a battery-run system clock. If you have an older ST with no clock - and feel that this function is useful - you can enter the exact time every time you turn on your computer using the accessory provided on the ATARI system disk.

- **Save**
This function allows you to activate a screen saver that turns off screen output after 1, 2 or 5 minutes. This reduces the wear on your monitor. The picture returns immediately when you click on one of the mouse buttons.

MIDI Keyboard



The MIDI keyboard is extremely useful when you want to play a sampler without having to go to the 'real' keyboard or change your MIDI cables.

When this function is called a 9-octave keyboard appears on the screen that you can even play at different velocities using the mouse. The higher you hit a key, the greater the velocity.

You can adjust

- **MIDI Out Port**
The MIDI port to which the notes should be output.
- **MIDI Output Channel**
The MIDI channel for the notes.

This dialog box also allows you to define a Music Input. All MIDI commands - except SysEx data such as samples and synthesizer sounds - that are received via this port are then redirected to the MIDI OUT port without being processed. This makes it possible to play a connected sampler from a master keyboard without having to exit AVALON.

If you also activate the Keep Input Channel button, the Channel Messages received (ie notes, controllers, etc) are not redirected to the MIDI output channel but retain their original channel designation. If you have several samplers connected to one MIDI port, this enables you to communicate with the different machines from your master keyboard by changing the MIDI channel.

Reference

Preferences

Preferences

GLOBALS

Handling: ☐ Normal ☒ Expert

☒ Screen Messages

☒ Show Extra Help

Value Edit: ☒ Slider ☐ Keypad

TIME DOMAIN EDITOR

☐ Auto Scrolling

☒ Auto + Correction

☒ Backup Last Edit

☒ Vertical Lines

☒ Link Samples

☐ Fill Spaces

☒ Play whole Sample

☐ Play marked Block

☐ Play zoomed Part

☒ Draw all Samples

☐ Default Overview

USER INTERFACE

Alertboxes: ☐ are centered ☒ pop up at Mouse Position

Dialogs: ☐ are centered ☒ pop up at Mouse Position

FREQUENCY DOMAIN EDITOR

Analysis/Resynthesis: ☒ Standard ☐ Professional

OK

The Preferences Dialog Box contains a variety of basic settings for AVALON and individual program modules.

- **Global**
These are settings that affect several parts of AVALON or the program as a whole.

- **Handling**
This option allows you to choose between Normal and Expert Modes.

In Normal Mode you are always asked for confirmation before an operation is carried out which results in the modification or erasing of data. You thus have an opportunity to reconsider all decisions and are better able to avoid making serious errors.

In Expert Mode functions which modify or erase data are immediately carried out as soon as you have moved or clicked the appropriate icon.

Reference

Although you will certainly soon learn to appreciate the high speed at which you can work in Expert Mode, you should also note that in stress situations or when you are in a 'creative high' it is easy to make mistakes which can no longer be corrected.

Thus if you decide to work without receiving confirmation and warning messages, you should at least regularly backup the samples you are working on (preferably on a 'physical' disk drive, ie floppy disk or hard disk, which also counteracts the danger of a power cut) and activate Backup Last Edit for the TDE (see below).

- **Screen Messages**

If Screen Messages is active, AVALON displays messages ('Analyzing Block', 'Fading Sample') to let you know what is going on when the program is 'busy' carrying out calculations.

Depending on the length of the sample involved, some of the complex editor functions can require considerable processing times. Thus, you will probably appreciate status messages that tell you what AVALON is doing and how long it will take, especially when you first start working with the program.

For most functions a bar and minutes/seconds display appear on the screen indicating the time still required before the necessary calculations are completed.

- **Extra Help**

AVALON is an extremely complex program with many different levels; strictly speaking it is a package of programs consisting of numerous interlocking and interdependent subprograms.

You can thus soon find yourself lost in this 'labyrinth'. In such a situation, referring to a manual is often very difficult as you don't know what to look up; specific objects on the screen or functions may be completely unclear to you.

This is the point of AVALON's Extra Help Mode. When active, the function of the icon at the mouse pointer is indicated at the top righthand corner of the screen.

Reference

Using this manual and Extra Help Mode together will enable you to quickly find your way around AVALON.

• Value Edit

This option allows you to choose between the two different means of entering figures - Value Slider (a simulated slide control) and Numeric Keypad (a small number pad). They are described in the section "Communicating with your computer".

When you have selected one of these techniques you don't have to completely abandon the use of the other. You can activate it by clicking on the value you want to alter whilst holding down the [SHIFT] key.

• Time Domain Editor

This is where you will find important basic settings for the TDE.

• Auto Scrolling

If you activate Auto Scrolling, the visible screen area is automatically moved forward as soon as the pointer is moved into the right third of the active sample window. You are thus able to 'draw' or modify longer waveforms without having to stop at the edge of the screen.

As this form of scrolling involves a great deal of processing, to avoid dramatic jumps within the new waveform you should not move the mouse up or down too fast.

Auto Scrolling can only be used to move the visible screen area to the left; this function will not work in the other direction.>

• Auto +- Correction

When adding two separate sound elements with ADD BLOCK it can occur that the two signal levels together exceed the maximum level allowed. This results in distortion. If Auto+- Correction has been switched on, however, this can be avoided as both elements are reduced to half their original volume before mixing takes place.

Reference

• Backup Last Edit

If this mode is active, you can cancel the last editing function you carried out using Undo Last Edit in the Edit Menu. Activating this mode causes AVALON to reserve a buffer in memory where the last version of the current sample is stored. This buffer is emptied when you select another sample or leave the editor.

Please note that the buffer is always exactly as long as the part of the sample which was last modified - this means that if you filter a 4-second sample, the buffer required will also be 4 seconds long. The memory available can thus be dramatically reduced when editing longer samples. AVALON will draw your attention to the fact that there is no more memory available if this occurs. In such an event you can either make do without the Undo option or remove those accessories, samples or block buffers from memory which are not absolutely necessary.

• Vertical Lines

If this mode is active (at a resolution of more than 1:1) each sample value is represented on the screen as a vertical line. The distance between these lines enables you to see how far you have 'zoomed in'. If, for example, you can only see 51 sample values on the screen (instead of 512 in 1:1 resolution) you have thus reached a zoom level of 10:1.

• Link Samples

When this mode is active, all sample values are joined together by a line - at resolutions higher than 1:1. This enables you to get a very exact impression of the shape of a waveform even at extremely high resolutions.

• Fill Spaces

When this mode is active, the spaces between the sample values are filled in in black - at resolutions higher than 1:1. However, this takes longer than the normal display mode using vertical lines.

• Play Whole Sample

In this mode the whole sample (ie the area between the sample's start and end point) is always played back when you click on the Play Icon.

The Loop Mode option in the Edit Menu enables you to determine which loop is to be sustained when you keep the mouse button pressed.

Reference

If you want to play back the areas in front of a sample's start point and behind its end point, click on the Play Icon with the right mouse button.

- Play Marked Block

In this mode clicking on the Play Icon only plays back the marked block.

- Play Zoomed Part

If this mode is active, clicking on the Play Icon only plays back the section of the sample currently visible in the sample window.

- Draw All Samples

In Draw All Samples Mode a low resolution display (eg the Full Wave Display) will have the same characteristics as a high resolution display (ie 1:1 or higher).

At zoom factors greater than 1:1, several sample values are represented by one point on the screen (eg 12 values at a factor of 1:12). If only one of these values (eg the first) is taken into consideration when the display is calculated, this results in a distorted, 'jagged' representation of the sample.

However, by activating Draw All Samples you can ensure that all the sample values represented by one point on the screen will be checked and the highest sample value used to produce the display.

This mode can be deactivated as it increases the time required to calculate the screen image.

- Default Overview

If this function is active, Overview Display is automatically switched on when you call the TDE.

- User Interface

This option allows you to adjust the settings that affect communication with AVALON.

- Alert Boxes

This option allows you to choose between two different positioning modes for alert dialog boxes.

Reference

- Centered

As in most GEM programs, dialog boxes always appear in the middle of the screen.

- Pop Up
At Mouse
Position

Dialog boxes always appear at the current pointer position. This is particularly useful if you are using a large monitor, as it saves you the trouble of having to locate the dialog box.

- Dialogs

This option allows you to choose between two different positioning modes for decision and settings dialog boxes.

As under Alert Boxes (see above), the two modes are

- Centered

- Pop Up At Mouse Position

- Sample Playback

This option allows you to determine how samples, sample fragments, spectrums and synthesized sounds are to be played back.

- Internal Speaker

Playback occurs via the built-in loudspeaker of the SM 124 monitor. Sound quality is rather low due to the limited capabilities of the sound chip and monitor loudspeaker, which were not conceived for such applications.

- 16-bit D/A Board

Playback occurs via a 16-bit D/A Board.

The installation of the D/A Board is described in the "Hardware" section. Additional information can be found in the D/A Board's instruction booklet.

Only select this setting if you have a D/A Board connected to your computer.

- ATARI DMA Sound

Playback occurs via the DMA chip of the ATARI STE.

Only select this setting if you are using an STE or MEGA STE computer.

Reference

- Professional Analysis/Resynthesis
When you activate this mode, Fourier analysis and resynthesis are carried out with higher precision. This leads to improved sound quality. This mode compensates for the errors that occur when sounds are processed in the frequency domain.

Analysis and resynthesis in Professional Mode takes somewhat longer than normal.

M.Only Bank Save

If this mode is active, only samples that have been modified will be saved when you save a bank.

To save a bank, move the appropriate bank icon to the icon representing the drive where the bank is to be saved. You do not need to enter file names before saving; the samples in the bank are saved under the names displayed at the individual sample icons. Clearly, unoccupied sample locations ('UNTITLED') are not saved.

If you save a bank to the drive or folder from which they were previously loaded, the original samples are always overwritten. Normally, all 8 samples are saved. However, if you have activated M.Only Bank Save, only those samples which were modified in some way will be saved back to disk. This can save a lot of time.

Example: You loaded the samples PIANO1 to PIANO8 from the folder PIANO on the disk in drive B:, but only modified the samples PIANO5 and PIANO7 in the TDE (eg changed the loop points). You now want to save the new PIANO multisample back to disk. You thus move the PIANO bank icon to the Disk B icon, open the PIANO folder and confirm with OK or [RETURN]. If M.Only Bank Save is active, only samples PIANO5 and PIANO7 are saved - these were the only samples modified. Don't forget that if this function is active it also effects the way banks are saved when they are saved to another location. Following the above example, if you wished to save the modified PIANO bank loaded from drive B: to drive E:, the program would still only save PIANO5 and PIANO7 there, as they were the only samples modified.

Reference

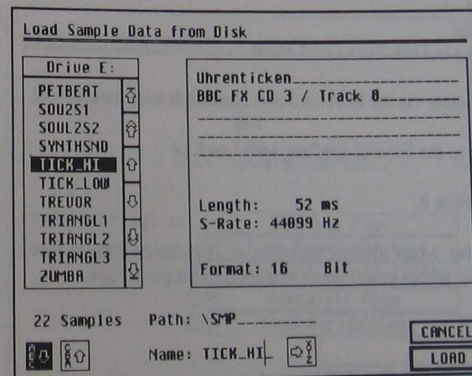
Disk Scan Load

Activate Disk Scan Mode.

Normally when the File Selector appears you will only see the samples which are located at the 'root' of the disk or hard disk or in the default folder. In order to get a complete overview you would then have to open the first folder, have a look, close it again and open the next one, and so on...

Disk Scan is a function which scans all the folders on a disk every time it is accessed and then presents you with a complete alphabetical list of all the samples found in that drive. You no longer need to worry about where you originally saved a sample, but can load it immediately.

As soon as you have told the computer that you want to load a sample from disk (by dragging a disk icon to a sample icon) Disk Scan Load scans the appropriate disk and lists all the samples found in alphabetical order. The two ABC buttons allow you to determine whether the samples should be listed in ascending or descending order.



The ABC field in the middle enables you to select specific samples from a longer list. To do this, simply enter the first letter or letters of the sample's name in the text input field beside the list of samples and click on the ABC field in the middle. The list will scroll so that the samples whose names begin with the relevant letter(s) then appear.

Reference

If no letters are entered the display will jump to the middle of the list.

You can also obtain additional information about a sample by selecting it in the list (click on the name once).

- The sampling rate,
- The length of the sample in ms and
- The comment (if you saved one with the sample)

will then be displayed in the middle of the dialog box.

Particularly when using a hard disk, Disk Scan Load and the possibility to save a brief comment with each sample become extremely powerful tools. You can get a general overview within seconds and do not need to load ten different crash sounds to find out which one it was that you wanted. It is a good idea to maintain a simple system to keep your sample library organized (for example, 1st line: sampler used, when recorded; 2nd line: sound source; 3rd line: notes on the sound's characteristics...).

To load a sample whilst in Disk Scan Load Mode

- either enter its name via the keyboard and confirm it with Load,
- select the name in the list and confirm with Load, or
- simply double-click it.

This mode is also active when loading samples in Digidesign format (see "Import/Export" below); however not when you load a complete bank.

Reference

Import/Export

If Import/Export Mode is active, you can load Digidesign sample files or save AVALON samples in Digidesign format.

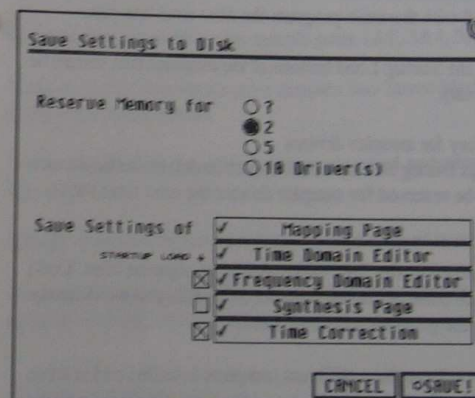
You should activate this mode if you want to import samples created using Digidesign, Turbosynth or other American programs of this type.

Samples in the Digidesign format are given the file extension '.SD' (instead of '.SMP').

While Import/Export is active you cannot read or write files in the AVALON '.SMP' format.

If you have activated Disk Scan Load, it also remains active in Import/Export Mode.

Save Settings



The Save Settings option allows you to save all AVALON's basic settings.

As these involve a total of well over 150 different parameters we will not bother you by listing them here.

Reference

If you want the basic settings to be used the next time the program is run, 'AVALON.SET' (the settings file) has to be saved in the 'AVALON' folder where the 'AVALON.IMG' and 'AVALON.PAC' files are located. It is, however, also possible to save settings under different names (for different purposes) and rename the configuration you require as 'AVALON.SET' before running the program.

Before the settings are saved, a dialog box will appear in which you can carry out various adjustments. When you have finished, click on the Save button. A File Selector will then appear and you can enter a name for the new settings file.

- Program module settings and Startup Load status

This option allows you to decide

- whether the settings of each program module should be saved, and whether the
- Frequency Domain Editor,
- Synthesis Page and/or
- Time Correction module

should be loaded with the main program the next time it is run.

Simply click on the Startup Load buttons of the modules that should be loaded into memory.

- Reserving memory for sampler drivers

The Save Settings Dialog Box also enables you to determine how much memory should be reserved for sampler drivers the next time the program is run.

The drivers are stored in the 'DRIVERS' folder as separate files. Don't erase this folder, or erase or rename sampler drivers, as this would make it impossible for the program to reload them.>

For example, if you have three different samplers installed and reserve enough memory for five sampler drivers in the Save Settings Dialog Box, the next time you run the program the driver files for the three s samplers will be loaded into memory and there will still be enough space available for two more.

Reference

If you select the '?' setting, memory will be reserved for the current number of drivers, ie if there are four drivers installed when saving takes place, room is reserved for four drivers when the program is next run. Of course, you can then always remove the installed drivers from memory and replace them by others, but their total number remains limited to four.

Quit AVALON

This function allows you to leave AVALON and return to the desktop.

AVALON will warn you if there are any modified samples still in memory that have not been saved.

Mode Menu

1&2 Mono/Stereo - 7&8 Mono/Stereo

(De)activate Stereo Mode for sample pairs 1&2, 3&4, 5&6 and 7&8.

You can switch two consecutive samples into Stereo Mode by clicking on the appropriate menu option.

Stereo samples can be modified, copied, saved and transferred to a sampler together.

Sample icons that are linked in this way are displayed in inverse video (ie black).

To split a stereo sample back into two mono samples, simply click on the appropriate menu option again.

Clip Mono/Stereo

Switch the clipboard to Stereo Mode.

Invisible Grid

It is almost impossible to position objects exactly in line with one another when you move them on the screen, eg two disk icons that you want to reposition. However, if the Invisible Grid option is activated, a hidden grid automatically lines up the icons.

This makes 'tidying up' the Mapping Page very much easier.

Lock Positions

When Lock Positions is active, you cannot reposition the objects on the Mapping Page - their positions are fixed.

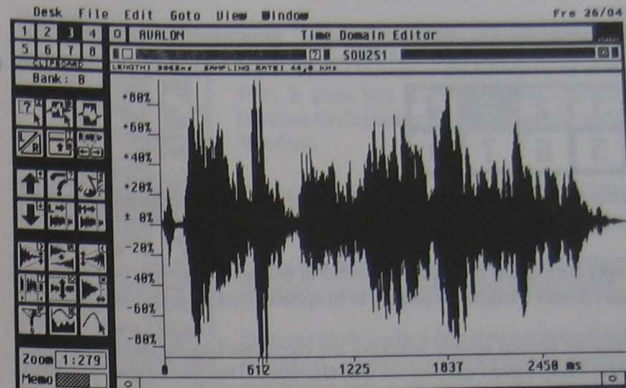
You have to switch off Lock Positions first before you can reposition the icons on the Mapping Page.

We recommend that you first rearrange the Mapping Page to suit the way you work - with Lock Positions turned off - and that you then reactivate it. You will thus avoid 'messing up' the screen with a careless movement of the mouse.

Show/Hide Steinberg

In addition to the 'real' icons listed, above the STEINBERG logo also appears on the Mapping Page. Within the program it has no further function and you can switch it on or off via the Show/Hide STEINBERG option in the Mode Menu.

The Time Domain Editor



In the Time Domain Editor (TDE) you are able to edit samples, which are represented on a two-dimensional basis (horizontal axis: time, vertical axis: amplitude, ie volume).

Sections of a sample (blocks), individual samples and stereo samples can be displayed, edited and copied in a great variety of ways. All modifications are carried out directly on the sample being edited and are thus transferred back to Level 1 (the Mapping Page).

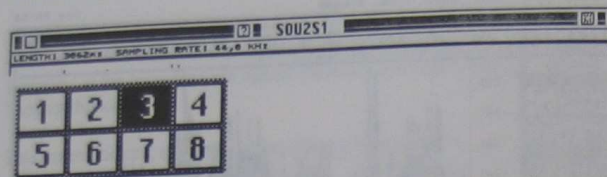
The layout of the Time Domain Editor

If the Mapping Page is your desktop, then the TDE corresponds to the tools in your top drawer: it contains a whole arsenal of digital scissors, erasers, drawing implements and templates which allow you to process and edit your sample data. All the functions which you will continually need have been put together in a 'toolbox', which is located on the lefthand side of the screen.

Samples are displayed in windows. There are various modes and functions that affect their representation.

Reference

Window boxes 1-8 & the clipboard



Above the toolbox you will find nine numbered boxes that are used to access the appropriate sample window.

Simply click on the box with the number of the sample that is to be displayed. If the relevant window has already been opened, clicking it on will activate it.

Previously opened sample windows are represented by boxes displayed in inverse video. Stereo samples are also identified. Thus, if you have activated sample 3 and then activate Stereo Mode, both boxes 3 and 4 will be displayed in inverse video.

Clicking on a window box with the right mouse button will playback the relevant sample.

Reference

Toolbox

General



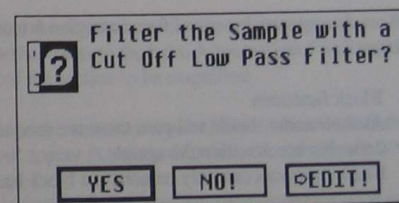
The toolbox is located on the lefthand side of the TDE. It gives you access to all the important functions for displaying and editing samples within the editor.

The toolbox offers you more than thirty different functions for precision editing of sample data.

All toolbox functions are activated by clicking on the appropriate icon with the left mouse button.

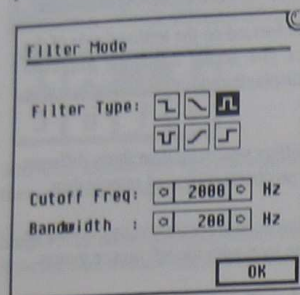
In Expert Mode toolbox functions are immediately carried out; in Normal Mode a dialog box appears that enables you to reconsider your decision and change the settings affecting the relevant function.

For example, if you want to filter a sample, you click on the Filter Icon in the lower lefthand corner of the toolbox. In Expert Mode the sample is filtered immediately. In Normal Mode, however, the following warning message appears:



Reference

If you click 'YES', the sample is filtered. If you click 'NO', nothing happens. If you click '>EDIT', the following window appears:



and you can adjust the filter settings.

Then, as soon as you exit the dialog box by clicking the OK button, the warning message appears again and you can either filter the sample or abort the process with 'NO'.

You can also call the appropriate settings dialog box for a function icon (ie Fade, Filter, Optimize, Resample, Erase, Cut, Delay, Loop or Replicate) by clicking it on with the right mouse button.

There are two special groups of functions, that are only active under certain conditions.

- Block functions
As their name should tell you, these are used to manipulate a freely definable block within the sample.
Block functions can only be used if a block has been defined.
- Stereo functions
These can only be used if
 - a) two consecutive samples (1&2, 3&4, 5&6, 7&8) are available for stereo processing, and
 - b) Stereo Mode is active.

Reference

As AVALON can only undo the last modification you make, we would strongly advise you to make a copy of a sample in memory before you begin editing it. This will help you to avoid unnecessary waiting times should you make a serious mistake and then have to reaccess the original sample.

Zoom Mode



Zoom Mode enables you to enlarge the display of an individual segment of a sample and reduce it again ('zoom out').

When you want to cut and loop samples, you will generally need to enlarge the display of a specific sample segment. This is the purpose of Zoom Mode.

When you open a sample window, you will initially see an overview of the whole sample. This is known as Full Wave Display.

To enlarge a section of a sample, click on the Zoom Mode Icon and reposition the mouse pointer in the sample display. The pointer will then change its form.

Move the pointer to the start of the segment you wish to enlarge, press down the left mouse button and move the mouse to the right. A 'rubber band' will appear, which indicates the extent of the section to be magnified.

When the rubber band 'covers' the section you want, release the mouse button and the sample segment will appear in enlarged form in the sample window.

A Fast Zoom Mode is also provided to enable you to adjust magnification very quickly.

To do this, keep the [SHIFT] key on the computer keyboard pressed and click the mouse pointer within the sample display. The distance between the left edge and the position at which you click the mouse determines the size of the section to be zoomed.

Reference

Thus, for example, if you click the mouse at the centre of the window you will then see only half of the complete sample, whereas clicking the mouse near the right-hand edge of the window displays roughly nine-tenths of it.

To reduce magnification again, keep either the [ALTERNATE] or [CONTROL] key pressed and click the mouse anywhere within the sample display. The zoom factor will be automatically reduced by half (if you could previously only see a quarter of the sample, this operation makes half of it visible on the screen again).

Zoom Mode remains active until you either switch it off again by clicking on the Zoom Icon or change to another mode (Block, Loop, Marker).

If you click on the Zoom Icon with the right mouse button, the display will present the whole sample. This corresponds to the Full Wave Display function in the View Menu.

Block Define Mode



Block Define Mode allows you to define a block.

A block can be modified, copied or erased using the block function icons, without these operations affecting the rest of the sample.

Block Mode is indispensable for precision sound editing. The Frequency Domain Editor can also only be used to analyze a previously defined block segment.

To define a block, click on the Block Define Icon and move the mouse pointer into the active sample window. There it will take on the shape of a stylized block.

Reference

Move the pointer to the start of the segment you want to work on, press the left mouse button and move the mouse to the right. A 'rubber band' will appear which indicates the extent of the block area.

When you are satisfied with the area defined in this way, release the mouse button and the block will be displayed in the sample window in inverse video (white on black). You can then begin editing the block with various block functions.

A block can be defined in each of the different sample windows; AVALON also 'remembers' where blocks have been defined if you leave the editor. However, they are not saved to disk with other sample data.

Blocks can be defined at any zoom level. This enables you to edit very small areas of a sample in a very precise fashion. It is thus possible to 'zoom out' a single waveform from a long sample and then define it as a block for further use.

There is also a Block Fine Adjust function which makes it possible to subsequently alter the points at which a block begins and ends after it has been defined. You can do this by holding down the [SHIFT] key on the computer keyboard and then clicking the mouse near the beginning or end of the block - depending on which you would like to change. You can then make fine adjustments to the position of the start or end point whilst the mouse button remains pressed. When you are happy with the new position, release the mouse button and the block will be redefined.

Block Mode remains active until you either switch it off by clicking on the Block Define Icon a second time or change to another mode (Zoom, Loop, Marker).

Clicking on the Block Define Icon with the right mouse button erases the block markers. This corresponds to the Hide Block Marks function in the View Menu.

Block Function



This icon is used to call the pop up menu containing the block functions.

Block functions can only be used if you have previously defined a block using the Block Define Icon. Then clicking on the Block Function Icon calls the pop up menu with the various block functions.

To select an option you have to keep the left mouse button pressed. Release the mouse button when the pointer is positioned at the desired function.

Most block functions correspond to the sample functions of the same name; Fade In Block thus works in exactly the same way as Fade In Sample.

As a rule, however, you should tend to work with block functions as they enable you to edit samples in a much more precise fashion. For example, you will probably not want to 'smooth out' a 3-second piano sample that breaks off abruptly at the end by using Fade Out Sample as the complete sound then has a new, unnatural envelope. Much better - and more realistic - results can be achieved by defining the last 500 milliseconds of the sample as a block and then fading out only this part of it.

In addition to these functions AVALON also provides you with various functions to copy, cut, add and subtract blocks. They enable you to combine different sample values and sample sections in various ways.

In the following sections we will not describe in detail block functions which are equivalent to 'normal' sample functions, ie to get information on how to use Fade In Block see Fade In Sample.

Copy Block

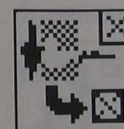


This function copies a defined sample block into a temporary buffer in memory. The data stored there can be read out by the functions: Insert Block, Overwrite Block, Add Block and Subtract Block. When a block is copied, it is not removed from the original sample, but simply duplicated in memory.

Please note that the Copy Block function uses the clipboard as its storage buffer. This means that you shouldn't leave any irreplaceable samples on the clipboard if you want to use this function as they would be overwritten.

You can also copy a block to another sample window. To do this, move the pointer to the appropriate block, press the left mouse button and drag the block to the desired position - with the mouse button pressed. This can be an empty window ('UNTITLED') or one that is already occupied. In the latter case, the block is inserted into the destination sample at the position where the block beginning is located when you release the mouse button.

Cut Block



This function cuts a block out of a sample and stores it in the buffer. The block is thus removed from the original sample and only exists in the buffer.

The Cut Block function also uses the MP clipboard as storage buffer.

Reference

Erase Block



This function deletes a block from the sample. The section of the sample behind the block is moved forward.

Erasing or cutting blocks generally leads to audible clicks within a sample. The Erase/Cut Mode option in the Edit Menu enables you to suppress this by applying a programmable crossfade function. The 'joins' are then faded into one another within a freely definable range.

You can also erase a block by doing the following: move the pointer onto the block, press the left mouse button, then drag the block out of the sample window onto the grey screen background and release the mouse button. The block will then be erased from the sample.

Fade Out Block



Corresponds to the Fade Out Sample function.

Reverse Block



Corresponds to the Reverse Sample function.

Reference

Fade In Block



Corresponds to the Fade In Sample function.

Insert Block



This function allows you to re-insert a block - that was previously stored in memory using Cut Block or Copy Block - into a sample. The block is inserted at the position of the block start marker in the destination sample.

Any data located behind the block start marker in the destination sample is moved back behind the inserted block.

Overwrite Block



This function allows you to re-insert a block - that was previously stored in memory using Cut Block or Copy Block - into a sample. The block is inserted at the position of the block start marker in the destination sample.

In contrast to Insert Block, the Overwrite Block function will overwrite the area behind the point where the block has been inserted.

Reference

There are two different possibilities here:

- The block to be inserted is shorter than or the same length as the block defined in the destination sample. In this case, only a segment as long as the inserted block is overwritten.
- The block to be inserted is longer than the block defined in the destination sample. In this case a dialog box will appear and you can:
 - overwrite the area behind the defined block (OK),
 - limit the size of the block to be inserted to the length of the defined block in the destination sample (LIMIT), or
 - abort the process (CANCEL).

Negate Block



This function turns the phase of the recorded signal within a block. This means that each positive value is transformed to negative and vice versa.

The function can be used, for example, to match two waveforms that are out of phase.

Add Block



This function allows you to add a block - that was previously stored in memory using Cut Block or Copy Block - to a sample.

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Reference

The stored block is added to the destination sample starting from the position defined by the block start marker.

Subtract Block



This function allows you to subtract a block - that was previously stored in memory using Cut Block or Copy Block - from a sample.

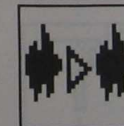
The stored block is subtracted from the destination sample starting at the position defined by the block start marker.

Optimize Block



Corresponds to the Optimize Sample function.

Replicate Block



This function allows you to reproduce an area which has been defined as a block between one and 999 times at the end of the block.

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Reference

Replicate Block thus represents a useful shortcut to carrying out the following operations: defining a block, storing it in memory using Copy Block, re-inserting using Insert Block, and then repeating this a number of times. You can thus, for example, 'stretch' the sustain phase of a string sound to any length you wish - an excellent alternative to using loops with samplers whose loop and sample end points are identical (eg AKAI S900). It also enables you to transform a single waveform into a longer sound for further editing.

The number of replications is set via the Replicate Mode option in the Edit Menu.

Delay Block



This function provides you with a programmable echo facility that enables you to delay or predelay a sample block and then mix it with the original signal.

This function is extremely flexible as it can work with delays ranging from a single sample value up to a period of around 10 seconds.

You can determine the number of repetitions and their volume via the Delay Mode option in the Edit Menu.

Eliminate DC From Block



This function removes DC components from a block.

DC components reduce the dynamic range of a sample. Samples with DC components can be identified by their unsymmetrical display (they look as if

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Reference

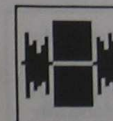
they are shifted in proportion to the zero-line). Eliminating DC components from a signal thus allows you to improve signal to noise ratio.

Keep Block



This function reduces all sample values outside the block to zero - ie it erases the whole sample with the exception of the block. It is extremely useful when you want to isolate a specific passage in a long sample.

Mute Block



This function decreases all sample values within the block to zero - ie it creates a pause that lasts as long as the defined block. It is very useful when you want to eliminate unwanted noise.

Filter Block



Corresponds to the Filter Sample function.

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Analyze Block

This function subjects a block to Fourier analysis. When this is completed, it is automatically transferred to the Frequency Domain Editor.

The analyzed block is then no longer displayed 'two dimensionally' (within a time/amplitude grid), but is represented in three dimensional form (time/frequency/amplitude) and can be edited with the functions offered by the Frequency Domain Editor.

If the Frequency Domain Editor is not present in memory, it is automatically loaded when this function is called.

Fast Fourier analysis and the resynthesis of a sample after editing in the Frequency Domain Editor are the most complex data processing operations carried out by AVALON. You should thus expect the analysis of one second of sample material to take around one minute of processing time. This is also why there is no Analyze Sample function. As the same amount of time is also required for resynthesis when you have finished editing your sample material in the Frequency Domain Editor it is generally advisable to proceed as economically as possible - ie carefully select only the section of the sample that you really want to work on before you transfer it to the editor.

If you really need to analyze the complete sample, however, you can always do so by using the Block = Sample function in the Display Menu and then activating Analyze Block.

Please refer to the "Frequency Domain Editor" section for further details.

Block Time Correction

The Time Correction function - described in detail in the "Time Correction" section of the "Edit Menu" chapter - can also be applied to blocks. This can be very helpful, for example, when making fine adjustments to a long vocal passage.

If the Time Correction module is not currently in memory, it is automatically loaded when you call this function.

Play Block

This function plays back the defined block.

Reference

Stereo Mode



Stereo Mode allows you to link two samples together to form a stereo sample.

As already mentioned above, stereo samples can also be edited in the Time Domain Editor. Then all functions affect the two components of a stereo sample.

Two preconditions must be fulfilled to carry out stereo editing.

- The samples that you want to combine into a stereo sample must follow one another within a sample bank (ie they must be stored at sample locations 1&2, 3&4, 5&6 or 7&8).
- Only one of the two samples can be in the editor. Otherwise the following error message will appear: 'Other sample has already been opened for Mono Editing'. If this occurs, you have to close the other sample window before you can begin stereo editing.

A dialog box will appear if the two samples are of different lengths (something that should not be the case for a stereo sample).

In Stereo Mode both samples are displayed one above the other and their names appear in the common Move Bar of the stereo window.

If Stereo Mode is active, you can call the stereo functions described below via the Stereo Mode Icon. When you click it on with the left mouse button, a pop up menu will appear that contains the stereo functions.

To select an option you have to keep the left mouse button pressed. Release the mouse button when the pointer is positioned at the desired function.

It must be stressed that these are additional functions - the real advantage of Stereo Mode is that all 'normal' editing commands affect both sample components.

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Reference

Edit Left



This enables you to cancel the link between the two samples and again separately edit the left sample in Mono Mode.

Thus, for example, if you accessed a vibraphone sample - recorded in stereo - in Stereo Mode via its right channel and have filtered, optimized and looped the two samples together, Edit Left Sample allows you to temporarily 'get into' the left sample and edit it separately in order to increase the stereo effect (eg by shifting it slightly, etc).

Edit Right



This enables you to cancel the link between the two samples and again separately edit the right sample in Mono Mode.

Erase Left



This function erases the lefthand component of a stereo sample. After it has been erased the program automatically switches back to Mono Mode as there is no longer a second channel available.

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Reference

Erase Right



This function erases the righthand component of a stereo sample.

Compare Left And Right



This function compares the two parts of a stereo sample and then informs you whether they are identical or different. It will also check whether they are the same length.

This function provides you with a simple control mechanism if you have recorded a sound in stereo, but are not really sure whether it is a real stereo signal.

Copy Left To Right



This function can be used to copy the left sample element to the right. The sample previously stored as the righthand channel is then erased.

Reference

Copy Right To Left



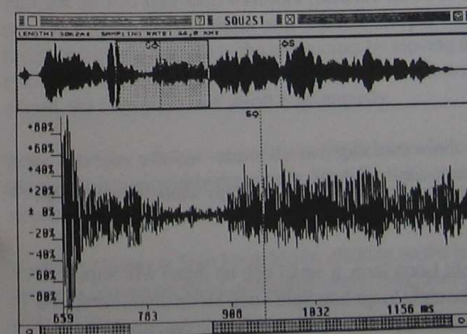
This function can be used to copy the right sample element to the left. The sample previously stored as the lefthand channel is then erased.

Overview Mode



In Overview Mode a second, smaller window is displayed above the sample window itself. This upper window always displays the complete sample whereas the sample window below can be used to display sections of the sample at different zoom levels.

The zoomed section of the sample is displayed in the overview window as a grey shaded area. This means that you can always see where you are within a sample.



Reference

For example, you might need to find a background noise (a click) within a 5-second sample in order to 'rub it out' in Draw Mode (see below).

To do this, you switch to Draw Resolution - where each point on the screen corresponds to one sample value.

You now have a very high resolution display, but you can no longer see where the click is located.

If you activate Overview Mode, you will see the enlarged area represented in the overview window in grey, and you will then be able to find and eliminate the undesirable noise very quickly.

Zoom, Zoom Out, Block and Block Fine Adjust operations can also be carried out in the overview window.

Simply define the zoom or block area as normal (see the relevant manual sections) - not in the 'large' sample window, but in the overview window.

<It often makes more sense and is more convenient to first define block and zoom areas fairly roughly in the overview window. You can always see the whole sample displayed there and orientation is easier. You can then carry out finer adjustments in the 'real' sample window.

Overview remains active until you click on the Overview Icon a second time.>

Split Loop Mode



Split Loop Mode allows you to edit loops within a display window system that has been specially designed for this purpose.

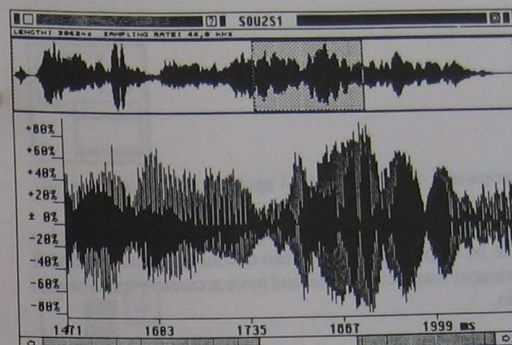
When you click on the Split Loop Icon, a small pop up menu will appear. It allows you to click one of the eight loops into Split Loop Mode (here, the option for sample start and end points appears in grey and has no function).

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Reference

In Split Loop Mode a sample window is divided into the following sections:



- an overview window (above) that displays the complete sample with the loop start and end points,
- a window (left) that displays the sample section in front of the loop end point, and
- a window (right) that displays the sample section behind the loop start point.

The screen is arranged like this to display the crucial transition area in the same way as it is heard, ie with the loop end running into the loop start.

Beneath the two split loop windows there are scroll bars whose function and appearance is similar to those in a 'normal' sample window. You can make rough or fine adjustments to the loop points by moving the white scroll box or clicking on the arrow boxes. If you click on one of the arrow boxes for a longer period, the loop point will scroll continuously.

In the overview window - above the two split loop windows - you can see where the loop points are within the sample. Modifications cannot be carried out in this window.

If you are already in Split Loop Mode, clicking on the icon again will call a pop up menu containing three additional icons: Loop Find, Crossfade and Exit.

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Reference

Loop Find



This function determines a - mathematically - optimum loop.

Clearly this function cannot find a perfect loop within sound material which is totally unsuitable to be looped. However, it can undoubtedly test and define loops in a long continuous sound much faster and more accurately than a human being could ever do.

The parameters for this function can be set using the Loop Mode option in the Edit Menu.

Crossfade



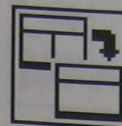
This function modifies a predetermined area around each of the loop points so that there is a 'smoother' transition between the two.

A precondition for this, however, is that the loop points selected match one another to a considerable degree; the gradient and amplitude of the waveform at both sides of the transition point should not be too different. On no account should the two points be out of phase (eg a 'wave valley' may not be immediately followed by another).

The parameters for this function can be set using the Loop Mode option in the Edit Menu.

Reference

Exit



This icon enables you to leave Split Loop Mode.

Zoom In



This function increases the display magnification along the vertical (amplitude) axis without shifting the horizontal (time) axis. This enables you to magnify the amplitude of a sample segment - if its volume is very low.

Sample Info

Sample Information		CANCEL	OK
Name :	TRIANGLE	PPP Standard Triangle	
Length :	28298 Samples = 468 ms	51000 Sample (05/05/91) by Pete	
S.-Rate :	44100 Hz	Use: 'Ratio', Track 1	
Pitch :	62		
Format :	12 Bit		
Modified :	☐		
Marker #1 :	0	START POINT	END POINT
Marker #2 :	0	Sample :	0 28298
Marker #3 :	0		
Marker #4 :	0		
Marker #5 :	0		
Marker #6 :	0		
Marker #7 :	0		
Marker #8 :	0		

Clicking on this icon calls the Sample Info Dialog Box. It is described above in the "Mapping Page" section under the heading "Sample Icon > Info Icon".
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Play Sample



The Play Icon plays back the sample in the active window.

From within the TDE - as on the Mapping Page - it is also possible for you to playback samples directly - ie without having to first transfer them back to a sampler. The great advantage of this is that you are immediately able to monitor any modifications made to a sample.

Depending on the setting in the Preferences Dialog Box, clicking on the Play Icon will playback either:

- the whole sample
- a defined block (see above), or
- the zoomed area.

The PLAY function can also loop samples. This means that if the sample to be played contains a sustain loop, if desired it can be repeated for as long as you keep the left mouse button pressed. You can determine which loop is to be sustained in the Loop Mode Dialog Box.

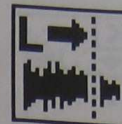
Clicking on the Play Icon with the right mouse button will always play the whole sample.

Zoom Out



This function cancels magnification of the vertical (amplitude) axis, without changing the horizontal (time) display.

Set Loops



Set Loops enables you to set up to eight loops and the start and end points of the sample as a whole.

Start/End	
Sus	Rel
LP 3	LP 4
LP 5	LP 6
LP 7	LP 8
Set Loop Points	

Clicking on the Set Loops Icon calls a pop up menu that contains entries for the start and end points of the eight loops (Sustain, Release, 3, 4, 5, 6, 7, 8) and the sample as a whole.

Clicking on one of these options enables you to determine the appropriate start/end points within the active window - press the left mouse button to set a start point and the right mouse button for the corresponding end point.

As soon as you move the mouse pointer into the sample window it takes on the form of the relevant icon and you only have to press the right or left mouse button at the desired position.

The start and end points you define will then be marked by a dotted line with an appropriate symbol at the top.

You can change the position of loop (or sample) start and end points as often as you wish.

To delete a loop point, simply click the mouse outside the active window or set its position to zero using the Sample Info Dialog Box.

Set Loops remains active until you switch to another mode.

Reference

Set Markers



Set Markers enables you to set up to eight markers.

AVALON allows you to position a maximum of eight markers within a sample window.

They can be used to mark important positions within a complex sample for subsequent editing (eg alternative loop points or potential 'loop transitions').

Clicking on the Set Markers Icon calls a pop up menu containing an entry for each of the eight markers.

Clicking on one of these entries enables you to set the appropriate marker by clicking the mouse inside the active window.

As soon as you move the mouse pointer into the sample window it takes on the form of the relevant marker and you only have to click the mouse at the desired position.

Markers are indicated by a dotted line with a marker symbol at the top.

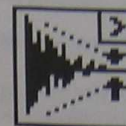
You can change a marker's position as often as you wish.

To delete a marker, simply click the mouse outside the active window or set its position to zero using the Sample Info Dialog Box.

Set Markers remains active until you switch to another mode.

Reference

Fade Out Sample



This function enables you to fade out a sample, ie within a specific time period the amplitude of the signal is reduced from its original volume (at an initial point) to zero (at the end).

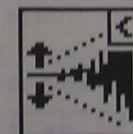
The Fade Mode option in the Edit Menu allows you to determine how this function modifies the signal.

Reverse Sample



This function reverses a sample along the time axis so that it is played backwards.

Fade In Sample



This function enables you to fade in a sample, ie within a specific time period the amplitude of the signal is increased from zero (at the start point) to its full volume (at a specific point).

The Fade Mode option in the Edit Menu allows you to determine how this function modifies the signal.

Reference

Truncate Sample



This function can erase either

- the area in front of the sample start point,
- the area behind the sample end point, or
- both.

The Truncate function enables you to cut out unneeded sections of a sample and thus reduce the amount of memory it occupies.

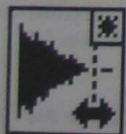
Optimize Sample



This function increases a sample's amplitude values so that its highest volume reaches 100% - or any other level you choose to select. It thus enables you to digitally amplify (or dampen) a sample.

In addition to this, the function can also be used to cut out extreme signal values.

Resample Sample



This function recalculates a sample on the basis of a specific factor.

Reference

In contrast to changing the sampling rate, which only affects sample playback, the Resample function actually alters the sample values themselves.

Resampling can be used to

p reduce sample data (eg if you sample a bass drum sound at 20 kHz although it only contains frequencies up to 10 kHz, resampling would make it possible to reduce its size by half and thus considerably decrease the amount of space it takes up.)

p modify sampling rate (eg if you want to convert a cymbal sound sampled at 30 kHz on a ROLAND S50 to the 44.1 kHz sampling rate used by an AKAI S1000).

The parameters affecting this function can be accessed via the Resample Mode option in the Edit Menu.

Filter Sample



This function filters a sample.

You can set the parameters for the filter via the Filter Mode option in the Edit Menu.

Re-envelope Sample



Using this function you are able to draw a line in the sample window which is then used as the basis for recalculating the signal levels. It thus enables you to give a sample a new envelope.

Reference

When you click on the Re-Envelope Icon you can move the mouse cursor into the sample window and then - with the left mouse button pressed - draw a line covering the entire sample or only one part of it.

As soon as you release the mouse button the amplitude of the 'redrawn' area is modified in line with the values you 'sketched in'.

Draw Mode



This mode enables you to edit individual sample values or whole areas of a sample with the mouse.

Draw Mode is particularly useful when you want to

- erase or correct small, but visible faults (eg clicks from sampled records),
- smooth out poor loop transitions, or
- erase disturbances (background noise) between specific parts of a sample.

Draw Mode can only be activated at a display resolution of 1:1 (Draw Resolution). At this resolution each vertical line on the screen represents a single sample value.

Draw Resolution can either be activated via the option in the View Menu of the same name or by clicking on the Draw Mode Icon with the right mouse button.

A full-width sample window on a standard monitor is 512 pixels wide. This means that you can see a total of 512 sample values (at a sampling rate of 44.1 kHz this represents a section of the sample lasting 11.6 ms).

Reference

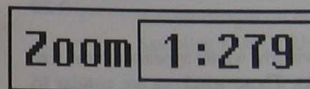
In order to draw sample values with the mouse, move the pointer to the desired position in the sample window and press the left mouse button. You can then define new sample values by moving the mouse whilst keeping the mouse button pressed. However, you shouldn't move the mouse too fast, as this could lead to errors.

Clearly, it is also possible to 'draw' completely new waveforms, however, in practice this has proved to be a rather ineffective method for synthesizing sounds. If you nevertheless want to try it out, there is no need to draw several hundred wave cycles. Simply draw one, define it as a block and reproduce it as many times as you need using the Replicate Block function.

Draw Mode remains active until you click on the Draw Icon again or switch to another mode or zoom level.

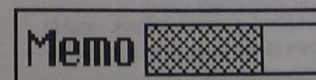
Clicking on this icon with the right mouse button selects Draw Resolution.

Zoom Factor Display



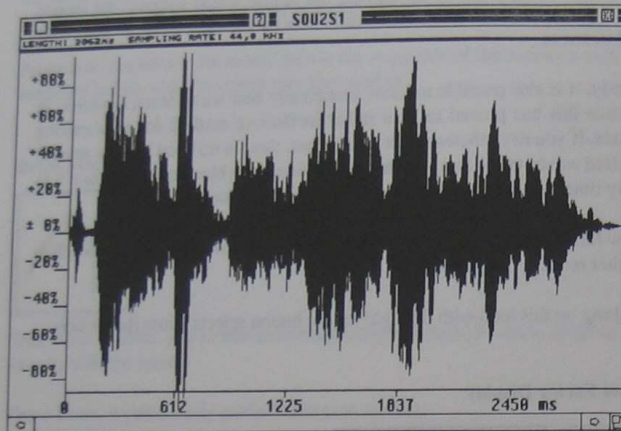
The Zoom Factor display indicates the relationship of screen points to sample values. '1:200' here means that 200 sample values are represented by one screen point, accordingly '4:1' means that each sample value is represented by a band of 4 screen points.

Memory Display



This screen provides information on memory usage. The grey part of the display indicates how much memory is already occupied.

Sample windows



Each sample (or stereo sample) has its own window whose appearance and size can be influenced by various basic settings. These windows are similar to 'conventional' GEM windows.

Only the last window called is active. To select another window click on either

- the window itself or
- the appropriate window button (at the top edge of the toolbox).

All menu and toolbox functions affect the currently active sample window.

You can access a maximum of eight sample windows - a complete bank - within the editor. However, opening all eight windows at the same time is not to be recommended as the screen then becomes rather cluttered.

Differences to GEM windows

AVALON sample windows differ from normal GEM windows in the following ways:

- The Move Bar contains two new elements (in addition to the Close Box, the sample's name and the Zoom Box):
 - Info
Clicking on this button calls the Sample Info Dialog Box (see "Sample > Info")
 - Modified
This button appears when the sample is modified in some way after loading.
- There is an Info Line beneath the Move Bar.
If the window contains a sample, the Info Line indicates the following:
 - Sample Length (in ms)
 - Sampling Rate (in kHz)

If an empty window has been opened, the Info Line indicates:

- total memory available

In Split Loop Mode it shows

- the number of the loop and its length in sample words.
- There is no vertical scroll bar as it would make little sense to move the display in a vertical direction.

Time Domain Editor menus and dialog boxes

Desk Menu

About

This function is identical to the function of the same name in the Desk Menu on the Mapping Page.

File Menu

Get Memory Info

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Mouse Speeder

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Screen Settings

This function is identical to the function of the same name in the File Menu on the Mapping Page.

MIDI Keyboard

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Preferences

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Call FD Editor

This function enables you to call the Frequency Domain Editor (FDE).

If the Frequency Domain Editor was not automatically loaded by AVALON when the program was run, selecting this function will load it into memory.

A File Selector will appear as <it is impossible to create a frequency spectrum within the Frequency Domain Editor from scratch. You can only enter it directly by loading one from disk. If this is not possible or you change your mind, then simply click Abort.

The sample in the currently active window is erased when you leave the Time Domain Editor using Call FD-Editor and load a spectrum from floppy disk or hard disk.> The reason for this is that when you modify the loaded spectrum and resynthesize it into a sample, the new sample is copied to this sample location.

The Frequency Domain Editor is described in detail in a special section of this manual.

Time Correction

Intelligent Time Correction

Time Correction Factor: %

☐ Quick Mode

☐ Effect Mode (>30%)

OLD SAMPLE

Time : ms Speed [bpm]

Length: Samples

NEW SAMPLE

Time : ms Speed [bpm]

Length: Samples

The Time Correction module enables you to shorten or lengthen a sample without modifying its pitch. Its programming leads to processing speeds and a quality of results that are comparable to those of much larger systems.

If the Time Correction module was not automatically loaded by AVALON when the program was run, selecting this function will load it into memory.

The Time Correction function can also be used to edit blocks - as Block Time Correction. This is very useful when you want to carry out precise editing within a long sample.

When you call the Time Correction module, the sample in the active window is analyzed. AVALON first ascertains the best algorithms to apply in order to stretch or compress the sound, and then carries out the necessary calculations.

The following parameters can be set:

- **Stretch Factor**
This is the factor by which the length of the sample is multiplied.
If this factor is
 - smaller than 1, the sample is shortened.
 - larger than 1, the sample is lengthened.
- **Quick Mode**
Compressing or stretching a long sample can take considerable time. Quick Mode allows you to have a 'rough' calculation carried out (ie it is not particularly accurate, but much faster).

Note: Modifying the length of a sample without changing its pitch is only possible within a relatively narrow margin on either side of the original length (here: 30%). Thus, when you select a stretch factor below 70% or over 130% the Effect Mode button is automatically activated. This indicates that Time Correction will probably lead to a result that will deviate from the original signal to a certain extent and the resultant sound data can thus only be used as an effect.

The following information is also displayed:

- Old Sound
- Time
Length of the original sample.
- Length
Number of sample values in the original sample.
- Tempo
The tempo base.
- New Sound
- Time
Length of the new sample.

Reference

- **Length**
Number of sample values in the new sample.
- **Tempo**
The tempo for the new sound on the basis of the tempo set for the old sound.

Using the tempo basis is very useful when compressing or stretching complete passages which have to 'fit' in a larger context.

Please note that the details about the length of the new sample are only for orientation purposes. The actual length depends on the sound material involved and can deviate from the predicted value by several per cent.

Erase FD Editor

This function allows you to remove the Frequency Domain Editor from memory and thus make more memory available for sample data.

Erase Time Corr.

This function allows you to remove the Time Correction module from memory and thus make more memory available for sample data.

Reference

Detect Pitch

Pitch Detection

Note : F#6

Cents: -50 -25 0 +25 +50

Frequency: 1500 Hz

Period : 20 Samples

Quality: LOW MID HIGH

>TAKE! OK

The Detect Pitch function determines the pitch of a sample.

If the pitch could be ascertained, a dialog box with the following information appears:

- **Fine Tune**
The deviation of the pitch detected from standard pitch in cent steps (120/semitone).
- **Note**
The detected pitch.
If you click on the >TAKE! button at the lower edge of the dialog box, this value will be transferred to the Sample Info Dialog Box.

- **Probability**
This value indicates the probability (in %) that the pitch detected corresponds to the actual pitch.

If it was not possible to detect the pitch, you will receive the message 'Could not detect a proper pitch'.

Quit Editor

This function enables you to leave the TDE and return to the Mapping Page.

Reference

Edit Menu

Undo Last Edit

This function enables you to cancel the last data-modifying operation carried out.

Please note that this is only possible if you have activated the Backup Last Edit option in the Preferences Dialog Box.

Clear Undo Buffer

This function in the Edit Menu clears the Undo buffer, ie the temporary storage location which contains the version of the sample as it was before it was last altered (see also "Undo Last Edit").

Its use should only be necessary - and only really makes sense - if there is no longer enough memory available (eg if you cannot carry out an Insert Block operation because the computer's memory is already full).

If you use this function, you also give up the option of being able to undo the last modification to your sample. For this reason it is generally better to make more room available by removing from memory a sample that is not being used (Don't forget to save it to disk before doing so!).

If Backup Last Edit Mode in the Options Menu is not active, then clearly there is no undo buffer and this function serves no purpose.

Clear Clipboard

This function empties the clipboard.

As the contents of the clipboard buffer are lost, this function should only be used if you are experiencing serious problems due to insufficient memory.

Reference

Sample Info

This function calls the Sample Info Dialog Box.

The Sample Info Dialog Box displays the following information on the sample in the active window:

- its name,
- a short comment (5 lines of 30 characters)
- the length of the sample (in sample words and milliseconds)
- the sampling rate (in kHz)
- the start and end points of the sample itself as well as up to 8 loops (in sample words)
- oop status (on/off)
Only the start and end points of active loops are displayed.
This improves the clarity of the display.
- reverse (on/off)
- 8 markers
- a Modified flag that indicates whether the sample has been edited since it was loaded/saved.

When you save a bank (see "Bank Icon"), the only samples saved are those whose Modified flags are active.

You can modify all the data in this dialog box with the exception of the sample's length.

Reference

Clipboard Info

This function enables you to call the Clipboard Info Dialog Box.

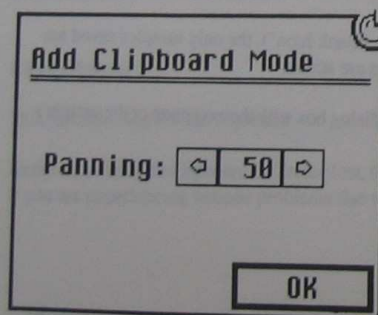
The Clipboard Info Dialog Box displays the same information as the Sample Info Dialog Box.

Block Info

This function displays information on the current block. The following is displayed:

- **Block Start Position**
This value is relative to the start of the sample in which the block was defined.
- **Block End Position**
This value is also relative to the start of the sample in which the block was defined.
- **Block Length**
The length of the block is calculated on the basis of the two values given above and the sampling rate of the sample in which the block was defined.

Add Mode



Here you can set the volume ratio between the clipboard buffer and the destination sample for using the Add Block function.

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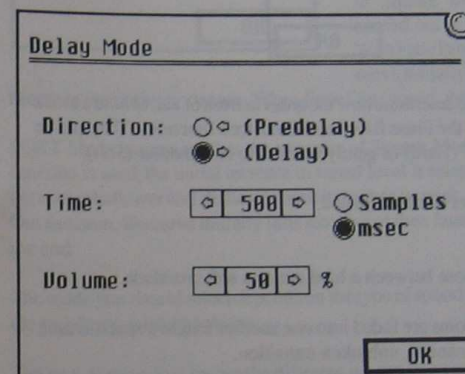
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Reference

If the Panning value is set at

- 0%, the clipboard is not added to the destination sample at all.
- 50%, the clipboard and destination sample are mixed at a volume ratio of 1:1.
- 100%, the clipboard buffer erases the appropriate area of the destination sample.

Delay Mode



Here you can set the parameters for the Delay Block function.

These are

- **Direction**
Here you can determine whether the block is to be moved back ('Delay') or forward ('Predelay').
- **Time**
The extent of the 'delay' or 'predelay'.
- **Volume**
How loud the echo(es) should be in relation to the original sample.

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Reference

Erase/Cut Mode



This option allows you to determine how the areas in front of and behind a block are joined when you use the Erase Block and Cut Block functions. They can be simply 'spliced together' (Hard) or gently faded into one another (Soft).

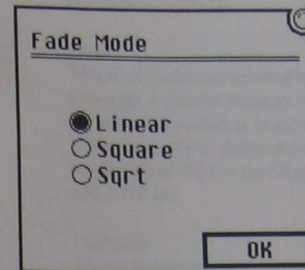
The following parameters can be adjusted:

- **Hard/Soft**
Here you can choose between a hard cut or a soft crossfade.

In Soft Mode the joins are faded into one another within a specific time period to create a natural, unbroken transition.
- **Time**
This is the transition period used in Soft Mode: the longer the time, the 'smoother' the join.

Reference

Fade Mode



The Fade Mode settings influence the Fade In and Fade Out functions.

In Linear Mode the signal is faded in/out in a linear fashion, ie half way through the fade in/fade out period the signal level also amounts to half its original volume.

In Square Mode fade in/out is carried out in an exponential way. With Fade In operations the volume curve initially rises slowly but then

becomes increasingly steeper. When Fade Out is used, the opposite is the case.

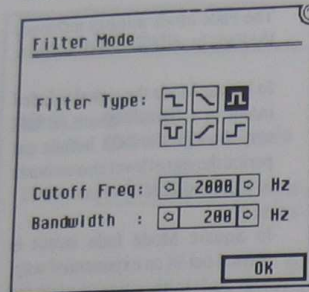
SQRT Mode (square root) is the opposite of Square Mode. When the Fade In function is used, the initial increase in signal level is relatively steep but it then becomes shallower and shallower until it reaches its peak. When using the Fade Out function, the curve initially falls slowly but then fades out very abruptly at the end.

The mode you should select depends on the type of sound material involved and the result you wish to achieve.

The best thing to do is try out the different modes on a simple sample to find out what effect they have.

Reference

Filter Mode



This is where you can program the Filter Sample and Filter Block functions.

The parameters are:

- Cutoff Frequency
- Bandwidth
- EQ Type

The following types of filter are available:

- Lowpass
All frequencies above the cutoff frequency are eliminated. This is the most useful filter type.
- Lowpass Shelving
Similar to Lowpass, but the effect of the filter is less abrupt. The frequencies above the cutoff frequency are faded out over a specific frequency range.
- Peak
This filter (which is really a combined lowpass and highpass filter) deletes the whole signal beyond a specific band on either side of the selected cutoff frequency - its width is determined by the Bandwidth parameter.

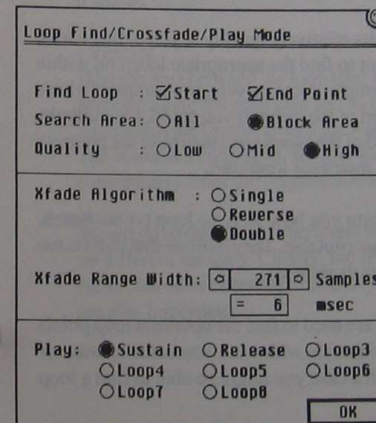
Reference

Thus, if you define the cutoff frequency as 1,000 Hz and set a bandwidth of 400 Hz, the filter will remove all frequencies lower than 800 and higher than 1,200 Hz.

- Notch
This filter (again a combined lowpass and highpass filter) allows you to eliminate a specific frequency band around the cutoff frequency from the signal - its width is determined by the Bandwidth parameter. In this case, if you define the cutoff frequency as 1,000 Hz and set a bandwidth of 400 Hz, the filter will remove all frequencies between 800 and 1,200 Hz.
- Highpass
All the frequencies below the cutoff frequency are erased.
- Highpass Shelving
Similar to Highpass, but the cutoff is less abrupt.

Clearly, it is possible to define and use different filters one after the other. This enables you to remove unwanted frequencies very precisely.

Loop Mode



This option offers powerful functions allowing you to create smooth, click-free loops.

- Find Loop
This item is used to program the Find Loop function (that is called via the Split Loop Icon when in Split Loop Mode).

Reference

The following parameters can be set:

- **Find Start/End**
This setting determines whether a loop start and/or end point is to be found.

If you want AVALON to find a loop start point within a specific block rather than somewhere in the whole sample, the loop end point must not be within the same block - otherwise AVALON might position the loop start behind its end point.

If the program has to find both loop points, it will set the end point at the next zero position and determine a matching start point.

- **Search Area**
This setting determines whether the appropriate loop point should be found somewhere in the whole sample (All) or only within a defined block.
- **Quality**
Here you can determine the accuracy with which loop should be found. Naturally, greater precision increases processing time. You can choose between three quality levels (Low, Mid and High).

You should be able to get good results relatively quickly if you define a loop start point and then allow the program to find the appropriate loop end within a fairly large block at the end of the sample. Alternatively, (for samplers whose loop ends always have to be at the end of the sample) you could, for example, set the release loop end point at the end of the sample and then have the program look through the whole sample for a matching loop start.

If a loop is found, AVALON will inform you how well the loop points match. Generally, values less than 90% are unacceptable. They indicate that there is too much variation within the sample area examined.

Please note that mathematical criteria are used to find the optimum loop points and the results may thus not always coincide with what you actually want or consider musically acceptable. In such a case you might be able to find a loop manually better than AVALON.

Reference

- **Crossfade**
This option contains the settings for the Crossfade function (that is activated by the Split Loop Icon when Split Loop Mode is active).

The parameters are:

- **XFade Algorithm**
Three different algorithms can be used to 'smoothen' the transition between loop points:
 - **Single**
If this algorithm is used, a specific area at the loop end is modified to match the loop start. The loop start itself is not changed. This technique is particularly suited for use with samplers whose loop and sample end points have to match as it does not result in any audible change at the loop start.
 - **Double**
Using this algorithm, the sample values around the loop start and end points are modified to produce a smooth loop. The advantage is that the changes are evenly distributed at both loop points, and as a result, the loop is less conspicuous within the sample as a whole.
 - **Reverse**
Here the area between the loop start and end are first copied to a buffer, reversed along the time axis and averaged with the original loop area. Afterwards the Double Crossfade Algorithm described above is applied. The result is a consistent, smooth loop. However, it can deviate considerably from the original sound.

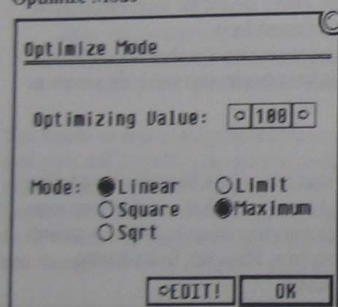
We would advise you to experiment with the various crossfade algorithms on different samples. Clearly, the length of the loop area is a decisive factor, but other criteria (particularly the dynamic range within the loop) are also important.

Reference

- **XFade Time**
This setting allows you to determine the size of the area at the loop points that should be adjusted. The larger this area is, the smoother and less noticeable the transition becomes. However, there always has to be an area on both sides of the loop points. Thus, for example, if you have set the loop end half a second before the end of the sample, the crossfade area must not be longer than half a second.
- **Play Loop**
This is the last setting in the Loop Mode Dialog Box. It is used to determine which loop should be sustained when the sample is played back. You can choose between all 8 loops (Sustain, Release and Loops 3-8).

If a loop is to be played back, then the appropriate Loop Status button must be active.

Optimize Mode



This dialog box allows you to set the parameters for the Optimize Sample function.

The settings are:

- **Optimize Mode**
There are three different modes for rescaling volume data.

Linear

In this mode all signal levels

are increased by the same ratio. This is the 'conventional' mode.

Square

If this mode is used, the amplitude values are increased in an exponential way, ie low volumes are not or only slightly amplified, whilst higher volumes are raised considerably.

Reference

Sqrt

In this mode amplification is based on a square root function. Low values are raised considerably and high values remain practically unaltered.

Please note that Square and Sqrt Modes do not only change the total volume of a sample, but also the interrelationship between the individual sample values themselves (and thus the frequency content).

- **Value**

This setting determines how high the maximum signal level should be.

Clearly, for the most commonly used application of the Optimize function, ie amplifying a sample to the optimum volume, this value should be set to 100%.

In addition to the normal ways of adjusting this figure - clicking the arrow buttons, using the value slider or keypad - you can also set it by the following means:

Click on the >EDIT! button in the lower righthand corner of the dialog box.

Two horizontal bars will then appear in the sample window. They indicate the maximum signal range and can be moved up and down with the mouse.

When you are satisfied with the setting selected, simply click on the left mouse button and the figure will be entered in the dialog box.

- **Take Value as Limit/Maximum**

In addition to its function as an amplifier, Optimize Block can also be used to cut out extreme signal levels.

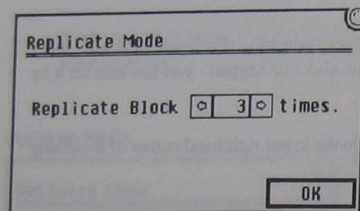
To do this, you have to switch over to Limit. The selected optimize value is then no longer the maximum level to which the sample is to be rescaled. Instead all sample values above this figure are cutoff, eg if it is set at 75%, all the amplitude values above this (76-100%) are 'reduced' to 75%, and the resulting signal is then raised to 100%.

Reference

This function should not be compared with an analog limiter that dampens an excessively loud signal. Here specific parts of a continuous signal are cut off - an operation that leads to a transformation of the waveform itself rather than the dampening of the sound as a whole.

This function is particularly useful when a low-volume sample cannot be optimized as it contains a few extreme values (eg transmission errors, clicks). The Limit function enables you to cut out such extreme sample values quickly and accurately.

Replicate Mode

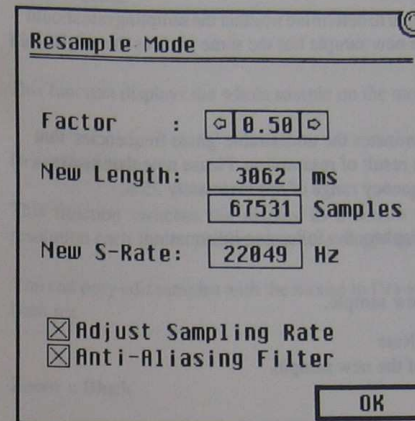


This dialog box determines the number of times a block is copied when you use the Replicate Block function.

The maximum setting allowed is 999.

Reference

Resample Mode



This setting affects the Resample function.

The parameters are:

- Factor
This is the factor by which the sampling rate is multiplied.
If this factor is
 - 1, the original sample is not changed.
 - less than 1, the sampling rate and resulting frequency range is reduced.
 - greater than 1, the sampling rate is increased.
However, frequency range is not increased as a result. It would be wrong to believe that a 25 kHz sample can be made extremely brilliant and sharp by doubling its sampling rate. The missing harmonics cannot be generated artificially (at least, not by this means); new sample values are merely inserted to maintain the original waveform.

Reference

- **Adjust Sampling Rate**
This function enables you to determine whether the sampling rate should be adjusted so that the new sample has the same pitch as the original sample.
- **Anti Aliasing Filter**
This optional filter eliminates the undesirable 'ghost frequencies' that sometimes appear as a result of resampling. Please note that its use results in a loss of frequency range of approximately 25%.

This dialog box also displays the following information:

- **New Length**
Length of the new sample.
- **New Sampling Rate**
Sampling rate of the new sample.

Goto Menu

The Goto Menu allows you to jump to specific points in a sample and can thus save you a lot of time.

The appropriate point is always displayed at the centre of the active window (unless it is so close to the beginning or end of the sample that this is impossible).

The current zoom level remains unchanged when you use a Goto function.

You can jump to the following points:

- Sample start/end
- Sustain loop start/end
- Release loop start/end
- Block start/end
- Markers 1-8

Simply click on the appropriate menu option and the relevant point will appear in the sample window.

Clearly, if a point has not been defined, you cannot jump to it in this way.

Reference

View Menu

Full Wave

This function displays the whole sample on the monitor.

Draw Resolution

This function switches the display to a screen resolution of 1:1. At this resolution each screen pixel represents a single sample value.

You can only edit samples with the mouse in Draw Mode if this resolution has been set.

Zoom = Block

This function enables you to magnify the display of a block, ie the display is zoomed so that only the block is visible in the sample window.

This function can only be used if a block has been defined.

Block = Sample

This function enables you to automatically define the whole sample as a block. This is necessary if you want to apply a function to a sample that can normally only be used on blocks, eg Block Copy, Cut, Insert, Add, Subtract and Analyze.

This function can also be called by clicking on the Block Function Icon with the right mouse button.

Hide Block Marks

This function removes the block markers (ie the marks at its beginning and end, not the block itself) from the screen.

Reference

This function is also activated by

- clicking on a block whilst keeping the [ALTERNATE] key pressed, or
- clicking on the Block Define Icon with the right mouse button.

X-Axis: Realtime/Samples

This function enables you to switch the scaling of the horizontal axis between milliseconds (calculated on the basis of the sampling rate) and sample words (individual sample values).

Generally, milliseconds are easier to work with, but precision editing sometimes requires scaling in words.

Zero-X-Snap

Activating this mode automatically moves all markers and sample, loop, block and zoom area start and end points to the next zero position.

This enables you to define very smooth loop transitions and zoom windows - without having to work very precisely - because you will always be working with complete waveforms.

In certain circumstances (eg at extreme magnifications or when setting loop points which are not meant to meet where there is no signal) it might be necessary to switch this mode off.

The only points automatically adjusted by Zero-X-Snap are those set after it has been activated. If you want to adjust any points set before this occurred, simply click them on once with the mouse.

Show X-Position

If you activate this mode, a display in the upper right corner informs you about the horizontal position of the cursor (either in milliseconds or sample words). This enables you to work very precisely.

Reference

When defining a zoom area an additional ' Δt ' (Delta t) display appears. It indicates the size of the area in milliseconds.

Y-Axis: Percent/Value

This function enables you to switch the vertical axis between relative and absolute scaling.

Relative scaling ranges from -100% to 100%.

Absolute scaling is based on AVALON's internal 16-bit sample format and ranges from -32768 to 32767.

As a rule, percentage values are easier to work with.

Y-Zoom

This mode is used to allow magnification along the vertical axis (with the appropriate functions).

If Y-Zoom is inactive, the vertical scale will show - depending on whether Auto Y-Zooming is on or off - either

- the full range (ie -100% to 100%) or
- the range determined by the highest amplitude in the visible section of the sample (see below).

Auto Y-Zooming

If this mode is active, the zoom factor for the vertical axis is automatically set in such a way that the highest amplitude currently visible reaches the top edge of the window. It is then no longer possible to zoom in closer using the Y-Zoom In Icon.

Reference

Show Scaling

This function activates the scaling of the active sample window.

Scaling is practically indispensable for precision editing. The units used can also be set within the View Menu.

Show Points

This function enables you to determine whether loop and sample start/end points should be displayed in the active sample window or not.

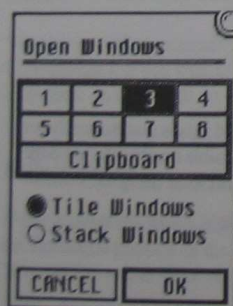
If you have set several loops, you may find that the display becomes rather cluttered. If this occurs, you can turn off loop display using this menu option. The loops are not erased, they are simply no longer displayed on the screen.

Show Markers

This function enables you to turn the display of markers in the active sample window on or off.

Windows Menu

Open



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The Open option enables you to determine which of the windows should be opened or closed.

When you select Open in the Windows Menu, a dialog box will appear with buttons for the eight samples of the current bank and the clipboard.

By clicking on these boxes you can decide which sample windows are to be opened.

Reference

The boxes of selected windows are represented in inverse video (white characters against a black background).

You can also determine whether the windows should be 'stacked' (Stack) or displayed next to one another (Tile).

When you close the dialog box, the windows you selected will be opened. Any other windows that were open will be closed.

Close Windows

This function closes the active window.

Zoom Windows

The Zoom Windows entry has the same function as the Zoom Box in the top righthand corner of each sample window.

This function allows you to switch between maximum window size and one that you can set at will. You can thus very quickly select one of four opened windows for editing and then reduce it back to its original size again afterwards.

Next Window

This option activates the next window.

This function can also be used to 'pull' a completely hidden window back to the 'top' of a stack.

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Reference

Redraw Windows

If display errors occur, this function enables you to cause the screen to be redrawn.

Extreme display errors, eg samples that shoot out beyond the edge of a sample window, indicate a system error of some kind. If this occurs, we would advise you to save your work and reboot your system.

Swap Windows

If this setting is active, the current window will be closed when a new window is opened.

This allows you to switch between different samples within the Time Domain Editor - instead of always having to open new sample windows.

Tile Windows

If several windows are opened simultaneously, this function allows you to arrange them on the screen in the optimum manner without them overlapping one another.

Stack Windows

This function 'stacks' several windows on top of each other so that you can switch between them at any time with the mouse. Nevertheless, AVALON will always display each window as large as possible.

Window Cache

This menu option is used to activate the window cache.

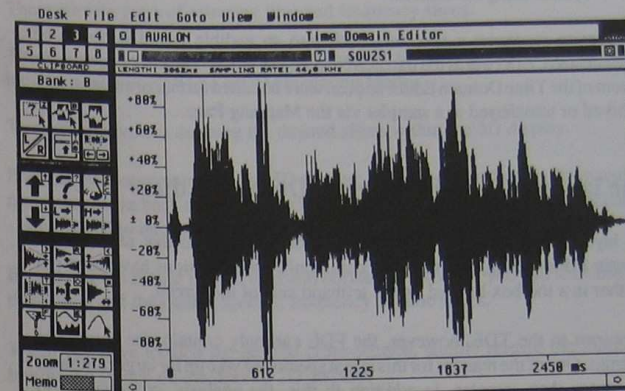
The window cache is a special buffer in memory that considerably increases the speed at which the screen is written when windows are moved or modified. Like all other memory-intensive functions, this cache can be disabled at any time.

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Reference

The Frequency Domain Editor



The Frequency Domain Editor (FDE) splits up a complex sound into its individual components (sine waves) on the basis of a Fourier analysis, and then displays them within a three dimensional grid (horizontal axis: frequency, vertical axis: amplitude, depth: time).

The frequency range registered by the analysis always amounts to half the sampling rate, ie if you sample a sound at 44.1 kHz, frequencies between 0 and 22,050 Hz are registered.

In the FDE modifications can be made on three levels:

- Pixel range: modification of the volume of an individual frequency within the smallest possible time segment.
- Micro range (with the aid of 'rubberbands'):
 - modification of an individual frequency along the complete time axis or
 - modification of all frequencies within the smallest possible time segment.

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- Macro range: modification of the complete frequency/amplitude/time system using 3D macros.

In order to reconvert a spectrum back into an audible sound it has to be resynthesized. Afterwards it is transferred to the two-dimensional time/amplitude system of the Time Domain Editor and can there be edited further or alternatively archived or transferred to a sampler via the Mapping Page.

The layout of the Frequency Domain Editor

The layout of the Frequency Domain Editor is similar to that of the Time Domain Editor. Here, too, the most commonly used functions have been put together in a toolbox located on the lefthand side of the screen.

In contrast to the TDE, however, the FDE can only contain one sample or spectrum. One of the reasons for this is that spectrums take up far larger amounts of memory than samples. In addition to this, the analysis, processing and resynthesis of frequency spectrums are such complex functions that it makes little sense to edit more than one spectrum at a time.

As a result, the FDE only has a large spectrum display whose size cannot be changed.

You cannot directly edit a three dimensional spectrum display - unfortunately a three dimensional mouse for editing spatial structures does not yet exist. However, what you do have here is a special 'cross-wire' that enables you to 'target' a frequency/amplitude and time/amplitude cross-section, ie two 'slices' of the 'wave mountain'.

The frequency/amplitude slice contains the amplitude values of all frequencies at a specific time point. It thus indicates how loud the frequencies in the sound spectrum are at a specific time. From now on we will refer to a cross-section of this kind as a frequency slice.

The time/amplitude slice contains the volume data of a specific frequency along its time axis, ie its envelope. This type of spectrum cross-section is referred to as a time slice.

Defining 2D Time and Frequency Slices

There are two ways of selecting time and frequency slices.

The first consists of clicking on the NEXT and PREV icons at the bottom of the toolbox whilst in 2D Time or 2D Frequency Mode.

The second involves defining the desired slices within the 3D display.

Position the mouse pointer in the 3D display, hold the mouse button down and move the mouse back and forth a little. You will see two lines begin to flicker - one parallel to the time, the other parallel to the frequency axis.

By moving the mouse in a vertical and horizontal direction you can shift both these lines and thus select specific frequency or time slices.

When you have located the desired cross-section, simply release the mouse button and switch on the desired 2D display.

A slice can then be edited in two dimensional display mode. You can, for example, dampen a particularly loud frequency or analyze the frequency content of a sound at a specific time point.

You can simply switch between the two 2D and 3D Modes as required. The only things that change are the editing functions at your disposal.

Various pieces of information are displayed above the toolbox.

- Name
This is the name of the spectrum (or the sample on which the spectrum is based).
- Block Start
The start point of the analyzed area within the sample.
- Block Length
The length of the analyzed area.

Reference

Toolbox

General



The toolbox is located on the lefthand side of the Frequency Domain Editor. It contains the display and editing functions for both the two and three dimensional levels.

Some of these icons only modify the way the spectrum is displayed, others control important functions.

3D Display Mode



Clicking this icon switches to a three dimensional spectrum display.

The three dimensional display can either be viewed as a 3D graphic or as a spectrogram.

You can choose between these two viewing submodes using the 3D Graphic and Spectrogram options in the View Menu.

In contrast to the high resolution 2D display modes, a 3D display is relatively coarse in order to maintain its simplicity.

If you subsequently change to 2D editing of frequency envelopes and frequency bands, you can return to 3D display by clicking on this icon or by calling the 3D Full View function in the Options Menu.

Reference

Change 3D Viewpoint



This icon enables you to adjust the position and size of the 3D grid system if the viewpoint selected by the program is unfavourable.

After you have clicked on the icon, move the mouse pointer into the spectrum display and click on the mouse once again.

The outline of the 3D display will begin to move.

If you move the mouse left or right, the display turns around its central point. This enables you, for example, to move the grid's zero point towards the back.

If you move the mouse up or down, the vertical position changes so that you can easily switch between a 'bird's eye' and 'ground level' perspective.

It is also possible to adjust the zoom factor, ie the size of the grid system.

Simply keep the right mouse button pressed and move the mouse up or down - the size of the display will change.

Release the mouse button when you are happy with the new size of the 3D grid.

You will discover, however, that it is not possible to move in so close to the grid that its boundaries 'disappear' off the edge of the display.

With very long samples it can take some time to recalculate a 3D display.

Reference

Play Sample



This function allows you to playback the section of the sample that has been transferred to the FDE.

If you have carried out any modifications to the spectrum data, the sample has to be resynthesized, a process that can take some time.

If you click on the icon with the right mouse button, the complete sample is played back.

Clicking on the icon with the [SHIFT] key pressed activates the Spectral Play function.

Spectral Play is a special function that allows you to monitor the frequency spectrums at a specific time point.

It resynthesizes the sound spectrum of the current frequency slice (how you select individual frequency slices is described in the section on "Defining 2D Time and Frequency Slices") and creates a loop around the resynthesized sound segment. This enables to hear a kind of 'frozen' image of the sample at a specific time point.

2D Frequency Mode

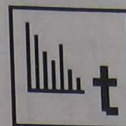


This icon enables you to select a two dimensional display in which the horizontal axis shows frequency and the vertical axis amplitude (volume) at a specific time point.

Reference

The time point involved can not only be selected in the 3D display - as described in "Defining 2D Time and Frequency Slices" above - it is also possible to use the PREV and NEXT icons in the toolbox. Clicking PREV selects the previous time point and NEXT the next time point.

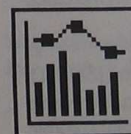
2D Time Mode



This function enables you to switch to a display in which the vertical axis shows the amplitude (volume) of a specific frequency and the horizontal axis represents time - from the beginning to the end of the analyzed block.

The frequency to be displayed can either be set in the 3D display or by using the PREV and NEXT icons in the toolbox. Clicking PREV moves one frequency band lower, NEXT one frequency higher.

Define Rubberband



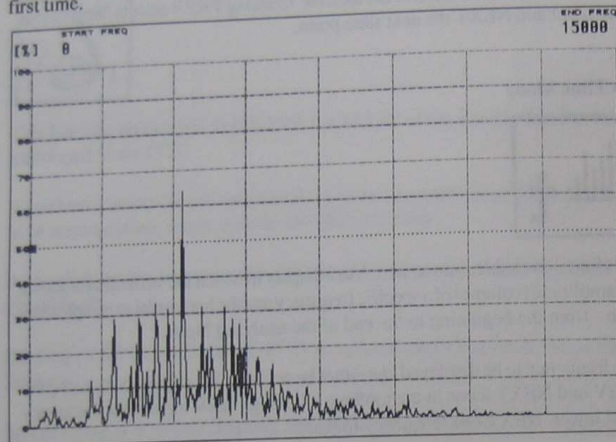
This function is used to define a rubberband.

A rubberband is a modification scale which you can define with the mouse inside the 2D display. A scale of this kind indicates the factor by which each amplitude value is to be increased or decreased. It thus represents a very handy tool which allows you to carry out wide-ranging modifications very quickly.

There is both a 2D frequency rubberband and a 2D time rubberband. They can be defined and used independently of one another. When used together as a 3D filter macro they represent a very powerful tool.

Reference

Rubberband Mode is activated when you click on the Rubberband Icon for the first time.

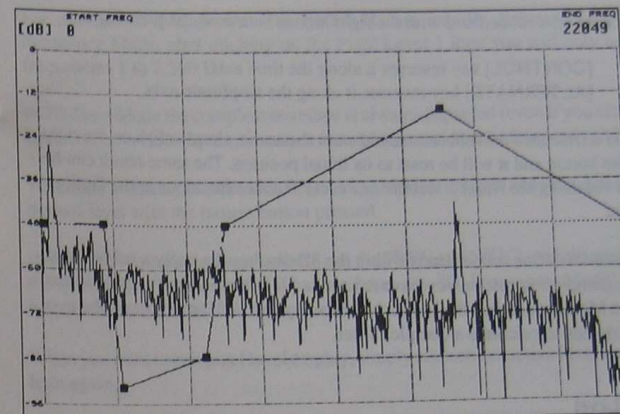


A horizontal line with a black box at each end will appear in the middle of the screen. This is the initial linear rubberband which has no effect on the sound data whatsoever - the amplitude values are left at their original level as it increases/decreases them by a factor of 1.

If you click on a position anywhere along this line while holding down [SHIFT], an additional box will appear. This box can also be clicked and then dragged up or down with the mouse button pressed. At this point you will probably be able to see how the rubberband got its name.

You can create as many boxes on this line as you choose and thus stretch it in any direction. To delete a box, click on it again while holding down [SHIFT].

Reference



The outcome of this process is that when you finally use the rubberband each amplitude is raised or lowered by the factor set by the rubberband at its position.

A maximum increase (when the rubber band is at the top edge of the display) represents a doubling of amplitude, a decrease to the lowest possible level (at the bottom edge of the screen) represents the deletion of the relevant value.

When you are happy with the rubberband, simply click on the Define Rubberband Icon again. A dialog box will then appear offering you three alternatives:

- Use this Rubberband?
Immediate application of the rubberband.
This corresponds to the functions Use F Rubberband and Use T Rubberband in the Options Menu.
- Edit
Continue editing rubberband.
- Store
Save the rubberband rather than use it immediately.
If this option is selected, the rubberband disappears from the screen, but it is saved in memory and can be used again by clicking on the Define Rubberband Icon or calling the Use Rubberband function in the Options Menu.

Reference

Clicking on a rubberband with the right mouse button whilst pressing the

- [CONTROL] key reverses it along the time axis.
- [ALTERNATE] key reverses it along the amplitude axis.

If you do not need the rubberband and want to erase it, simply click on the right mouse button and it will be reset to its initial position. The same result can be achieved using the Reset T Rubber or Reset F Rubber functions in the Options Menu.

You cannot define rubberbands within the 3D display. To apply a rubberband to the complete spectrum you have to use the 3D Filter Macro function in the Macro Menu rather than the Use Rubberband option. A 3D macro uses the two rubberbands to calculate a complex filter.

Pixel Level



The four Pixel Level icons enable you to further increase display resolution and carry out very fine adjustments to the amplitude values of individual frequencies.

Each of the four Pixel Level icons allow you to magnify the display of

- a quarter of the frequency envelope displayed in 2D Time Mode, or
- a quarter of the frequencies visible at a specific time point in 2D Frequency Mode.

From left to right, the four icons represent the pixel areas 1-128, 129-256, 257-384 and 385-512, ie the four quarters of the display previously visible on the screen.

Each point of the 2D display is magnified by a factor of four, and this makes it possible to carry out editing with the mouse very easily.

Reference

For example, if frequencies 1 to 10,000 Hz are initially displayed in 2D Frequency Mode, after clicking on the Pixel Level 1 Icon you will only see frequencies 1 to 2,500 Hz.

In 2D Time Mode the complete envelope is always displayed (even if you click on a Pixel Level Icon) as the time scale is much 'coarser' than the frequency scale.

To modify a value, point the mouse at the appropriate column and drag it to the desired level with the mouse button pressed.

In Pixel Level resolution you can also use the PREV and NEXT icons to access the previous or subsequent frequency band (in 2D Frequency Mode) or frequency envelope (in 2D Time Mode).

When you want to exit pixel level display, simply click on the active Pixel Level Icon again.

Define Envelope



This icon enables you to create a 'high definition' rubberband with 512 envelope points and use it in the following ways:

- In 2D Time Mode:
Modify the envelope of an individual frequency band. You can thus define a highly complex envelope with a single movement of the mouse.
- In 2D Frequency Mode:
Modify the amplitudes of all frequencies at a specific time point or along the complete time axis. If you wish to do the latter, you have to click on Filter in the dialog box that appears after you have defined the envelope. This function is the equivalent of a 512 band equalizer. It allows extremely precise erasure and/or amplification of spectrum components.

Reference

You activate this function by clicking on the Define Envelope Icon and then 'drawing' the desired envelope with the mouse button pressed.

When you click on the icon again, the time or frequency axis is modified by the new envelope.

Move Area



This icon is used to define and move areas.

You can only define and work with areas in 2D Frequency Mode, but they are projected onto all other frequency slices.

In other words, you can define a specific frequency range with the mouse and then move or copy it (depending on the area function selected in the Options Menu) over the whole time axis.

To define an area, click on the Move Area Icon and move the mouse to the position where the area should begin. Press the left mouse button, drag the pointer to the end of the area and release the mouse button.

Pressing the left mouse button again within the display then allows you to drag the 'outline' of the area to the desired position. When you release the mouse button, the area will be moved or copied (depending on the area function selected in the Options Menu).

Reference

Reverse Area



This function enables you to define and then invert an area.

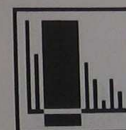
You can only define and work with areas in 2D Frequency Mode.

Click on the icon, move the mouse into the 2D frequency display and mark the desired area whilst keeping the left mouse button pressed.

As soon as you release the mouse button, the marked area will be reversed, ie low frequency bands are transposed upwards, and vice versa.

When this function is used on extremely high frequency signals, the high frequencies are transposed into the audible range.

Erase Area



This function enables you to erase an area, ie whole frequency ranges, from a signal.

You can only define and work with areas in 2D Frequency Mode.

Click on the icon, move the mouse into the 2D frequency display and mark the desired area whilst keeping the left mouse button pressed.

As soon as you release the mouse button, the marked area will be erased.

You can use this function to remove undesirable frequencies from a signal.

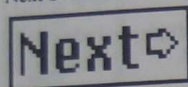
Reference

Previous Slice Icon



Clicking on this icon in 2D Frequency or 2D Time Mode switches to the preceding slice.

Next Slice Icon



Clicking on this icon in 2D Frequency or 2D Time Mode switches to the succeeding slice.

Frequency Slice Display

This is the frequency of the current frequency slice.

Frequency Slice

This is the number of the current frequency slice. In 2D Frequency Mode you can move to the preceding/following slice using the Previous Slice/Next Slice icons respectively.

Time Slice Display

This indicates the time point of the current time slice.

Time Slice

This is the number of the current time slice. In 2D Time Mode you can move to the preceding/following slice using the Previous Slice/Next Slice icons respectively.

Reference

Slice Magnitude

This is the volume of the selected time slice at the time point of the selected frequency slice, ie this display indicates the volume of a frequency at a specific time point.

Resynthesis



When you click on this icon, the spectrum is resynthesized into a sample and you are returned to the TDE.

Clearly, this is only necessary if you have modified the spectrum in the FDE.

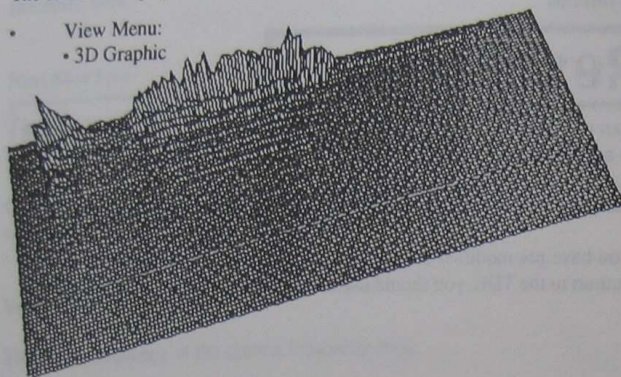
If you have not modified the spectrum or do not want to transfer a modified spectrum to the TDE, you should use the Quit Editor option in the File Menu.

Spectrum Display

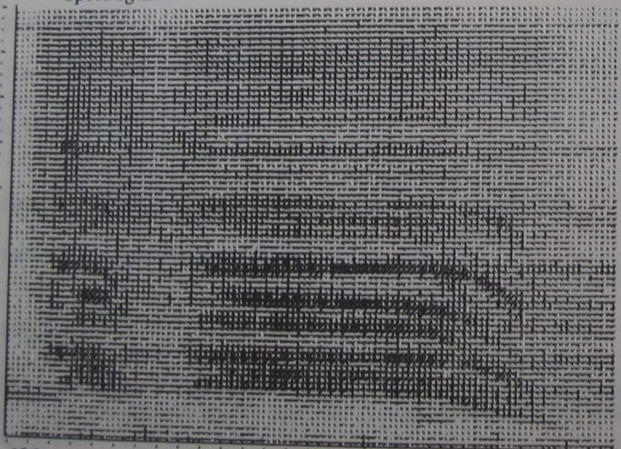
We have already described the different ways frequency spectrums can be displayed under the subheading "Representation" in the "Digital sound editing" section.

The different display modes can be called using the following functions in the

- View Menu:
 - 3D Graphic



- Spectrogram



- No Graphic Display

- Toolbox:

- 3D Display Mode
- 2D Frequency Mode
- 2D Time Mode

You will find detailed descriptions of these functions under the appropriate headings in this section of the manual.

Frequency Domain Editor menus and dialog boxes

Desk Menu

About

This function is identical to the function of the same name in the Desk Menu on the Mapping Page.

File Menu

Load Spectrum

This function loads a spectrum into memory from disk for further editing or resynthesis.

If there is already a spectrum loaded in memory, it is erased.

As the analysis of longer samples can take a considerable time, this function enables you to re-access a - previously analyzed and saved - spectrum quickly if you are unhappy with the changes you have made in the FDE.

Reference

Save Spectrum

This function enables you to save a spectrum to disk for later editing.

Spectrums are given the file name extension '.SPC'.

When saved, a spectrum takes up four times as much disk space as a sample of the same length. You should bear this in mind when planning available storage space.

Load F Rubber

This function loads a frequency rubberband.

Save F Rubber

This function saves the current frequency rubberband.

Load T Rubber

This function loads a time rubberband.

Save T Rubber

This function saves the current time rubberband.

Get Memory Info

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Reference

Mouse Speeder

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Screen Settings

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Preferences

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Quit Editor

This option allows you to leave the FDE and return to the Time Domain Editor without transferring the modified version of the spectrum. You thus return to the original sample.

If you want to resynthesize an edited spectrum and transfer it to the TDE, you have to use the Resynthesis Icon at the bottom of the toolbox.

View Menu

2D Lin. Scaling

This function enables you to switch to two dimensional linear scaling.

Linear scaling, as in the TDE, displays all amplitude values in a linear relationship to one another; this means that a value which is only half as high as the previous one is also only half as loud.

Reference

2D Log. Scaling

This function enables you to switch to two dimensional logarithmic scaling.

The human ear is scaled logarithmically. This means that a signal which is ten times louder than another is only perceived as twice as loud. Logarithmic scaling in decibels takes this into account. 100% is thus displayed as 0 dB, 50% as -6 dB, 1% as -40 dB, and 0% (which is really 'minus infinity' dB) is displayed as -96 dB (the maximum resolution in 16 bits).

Memorize Viewpoint

This function saves the current 3D display viewpoint.

Recall Viewpoint

This function reactivates a 3D display viewpoint that was previously saved using Memorize Viewpoint.

Recalculating the 3D display of very long samples can take some time.

3D Graphic

In this mode the frequency spectrum is displayed 'three-dimensionally'.

In this display mode the axes represent the following:

- horizontal axis: frequency,
- vertical axis: amplitude, and
- 'depth' axis: time.

The range covered by each of the three axes depends on the type and length of the analyzed block. The frequency and time axes indicate the maximum value (in Hz and ms respectively).

Reference

Spectrogram

In this mode the frequency spectrum is displayed as a spectrogram.

Amplitude is then represented by variation in density (blackness). A fading tone thus appears as an increasingly thinner line, the vibrato of a human voice as several parallel waves.

No Graphic Display

This function switches off the graphic display of the spectrum.

As rescaling the 3D display can take a considerable amount of time, it often makes sense to switch off the display when editing large spectrums.

Rescale Display

This function allows you to rescale the 3D display.

If you use a very strong filter on a spectrum, it sometimes becomes difficult to read the display as the amplitudes of the individual spectrum lines are too small. Calling Rescale Display causes AVALON to rescale the 3D display so that two and three dimensional screen output becomes legible.

Please note that only the display is rescaled and there is no change in the spectrum itself.

Options Menu

Reset F Rubber

This function resets the frequency rubberband to its initial value (ie to a linear scale).

Reset T Rubber

This function resets the time rubberband to its initial value (ie to a linear scale).

Reference

Copy F > T Rubber

This function copies the frequency onto the time rubberband.

Copy T > F Rubber

This function copies the time onto the frequency rubberband.

2D Use Rubberband

This function modifies

- the frequency slice currently visible on the screen by applying the appropriate frequency rubberband, ie its amplitude values are recalculated accordingly.

Modifying a frequency/amplitude slice with a rubberband means that all the frequencies present in the spectrum at the selected time point are given new amplitude values. You thus alter the spectral content at a specific time point, ie apply an exactly timed filter function.

- the time slice currently visible on the screen by applying the appropriate time rubberband, ie its amplitude values are recalculated accordingly.

Modifying a time/amplitude slice with a rubberband means that the selected frequency receives new amplitude values along its complete time axis. You thus give this one frequency a new envelope.

The program will inform you if distortion ('clipping') occurs when a rubberband is used.

Peak Picker

This option gives you exact information on the frequency, amplitude and time of a specific point on the display.

Reference

Fourier analysis splits a sound into a limited number of frequency bands (namely 512). This means that specific frequencies 'slip through' the grid, ie if the time/frequency scale presents the values 430, 440, 450 Hz etc, you cannot be certain whether a specific frequency you want to analyze or edit is actually 440 or 444 Hz.

The Peak Picker supplies this information. It examines the available values and provides exact information on the actual frequency of a signal.

When you activate this menu function in 2D Time or 2D Frequency Mode, a display will appear with the following information:

- Frequency
- Time
- Magnitude

You can switch this mode off by clicking on the menu option again or changing to 3D Mode.

Move Area

If this mode is active, an area is moved to the new position when you call the Move Area function. The spectrum lines in the destination area are then erased.

Copy Area

If this mode is active, an area is copied to the new position when you call the Move Area function. The spectrum lines in the destination area are then erased.

Add (+) Area

If this mode is active, using the Move Area function causes the area to be added to the existing spectrum lines from the selected position onwards.

Subtract (-) Area

If this mode is active, using the Move Area function causes the area to be subtracted from the existing spectrum lines from the selected position onwards.

Macros Menu

General

The macro level (the third level in addition to the pixel and micro levels which allow you to work in the 2D areas of the FDE) represents your most powerful sound editing tool. It enables you to modify the overall character of a sound throughout its complete frequency range without having to painstakingly work your way through one micro level after the other.

As the whole frequency spectrum cannot be modified directly with the mouse - such a process would also not make much sense - AVALON provides you with a complete set of macros, ie algorithms which carry out this work for you.

Macros enable you to do astounding things, but in the final analysis they only really vary a given frequency spectrum. We would thus advise you to experiment on various sounds with different macros and macro combinations to find out which applications they are best suited for.

Spectral Animation

This macro varies a spectrum whilst also taking account of the overall structure of the specified frequency range.

Its parameters are:

- Minimum Frequency
- Maximum Frequency
- Depth
The intensity of the effect.

Time Plexer

This macro is similar to the Delay Block function in the Time Domain Editor. It creates echoes of the original sound.

It is able to produce much more subtle effects than the TDE as it can be restricted to a specific frequency range.

Its parameters are:

- Minimum Frequency
- Maximum Frequency
- Mode
Here you can choose the direction ('To Block Start'/'To Block End')
- Depth
The intensity of the effect.

Spectral Dynamic

This macro enables you to compress or expand the dynamic range of a specific frequency range.

Its parameters are:

- Minimum Frequency
- Maximum Frequency
- Mode
Here you can choose between an increase ('Expand') and decrease ('Compress') in dynamic range.
- Depth
The intensity of the effect.

Reference

Enhancer

This macro adds a 'subspectrum' to a given spectrum. This results in a richer, more resonant sound.

The only parameter is

- Depth
The intensity of the effect.

3D Filter

This macro combines the two rubberbands (time and frequency) into a complex filter whose properties can vary along the time axis.

The frequency rubberband defines the basic features of the filter, eg a slight amplification of all frequencies above 10,000 Hz and the deletion of all frequencies below 50 Hz. You can define this rubberband in 2D Frequency Mode.

The time rubberband determines how the frequency rubberband affects the spectrum at each point along the time axis. If the time rubberband is set in the middle of the screen, the effect of the frequency rubberband is not changed. If it is raised to its maximum value, the effect is doubled, and if it is reduced to 0, the frequency rubberband has no effect. You can define this rubberband in 2D Time Mode.

A great variety of combinations are possible here, thus enabling you to carry out extremely accurate filtering.

If you have only defined a

- time rubberband, all frequencies are raised or lowered in a linear fashion, ie 3D Filter gives the spectrum a new overall envelope.
- frequency rubberband, all time slices are modified in the same way, ie the result is the same as applying the rubberband to every single time slice in 2D Frequency Mode.

Reference

Harmonic Edit

This macro makes it possible to manipulate the sample on the basis of its harmonic structure. First, the basic pitch and the first upper harmonic must be defined - manually or automatically. In other words, it is possible to amplify or dampen the harmonics of the basic frequency or all other frequencies.

The parameters are:

- Main Frequency
The basic pitch.
- 2nd Upper
The first harmonic.
- Mode
Here you can select whether frequencies should be amplified or dampened.
- Modify: Harmonics/Non Harmonics
Here you can select whether harmonic or non-harmonic frequencies should be modified.
- Basic Width 1/3/5 Harmonics
As you probably already know, fast Fourier analysis only provides information about a limited number of frequency bands (that have a specific width). The actual basic frequency of a sound can lie somewhere between two of the given frequencies within this grid. The Basic Width parameter enables you to select the number of frequency bands which are to be raised or lowered by the Harmonic Edit macro.
- Depth
The intensity of the effect.

Pitch Shifter

This macro transposes a signal up or down by a specific value without influencing its length, as would be the case if you simply changed the sample playback rate. All frequencies are shifted up or down in a linear fashion.

Reference

The only parameter is

- **Shift Value**
The transposition factor can be set in 10 cent steps (ie in tenth semitones) within a range of 2 octaves.

When a spectrum is shifted in this way, all its frequencies are transposed, including those that would normally remain constant while an instrument is being played, singing or speaking takes place etc. Thus, a transposed sound can sound unnatural despite the fact that length remains unchanged. This effect can be reduced somewhat by subsequent filtering.

Frequency Mixer

This macro generates additional harmonics.

Each amplitude value of a spectrum line is added to the neighbouring spectrum line with a specific intensity factor.

You can also limit the effect of this macro to specific frequency ranges, eg to give a high frequency range more 'sparkle'.

The parameters are:

- **Minimum Frequency**
- **Maximum Frequency**
- **Mode**
Here you can select whether frequencies should be added to the previous ('Down') or the next ('Up') spectrum line.
- **Depth**
The intensity of the effect.

Reference

Spectral Shaper

This macro modifies the amplitudes of a specified frequency range on the basis of a given 'control curve'.

When a frequency band passes through such a curve, its amplitudes are modified in a specific way. In this process, the amplitude of the signal determines whether it passes through the whole control curve or only parts of it. If the amplitude of the input signal becomes smaller, it only passes parts of the control curve.

Either the frequency or the time rubberband can be used as the control curve.

The parameters are:

- **Minimum Frequency**
- **Maximum Frequency**
- **Rubberband Display**
This display shows the rubberband used as the control curve. You will find a description of how to define a rubberband in the "Define Rubberband" section above.

The control curve has the following effect:

The x-axis displays the amplitude of the input signal. This means, the greater the amplitude of a frequency band at a specific time point, the further right it appears on the x-axis.

The y-axis displays the factor by which the amplitude is modified. Here, the same rules apply as when rubberbands are used normally:

- The middle line represents the factor 1 (no modification).
- The maximum value corresponds to the factor 2 (doubling of amplitude). Please note that distortion can easily occur here.
- The minimum value corresponds to the factor 0 (no output signal).

The simplest conceivable application for this macro would be as a compressor. A rubberband could fade from factor 1 to zero at a specific point. This would result in all amplitudes above this level being dampened. Of course, completely different applications are also possible.

Reference

- Use:
This parameter determines which rubberband (F Rubberband or T Rubberband) should be used as the control curve.

Threshold

This macro enables you to erase the frequencies below a specified amplitude. It can thus be used as a 'denoiser' to separate unwanted high frequency background noise from a useful signal.

The parameters are:

- Threshold
The amplitude value below which frequencies should be erased.
- Minimum Frequency
The lowest frequency from which the macro should take effect (as noise is generally only noticeable at higher frequency ranges).

Influence

This macro averages out neighbouring spectrum lines, ie the value of a line is averaged with the value of the previous and next line. This results in a more uniform structure.

The parameters are:

- Minimum Frequency
- Maximum Frequency
- Mode
This parameter determines the form of influence. In Increase Mode spectrum lines are raised and in Decrease Mode they are lowered.
- Depth
The intensity of the effect.

Reference

Peak Limiter

This macro represents a simple method of dampening extreme peaks (which might lead to distortion) without completely erasing them.

- Peaks over
This parameter determines the amplitude value above which peaks should be modified/eliminated.
- Reduce by
This parameter determines the percentage (of original volume) to which peaks should be reduced.

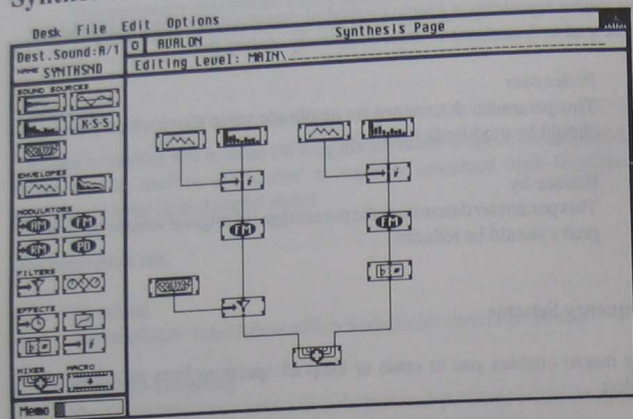
Frequency Selector

This macro enables you to erase or keep all spectrum lines except for the loudest.

The parameters are:

- Keep/Erase
You can either 'keep' or 'erase' the loudest spectrum lines.
- the loudest Frequencies
The number of spectrum lines to be kept/erased.

Synthesis Page



The Synthesis Page provides numerous sound editing functions which you can combine to simulate popular types of synthesizer or realise your own synthesis concepts.

When you have finished processing a sound on the Synthesis Page it can be transferred back to the Mapping Page or Time Domain Editor for further editing.

The layout of the Synthesis Page

At first glance the Synthesis Page closely resembles the Time and Frequency Domain Editors. However, it works on a completely different basis.

Whilst the TDE and FDE enable you to display and edit an existing sample, the Synthesis Page is used to create a kind of synthesizer 'blueprint' - for a synthesizer that produces specific sounds depending on the component modules used and the way they are linked together.

All the modules required to construct such a model are located in the Part Box on the lefthand side of the screen. They represent the different synthesis techniques described above in the "Synthesis" section. You can take the modules out of the Part Box with the mouse and arrange them within the Connection Display, the 'circuit board' on which the 'synthesizer' is put together.

You are then able to link the individual modules and match them together by adjusting various parameters.

A group of linked modules can be saved as a 'connection' and subsequently edited. Theoretically a connection can also consist of a single module.

An interesting aspect of the Synthesis Page is the fact that each module within the display is fully functional - whether it is part of a connection or not. In other words, you can monitor the sound created at any location within your synthesizer model and thus carefully follow how a complex sound is created out of simple components.

These sound components can then be transferred to the Mapping Page as a finished sample or copied to the Time Domain Editor for further editing.

The possibilities offered by the Synthesis Page are greatly increased by the fact that you can combine almost any number of modules into macros that can then in turn be built into a connection.

A special utility is used to link, separate, monitor or copy connections - a facility that you may already know from the CUBASE sequencer program. It involves a special toolbox 'installed' directly at the pointer. To avoid confusion, we will refer to it as the Mouse Toolbox.

You can call the Mouse Toolbox at any time by pressing the right mouse button and holding it down. It contains all the functions needed to create and monitor a synthesizer model.

Destination Sample

Global Information

NAME OF CONNECTION: SYNTHSND

POSITION OF FINAL SAMPLE:

Bank Identifier: ☐ A ☐ B ☐ C ☐ D

Sample Number: ☐ 1 ☐ 2 ☐ 3 ☐ 4 ☐ 5 ☐ 6 ☐ 7 ☐ 8

Sampling Rate: ☐ 44100 ☐ Hz

Number of Modules:

Number of Macros:

Number of Lines:

Free Memory: Samples

OK

Above the Part Box you will find a display that indicates the memory location to which a new sound is saved when you use the Copy To MP or Copy & Call TDE functions in the Mouse Toolbox.

Any sample data stored at the selected sample location will be erased by these functions.

If you called the Synthesis Page by

- double-clicking the Synthesis Page Icon, the default destination sample is memory location 1 in the current bank.
- dragging the Synthesis Page Icon to a specific Sample Icon, the specified sample location then becomes the default destination for the new sound.

However, this setting can also be subsequently altered.

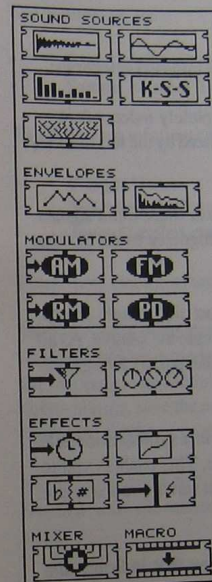
Simply click on the name of the destination sample and a dialog box will appear containing the Synthesis Page's global parameters. The dialog box allows you to change the destination bank and sample location. Its other settings are described under "Info" in the "Edit Menu" section below.

Name

This is the name that the synthesized sound is given when copied to the Mapping Page.

Part Box

General



Special rules apply to the various modules of the Part Box.

Double-clicking a module calls the dialog box containing all the parameters for the relevant synthesis or modulation technique. These parameters can be adjusted in the usual way.

If the module is still located in the Part Box, any adjustments made modify the basic settings for all future applications of the module.

If, however, you modify the settings of a module that is already 'installed' in the Connection Display, only the parameters of that module are changed.

You can erase a module which has not been linked into a connection by simply dragging it out of the Connection Display and then releasing the mouse button.

All modules have the following two parameters:

- Name On being inserted in the Connection Display a module is given a name made up of the abbreviation for the synthesis technique involved and a specific number (eg 'FM_01'). This name can be changed at any time.

Reference

- **Optimize** Each module creates or modifies a sample. The Optimize button allows you to cause the overall volume of the sample to be automatically raised to the maximum level. This corresponds to the Optimize function in the TDE.

As these parameters are common to all modules we will not refer to them below in the descriptions of individual modules.

Generators and Modulators

Basically, the modules in the Part Box can be divided into two categories:

- **Generators** These are modules that create completely independent signals (sounds, envelopes). They can be recognized by the fact that they do not have an input at the top.
- **Modulators** These are modules that require one or more input signals which are then modulated. Modulators can be recognized by the fact that they have one or more inputs at the top.

All generators have a setting that allows you to adjust the sample length. Obviously the longer the length, the longer it takes to create the sample. As all such modules provide this facility we will not refer to this parameter again in the descriptions of the individual modules below.

Clicking on the OK button with the right mouse button in a synthesis module dialog box plays back the sound created by that module. This enables you to quickly compare the effect of different settings without having to leave the dialog box.

All modulator modules provide a Sample Length display that indicates the length of the synthesized sound.

In the next sections you will find the following information on each module:

- the form of synthesis or modulation on which it is based, and
- the settings that influence it beyond those already mentioned above.

Reference

Mapping Page Sample



This module enables you to specify one of the 32 samples on the Mapping Page as a generator for synthesis.

The parameters are

- **Bank Identifier** The bank where the sample is located.
- **Sample Number** The number of the sample location.

The following information is also displayed:

- **Sample Name**
- **Input Sample Rate**
- **Input Sample Length**

Please note that loops are ignored.

The interesting thing about this module is the fact that it enables you to create recursive synthesis models. In other words, you can use Sample 1 in Bank A as the Mapping Page Module, eg send the signal through a ring modulator and two delay circuits, save the sound created at sample location 1 in Bank A and then repeat the synthesis process over again as often as required.

Please note that when you save a connection that contains an MP Sample Module, only the number of the sample location is stored and not the sample itself.

Reference

Wave Generator



This module is described in the "Synthesis" section above.

Four preset wavetables are available for wavetable synthesis:

- Sine
- Rectangle
- Triangle
- Mute

Additional wavetables can also be loaded using the Disk Icon.

Each wavetable is represented by a small icon. Double-clicking one of these icons enables you to activate an enlarged display of the waveform.

The preset wavetables can be arranged in the Wavetable Display that takes up most of the dialog box. It is made up of two parts:

- **Wave Overview** This display shows the complete time axis for the wave transformation. Its length is calculated on the basis of the selected sampling rate and sample length. Each wavetable appears here as a vertical column. Similar to Zoom Mode in the TDE, it is possible to enlarge specific sections of the Overview using the mouse. Simply keep the mouse button pressed and mark the area you want enlarged. When you release the mouse button, the relevant section of the Overview will be displayed in the Wave Display below. Within the Overview itself the enlarged area is marked by shading.
- **Wave Display** This is where you arrange the individual wavetables. Simply drag the desired wavetable icon to its destination position in the Wave Display. Each wave is displayed here as a small 'flag' - with the 'flag pole' marking its exact position and the 'flag' itself showing a reduced representation of the wave.

Reference

When you release the mouse button, another dialog box will appear in which you have to enter two values:

- **Form Factor** For standard rectangle and triangle waveforms this parameter enables you to modify the frequency content as follows:
 - **Rectangle** You can change the relationship between the upper and lower pulse width. Thus, using several rectangle waves with different pulse widths in a long wavetable allows you to simulate the pulse modulation of analog synthesizers.
 - **Triangle** You can vary the frequency content between triangle (only even harmonics) and sawtooth characteristics (all harmonics).
- **Frequency** This value represents the factor by which wavetables can be stretched or compressed. For example, the factor
 - 0.5 results in the wavetable only containing a half wave period.
 - 2 results in the wavetable containing two wave periods.

This dialog box can be called at any time by holding down the [ALTERNATE] key and clicking on a wavetable.

Once they have been inserted, wavetables can be repositioned in the Wave Display as desired.

To erase a wavetable, simply click it whilst holding down the [SHIFT] key.

The frequency content of the new sound is determined by the order of the different wavetables.

Fourier Generator



The Fourier Generator creates a sine wave with up to 7 harmonics ('1st Upper - 7th Upper').

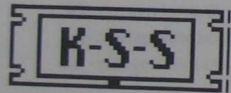
The parameters for each frequency are:

- Amplitude In per cent.
- Frequency In relation to the basic frequency, whose frequency is represented by the value 1.
You can adjust the figures in front of and behind the decimal point independently of one another using the mouse.
- Phase Phase relationship.

However, these three parameters cannot be modulated along the time axis (eg by envelopes). This results in a purely static sound. The Fourier Generator is thus generally only used to create a simple base sound that is then modified by other modules (eg envelope-controlled filters and amplifiers).

Clearly it is also possible to create very much more complex generators by connecting Fourier Generators in parallel.

Karplus-Strong Synthesis



This generator is described in the "Synthesis" section above.

The only adjustable parameter is:

- Frequency Length of the delay circuit.

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Fractal Synthesis



This generator is based on the principles of fractal synthesis described in detail in the "Synthesis" section above. We would advise you to read this section of the manual to gain an understanding of how the module works before trying it out.

The largest part of the dialog box is taken up by the display representing the fractal.

Within this display the horizontal axis represents time and the vertical axis frequency.

There are two display modes:

- In Tree Mode you are able to position the mother and daughter circles. This is how you lay down the 'guidelines' for the creation of the fractal itself.
This is done in the usual way: Point the mouse cursor at one of the points that you want to adjust and then drag it to its new position with the mouse button pressed.

You can change the following:

- The position of mother and daughter circles Simply drag the centre of the circle to the desired position.
- Base vector and size of the mother circle Point at the mother circle and set the desired angle and size by moving the mouse.
- In Draw Fractal Mode, the fractal itself is calculated and displayed. This mode is activated by clicking on the Draw Mode button. The fractal is influenced by the settings made on the lefthand side of the dialog box.

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Reference

You cannot modify the fractal itself in Draw Mode. However, if you click on the mouse within the display, 'cross-wires' will appear that you can use to determine the frequency, time, and volume of individual events. When you click the mouse at a specific position, these three pieces of information will appear next to the display.

It is also possible to mask off part of the fractal by pulling down a kind of 'blind'. The covered part of the fractal is then not converted into sound data.

In order to do this, move the pointer to the top edge of the fractal and drag down the 'blind' as far as it is needed. When you release the mouse button the covered section of the fractal is shaded grey. If you want to 'pull the blind up' again, simply click on the top edge of the display.

On the lefthand side of the dialog box you will find the basic settings affecting the way a fractal is converted into an audible sound.

The parameters are

- **Daughters**
The number of daughter circles on which the fractal is based. At least one daughter must be present; a maximum of three can be used.
- **Attack Sound**
events can be formed with the aid of a simple envelope (a so-called ramp generator). The attack time can be set within a range of 1,000 sample values.
- **T-Length**
The length of the sound events into which the fractal is converted. This value applies to all generated events.
- **Depth**
The number of generations to be calculated.
Please note that the processing time involved increases exponentially with the number of generations. With more daughter circles the time required is also multiplied again.
It is thus often advisable to first process a 'rough sketch' of a fractal using a low Depth value and then increase it later.

Reference

When you have finished processing a sound on the Synthesis Page it can be transferred back to the Mapping Page or Time Domain Editor for further editing.

Envelope Generator



This generator creates an envelope that can have a large number of individual steps.

The envelope is created graphically using the mouse. This is done within the Envelope Display, which takes up most of the module dialog box. The display's horizontal axis is scaled in ms, the vertical axis in per cent. It consists of two parts:

- **Envelope Overview**
This window always displays the complete time axis of the envelope. It is determined by the settings for sampling rate and sample length. You can enlarge specific areas of the Overview with the mouse. Simply mark the area of the envelope that you wish to enlarge whilst keeping the mouse button pressed. When you release the mouse button, the appropriate part of the envelope is displayed in the Envelope Display underneath. Within the Overview itself the enlarged section is then displayed as a shaded area.
- **Envelope Display**
Here you can position and move envelope points using the mouse.
 - To define a new point, simply click the mouse on the envelope with the [SHIFT] key pressed. A small black box will appear.
 - To move an existing point, simply drag it to the desired position with the mouse button pressed.

Reference

- To erase an existing point, click the point again with the [SHIFT] key pressed.

Clicking the mouse in the Envelope Display with the

- [CONTROL] key pressed, reverses the envelope along its vertical axis.
- [ALTERNATE] key pressed, reverses the envelope along its horizontal axis.

You can also reset the whole envelope (back to its initial form) by clicking inside the display with the right mouse button.

Envelope Follower



This module enables you to determine the overall envelope of a (complex) signal.

The input signal is sent through a lowpass filter, whose gradient is freely definable.

The filter attempts to join the individual amplitude peaks of the signal together to form an envelope curve that characterizes the signal.

The smoothness of the curve depends on the filter density selected. However, higher filter densities do not only lead to the amplitude peaks being joined together with more precision, it also means they are increasingly altered.

A compromise thus has to be found between

- a) too low a filter density (and resultant 'residual ripple' within the envelope, a phenomenon that could also be referred to as 'fraying'), and
- b) too high a filter density (that leads to the filtering out of important envelope characteristics).

Reference

The only parameter is

- Density Filter gradient.

Amplitude Modulation



This modulator is described in the "Synthesis" section above. It provides you with a sine generator whose amplitude can be modulated by an input signal.

The only parameter is

- Mix Level
The balance between the original input signal and the modulated signal.

Frequency Modulation



This generator is described in the "Synthesis" section above. It provides you with a sine generator whose frequency can be modulated by an input signal.

The only parameter is

- Mix Level
The balance between the input signal and the signal of the modulated sine generator.

Reference

Ring Modulator



This modulator is described in the "Synthesis" section above. It multiplies two input signals.

The only parameter is

- Mix Level
The balance between the two original input signals and the signal created within the ring modulator itself.

Phase Distortion



This modulator is described in the "Synthesis" section above. It provides a sine generator whose frequency can be modulated by any input signal.

This module has no adjustable parameters.

Digital Controlled Filter



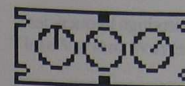
This modulator is a lowpass filter whose cutoff frequency can be controlled by another module. For example, envelope generators or envelope followers are well suited as control signals for the filter frequency.

Reference

The parameters are

- Cutoff Frequency
- Mix Level
The balance between the original signal and the filtered signal.

Parametric Equalizer



This modulator largely corresponds to the filter functions in the TDE.

It provides the basic filter types: Lowpass, Highpass, Bandpass and Notch.

You can also choose between Cutoff and Shelving Modes.

A Cutoff filter cuts out all frequencies beyond the cutoff frequency.

Shelving Mode makes it possible raise or lower the desired frequency bands more smoothly.

The parameters are

- Cutoff Frequency
- Bandwidth The range of the peak/notch filter.
- EQ Type The following filter types are available:
 - Lowpass Damp
This type of lowpass filter has already been described in the section on the TDE.
 - Lowpass Boost
This type of lowpass filter boosts low frequencies (rather than dampening high ones).

Reference

- **Lowpass Shelving**
This filter is like Lowpass Damp, but the frequencies above the cutoff frequency are not cut off as abruptly as with a normal lowpass filter. They are faded out over a specific frequency range.
- **Highpass Damp** See the description of the Highpass Filter in the section on the TDE.
- **Highpass Boost** This type of highpass filter boosts high frequencies (rather than dampening low ones).
- **Highpass Shelving** Like Highpass Damp, but less abrupt.
- **Peak** See the description of the Peak Filter in the section on the TDE.
- **Notch** See the description of the Notch Filter in the section on the TDE.

Delay



This modulator is a delay circuit - similar to the delay function of a digital sound processor.

However, as the calculations do not take place in realtime here (as there is no need for continuous input processing), the module itself is rather simple.

Nevertheless, it is still possible to create the same effects as when using a 'real' delay processor.

Reference

The parameters are

- **Delay**
- **Mix Level**
The balance between the original signal and the delayed signal.
- **Replication No.**
The number of echoes.

Time Variant Waveshaping



This modulator is described in the "Synthesis" section above.

A variety of waveshapes are available for use in time variant waveshaping. (You will find them in the 'SHAPES' folder on the AVALON disk II) They can be loaded by clicking on the Disk Icon.

In addition to these, you can also use the linear waveshape that does not modify the input signal. It can be used in combination with other waveshapes to add vitality to the distortion process.

You can arrange the waveshapes in the Waveshape Display. It takes up most of the dialog box and consists of two parts:

- **Waveshape Overview**
This window always displays the complete time axis for the waveshaping process. It is determined by the settings for sampling rate and sample length. Each inserted waveshape appears as a vertical column. Similar to Zoom Mode in the TDE, it is possible to enlarge a specific section of the Overview with the mouse. Simply keep the mouse button pressed and mark the area you want enlarged. When you release the mouse button, the relevant section of the waveshape table will be displayed in the Waveshape Display underneath. The enlarged section is then displayed as a shaded area in the Overview.

Reference

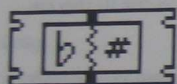
- **Waveshape Display**
You can arrange the individual waveshapes in any order in this section of the dialog box. Simply click on the Disk Icon to load a waveshape and then move it to the desired position with the mouse. Each waveshape is displayed here as a small 'flag' - with the 'flag pole' marking its exact position and the 'flag' itself showing a reduced representation of the wave. Double-clicking one of these icons activates an enlarged display of the corresponding waveshape.

Once they have been inserted, waveshapes can be repositioned in the Waveshape Display as desired.

To erase a waveshape, simply click it on with the [SHIFT] key pressed.

The chronological order of the different waveshapes determines how the frequency content of the input sound is modified.

Pitch Shifter



This modulator is used to transpose a sound. It corresponds to the Resampling function in the TDE.

The parameters are

- **Factor**
The transposition factor (in 10 cent steps, ie 120/octaves).
- **Mix Level**
The balance between the original signal and the transposed signal.

Reference

Digital Controlled Amplifier



This modulator is used to control the amplitude of an input signal by a second signal (usually an envelope generator). Its function thus corresponds to the voltage controlled amplifier (VCA) of an analog synthesizer.

The only adjustable parameter is

- **Mix Level**
The balance between the original signal and the modulated signal.

The Digital Controlled Amplifier module can also be used without an additional modulator and thus serve as a simple amplifier.

Mixer



This modulator can be used to mix the volume of up to ten input signals. It thus corresponds to a normal mixing desk.

The only adjustable parameter for each input signal is

- **Volume**

The following is also displayed for each input signal:

- **Input Module Name**

Please note that all sound components must be available at the same time in order to process a mix sound.

Macro

This is not really a conventional module at all. If we regard the Connection Display as the 'blueprint' or 'circuit board' for our 'synthesizer', then a macro is a kind of 'integrated circuit'. A macro can be installed in a connection like any other module, and a macro itself contains modules (which can in turn also be macros).

Each macro has ten inputs and ten outputs. When you open a macro (with a double-click or via the Open function), you will see the inputs at the top of the display and the outputs at the bottom.

You can work inside the macro in the same way as in the Connection Display itself. When you are satisfied with the macro's internal structure, you simply close it again and return to the next higher level.

You can insert one or more macros within an open macro and repeat this process as often as you wish.

This means that you can create complex connections in the form of a single 'integrated' macro and then very easily link these into other connections when required.

A maximum of 250 macros can be used simultaneously within the Connection Display, and it doesn't matter whether these are at the same level or nested within one another.

Macros do not have their own parameters. However, when you save a macro, the parameters of its constituent modules are saved with it.

Memory Display

This display provides information on memory usage. The grey part of the display indicates how much memory is already occupied.

Connection Display**General**

The Connection Display has a hidden grid (similar to the Invisible Grid on the Mapping Page) that automatically aligns the modules when they are inserted.

There are a few rules that should be followed to make work on the SP easier.

When you create a new synthesis model, always leave enough space between the individual modules.

The modules should always be arranged from top to bottom. Of course, if you wish, you can also arrange the modules from left to right or in any other pattern, but the amount of 'cabling' required is then greater - not to mention the lack of clarity involved.

Arrange all the modules belonging to the actual signal flow vertically and all additional modules on the left.

Modules or groups of modules can also be subsequently repositioned. However, when you do this, selected and unselected modules may not be linked together, ie only groups of activated modules can be moved.

Synthesizer simulation

Please bear in mind that you are not constructing a real synthesizer in the Connection Display, but designing a synthesizer that plays a specific sound at a specific pitch. If you want to create a complex synthesizer sound, you have to make sure that it is playable (ie its characteristics change according to velocity, controller use and keyboard range) - just as you would when preparing 'regular' samples.

Reference

The most important rules here are:

- Compensate for the 'Mickey Mouse effect' in the upper keyboard range by setting loops. Generally it is very easy to find loops for synthesized sounds.
- Modulation effects such as vibrato and tremolo should not be part of the synthesized sound, but should be realized at the sampler (by means of appropriate playback parameters). However, if such effects are an indispensable component of the sound, you should then create multisamples to prevent unnatural sound changes over the keyboard range.
- Generally, when processing your synthesized sounds at the sampler, you should also use playback parameters and settings that have already proved useful with 'natural' sound material.

Level Display

The nesting level is indicated at the top edge of the Connection Display.

When you first call the editor, this will always be the MAIN LEVEL.

If you then use macro modules and open them, the Level Display indicates the current path. Thus, if you have created a macro with the name 'MULTI_FM' and this contains a further macro called 'AM_4', when you open the latter macro, the Level Display will indicate the level 'MULTI_FMMAM_4'.

Mouse Toolbox

If the pointer is in the Connection Display and you hold down the right mouse button, the Mouse Toolbox will appear. It contains all the functions required to link, separate, playback and copy connections.

To select one of these tools, call the Mouse Toolbox by pressing the right mouse button and holding it down. Move the mouse to the desired tool and release the mouse button.

Reference

The top three tools can be used at any time, but the bottom three tools can only be used when the mouse pointer is located at a module.

When you no longer need a tool and wish to return to the standard pointer, simply click on the right mouse button.

The following sections describe the individual tools and their functions.

Standard Pointer

The standard pointer is used to select, open and move modules.

Clicking on a module selects it. You can then apply the functions of the Edit Menu (Open, Info, Copy, Cut, Erase and Paste) to the module.

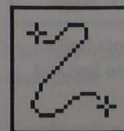
If you wish to select several modules at the same time, click on them with the [SHIFT] key pressed.

Selected modules can be moved inside the Connection Display - unless they are linked to other modules that are not selected.

Point at one of the selected modules, press the mouse button, drag the module to the desired position and release the mouse button.

Double-clicking a module calls the dialog box containing its parameters. If it is a macro module, you are switched to the level of that particular macro.

Link Tool



The Link Tool is used to connect modules together.

Reference

The following rules apply when linking modules:

- Links must always be made in the order output > input.
- Only one 'connecting cable' can lead from an output. If you want to direct a signal to more than one inputs, you have to add 'junction boxes' beneath the output. To do this, simply click on the line and direct the new cable that appears to the desired input.
- An input can always only be connected to one output.
- Recursive links (eg mixer output > mixer input) are not allowed.
- Links can cross, but a link must never cross a module.

A link is created in the following way:

- a) First click on the module whose output you want to link to another module.
- b) Then move the pointer towards the other module.

The pointer will drag a line behind it.
If this line is

- dotted, a link is possible.
- broken, one of the above rules has been violated.

If you change your mind and no longer want to make a link, simply press the right mouse button.

- c) Click on the input of the destination module. If you have not broken any of the above rules, the 'cable' is displayed as an unbroken line and the link has been made.

You can also lead connecting lines around other modules. To do this, simply click the mouse at a position where there is no module. An unbroken line appears up to this point; in effect, you have fixed its position within the display.

Reference

This technique allows you to bend the 'cable' to link individual modules. You can bend the links up to 20 times. Please note, however, that the Connection Display can become rather confusing as a result.

Erase Tool



The Erase Tool is used to delete modules and links between modules.

The following rules apply:

- If you erase a link, the linked modules are not erased.
- If you erase a module, all links with this module are also erased.
- If you erase a macro, all the modules it contains are also erased.

Simply click the Erase Tool on the link or module that you want to erase.

Play Tool



The Play Tool allows you to playback the sound of a module or a complete connection.

Simply move the pointer to the appropriate module and hold down the right mouse button.

The Mouse Toolbox will appear. The lower three tools are displayed in black, ie they can now be used.

Reference

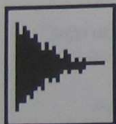
Move the mouse until the Play Tool is highlighted (displayed in inverse form) and release the right mouse button. The sound will then be synthesized in memory and played back.

It is also possible to monitor the synthesized sound from within an opened dialog box by clicking on the OK button with the right mouse button. This means that you can quickly compare the effect of different settings without having to leave the dialog box.

Please note that when you want a sound to be played back, all the modules located in front of it have to be synthesized in memory before this can be done. This can take a considerable time for complex connections.

Once synthesized, however, the sound components remain in memory (if there is enough room). Thus, when AVALON has 'worked out' a large connection once, you can then playback individual sound components later without having to wait.

Copy & Call TDE Tool



Activating this tool copies the sound of one module or one connection to the Mapping Page and immediately calls the TDE where the sound can then be edited further.

Move the pointer to the appropriate module and hold down the right mouse button.

The Mouse Toolbox will appear. The lower three tools are displayed in black, ie they can now be used.

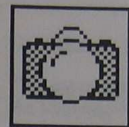
Move the mouse until the Copy & Call TDE Tool is highlighted (displayed in inverse form) and release the right mouse button.

Reference

The sound is copied to the TDE where it can be edited - just like any other sample.

The synthesized sound is stored at the memory location defined as the destination sample (ie the default location set when the SP was called or one entered subsequently). Please note that any sound stored at this location will be erased the next time you call the Copy To MP or Copy & Call TDE tools. You should thus either copy the synthesized sound to another memory location or change the Destination Sample parameter when you next use the SP.

Copy To MP Tool



This tool copies the sound of one module or connection to the Mapping Page.

Move the pointer to the appropriate module and hold down the right mouse button.

The Mouse Toolbox will appear. The lower three tools are displayed in black, ie they can now be used.

Move the mouse until the Copy To MP Tool is highlighted (displayed in inverse form) and release the right mouse button.

The sound will then be copied to the memory location on the MP that you defined as the destination sample when you called the SP (see the section "Synthesis Icon > Sample Icon"). If you called the SP by double-clicking the SP Icon, the sound will be copied to memory location 1 in the active bank.

Any sample previously stored at this memory location will be erased by this function.

On the Mapping Page you can either save the new sample, transfer it to a sampler or continue editing it in the TDE or FDE. Naturally, it is also possible to use the new sample as the raw material for another synthesis model...

Synthesis Page menus and dialog boxes

Desk Menu

About

This function is identical to the function of the same name in the Desk Menu on the Mapping Page.

File Menu

Load Connection

This function enables you to load a connection into memory. The current connection is erased as a result.

If you call this function when a single module is selected, it is possible to reload the parameters for that specific module. This is particularly useful in the case of modules with large numbers of parameters, eg envelopes or fractal sound generators.

Simply select the module whose parameters you wish to change and then reload a connection. The settings of the first module of the same type found within the loaded connection are then transferred to the selected module.

Merge Connection

This function enables you to load an additional connection into memory. The current connection is not erased as a result.

You can use this function to build a large structure out of several connections that were previously saved separately.

Save Connection

This function saves the current connection.

If a group of modules are selected, only these are saved as a connection. Selected modules may not be linked to unselected modules.

The settings of a single module can also be saved in this way.

This feature enables you to build up a library of standard modules that can be loaded as required using the Merge Connection or Load Connection functions described above.

Get Memory Info

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Mouse Speeder

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Screen Settings

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Preferences

This function is identical to the function of the same name in the File Menu on the Mapping Page.

Quit Synthesis

This function allows you to exit the SP and return to the Mapping Page.

Edit Menu

Open

This function is used to open a macro module. It can only be called if a macro module is selected.

The "Macros" section describes how you 'construct' a macro and install it within the Connection Display or another macro.

Info

This function displays information on the Synthesis Page and specific modules.

If a module is not currently active, selecting this option calls the dialog box containing the global settings of the SP. The parameters are:

- Name
The name of the connection, which can also be modified.
- Position of Final Sample
Here you will find the Bank Identifier and the number of the sample memory location where the synthesized sound is to be stored. These parameters can be modified directly with the mouse.
- Sampling Rate
The sampling rate for the synthesized sound.

The following 'statistical' information is also presented:

- Number of Modules
The number of modules used.
- Number of Macros
The number of macros used.
- Number of Lines
The number of links between modules.

- Free
Remaining memory (in sample values).

If only one module is selected, activating Info calls the settings dialog box for that module.

Info cannot be used if more than one module is selected.

Copy

This function copies one or more selected modules into a temporary buffer. This data can then be re-inserted into the connection using the Paste function.

Note: This memory buffer has nothing to do with the clipboard on the MP.

This function cannot be used if selected and unselected modules are linked.

Cut

This function moves one or more selected modules into a temporary buffer. This data can then be re-inserted into the connection using the Paste function.

This function can also not be used if selected and unselected modules are linked.

Erase

This function deletes the selected module(s).

In contrast to the last functions described, it is also possible to activate Erase when selected and unselected modules are linked together. If this is the case, the links to the erased modules are also deleted.

Reference

Paste

The Paste function allows you to re-insert one or more modules into the Connection Display that were previously stored in the temporary buffer using Cut or Copy. Accordingly, you can only call this function if there is already something stored in the buffer.

The modules from the buffer are inserted at the first free position in the Connection Display, beginning from the top lefthand corner.

You can paste the buffer data as often as you like. The modules stored there remain intact until they are overwritten by renewed use of the Cut or Copy functions.

Duplicate

This function duplicates one or more selected modules.

The 'doubled' module(s) appear at the first free position in the Connection Display, beginning from the top lefthand corner.

This function cannot be used if selected and unselected modules are linked.

Options Menu

Link Modules

This function transforms the standard pointer into the Link Tool. It is used to connect modules together within the Connection Display.

The rules affecting the linking of modules are described in the "Mouse Toolbox" section.

Reference

Erase Wires/Modules

This function transforms the standard pointer into the Erase Tool. It is used to delete modules and the links between modules.

The rules affecting the erasure of modules and module links are described in the "Mouse Toolbox" section.

Global Time Change

This function is used to stretch or compress the time-oriented parameters of all modules.

When you create sounds on the SP, processing times increase with the number of modules and the length of the sound. In the case of complex module structures, making adjustments to the length of all the modules can be a very time-consuming business.

This is why you should initially let the program calculate a relatively short sound (a kind of 'sketch' of the desired sound) and then multiply the sample length of all modules by a specific factor (between 0.25 and 4) using Global Time Change.

Using the Correct Waves/Envelopes button, it is also possible to stretch or compress envelopes, wavetables and waveshape tables.

Erase Synth. Samples

This function deletes the samples of all modules (but not the modules themselves).

When sounds are synthesized, a full length sample has to be calculated for each module. The amount of memory used can soon become very large. This function thus allows you to erase the synthesized samples from memory. The connection in the Connection Display itself is not affected by this.

Reference

Erase Connection

This function deletes all modules and clears the Connection Display.

You are asked for confirmation before this function is executed.

Auto Sample Erase

Activating this mode means that the synthesized samples for all modules (but not the modules themselves) are automatically erased when you leave the SP.

This enables you to regain the memory used to create sounds on the SP. The connection itself is not affected.

Reference

Software specifications

MP features

Format conversion All formats from 8 bit to 16 bit

Sample memory Computer RAM Sample slots 32, clipboard

Disc functions Load Sample, Save Sample, Load Bank, Save Bank, M.Only Bank Save, Disk Scan Load, SD Import/Export, Save Settings, Module Startup Load

TDE features

Windows Up to 8 sample windows (custom windows) Open, Close, Zoom, Next, Redraw, Swap, Tile, Stack, Cache, Auto Scrolling, Auto +- Correction, Auto Y-Zooming, Vertical Lines, Link Samples, Fill Spaces, Draw All Samples, Overview Mode, Full Wave, Draw Resolution, Zoom = Block, Block = Sample, Mark Block, Hide Block Marks, Set Markers, X-axis scaling (Realtime/ Samples), Y-axis scaling (per cent/ value)t, Show X-Position, Show Scaling, Show Points, Show Markers, Zoom Factor Display, Memory Display

Playback Play Whole Sample, Play Marked Block, Play Zoomed Part, Detect Pitch,

Scaling Fade In, Fade Out, Optimize, Reverse, Truncate, Resample, Re-Envelope, Draw Mode

Filter types Lowpass, Lowpass Shelving, Highpass, Highpass Shelving, Peak, Notch

Reference

Block functions	Cut, Copy, Erase, Fade In, Reverse, Fade Out, Insert, Overwrite, Negate, Add, Subtract, Optimize, Replicate, Delay, Eliminate DC, Keep, Mute, Filter, Analyze, Time Correction, Play
Stereo functions	Edit Left, Edit Right, Erase Left, Erase Right, Compare Left And Right, Copy Left To Right, Copy Right To Left.
Loop functions	Set Mode, Split Mode, Find, Crossfade (Single, Double, Reverse)

Undo Last Edit

FDE features

Display Modes	3D, Spektogram, Frequency Slice, Time Slice (linear or logarithmic scaling), Pixel Level, Memorize Viewpoint, Recall Viewpoint, Rescale Display
Playback	Play Sample, Play Spectrum
Area functions	Move, Copy, Add, Subtract, Reverse, Erase,
Rubberband functions	Define, Reset, Use, Copy F>T, Copy T>F
Macros	Spectral Animation, Time Plexer, Spectral Dynamic, Enhancer, 3D Filter, Harmonic Edit, Pitch Shifter, Frequency Mixer, Spectral Shaper, Threshold, Influence, Peak Limiter, Frequency Selector
Disc functions	Load Spectrum, Save Spectrum, Load F-Rubber, Save F-Rubber, Load T-Rubber, Save T-Rubber

Professional Analysis/ Resynthesis

Time Correction
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Reference

SP features

Synthesis techniques	Fourier, Wavetable, Frequency Modulation, Amplitude Modulation, Ring Modulation, Time Variant Waveshaping, Phase Distortion, Karplus/Strong, Fractal Synthesis, Subtractive Synthesis, (Digital Controlled Filter & Parametric Equalizer), Envelopping
Other modules	Mapping Page Sample, Delay, Pitch Shifter, Digital Controlled Amplifier, Mixer, Macro
Edit functions	Open, Info, Copy, Cut, Erase, Paste, Duplicate, Global Time Change, Erase Synthesized Samples, Erase Connection, Auto Sample Erase
Mouse Toolbox	Standard cursor, Link, Erase, Play, Copy to MP, Copy to MP & Call FDE
Disc functions	Load Connection, Merge Connection, Save Connection

Global functions

Key codes	for all relevant functions
Additional functions	Memory Info, Mouse Speeder, Screen Settings, MIDI Keyboard, Status Messages, Extra Help

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Hardware specifications

Computer models	ATARI ST 520 ST+ 1040 ST 1040 STE 1040 STF MEGA ST 1 MEGA ST 2 MEGA ST 4 STACY
Monitor	s/w ab 640 x 400 Pixel eg ATARI SM 124, SM 194, MATRIX 110
Storage media	All storage media for the ATARI ST computer
Options	16 Bit D/A Board SCSI Controller SMP 24 SMP 24 II MIDEX MIDEX+
Sampler (MIDI)	MMA STANDARD, AKAI S700, S900, S950, S1000, S1100, CASIO FZ-1/2, DYNACORD ADS, ADD-2, EMU EMAX, EMAX II, ENSONIQ EPS, EPS 16+, PROPHET 2000, ROLAND S330, S50, S550, YAMAHA TX 16W
Sampler (SCSI)	AKAI S1000, S1100, DYNACORD ADS, ADD-2,

Use of keys

Mapping Page

[F1]-[F8]	Play Samples 1 - 8
[F9]-[F10]	Play Clipboard Sample
[ALTERNATE][A]	About
[ALTERNATE][B]	Memory Only Bank Save
[ALTERNATE][G]	Invisible Grid
[ALTERNATE][L]	Disc Scan Load
[ALTERNATE][P]	Lock Positions
[ALTERNATE][S]	Show/ Hide 'STEINBERG'
[ALTERNATE][X]	Import/Export
[CONTROL][Q]	Quit Avalon
[CONTROL][1]	Switch Samples 1&2, 3&4, 5&6, 7&8 to Mono/ Stereo Mode
[CONTROL][4]	Switch Clipboard to Mono/ Stereo Mode
[CONTROL][5]	Get Memory Info
[CONTROL][G]	MIDI Keyboard
[CONTROL][K]	Mouse Speeder
[CONTROL][M]	Preferences
[CONTROL][P]	Screen Settings
[CONTROL][S]	Save Settings
[CONTROL][Y]	

Time Domain Editor

[0]	Full Wave Display
[1]-[8]	Close/Open Sample Windows 1-8
[B]	Block Mode on/off
[D]	Draw Resolution (1:1)
[E]	Re-Envelope Sample
[F]	Filter Sample
[H]	Hide Block Marks
[O]	Optimize Sample
[Q]	Quit Editor
[R]	Reverse Sample
[T]	Truncate Sample
[X]	Zoom = Block
[Z]	Zoom Mode on/off
[=]	Block = Sample
[?]	Sample Info
[<]	Fade In Sample
[>]	Fade Out Sample
[Cursor Down]	Zoom In on Y-axis
[Cursor Up]	Zoom Out on Y-axis
[F1]-[F8]	Goto Marker (1-8)
[SPACE]	Play Sample
[TAB]	Overview Mode on/off
[UNDO]	Undo Last Edit

[ALTERNATE][A]	About
[ALTERNATE][K]	Auto-Y-Zooming
[ALTERNATE][M]	Show Markers
[ALTERNATE][N]	Show Scaling
[ALTERNATE][P]	Show Points
[ALTERNATE][S]	Zero X-Snap
[ALTERNATE][T]	Show X-Position
[ALTERNATE][U]	Clear Undo Buffer
[ALTERNATE][V]	Clear Clipboard
[ALTERNATE][X]	X-Axis Scaling
[ALTERNATE][Y]	Y-Axis Scaling
[ALTERNATE][Z]	Y-Zoom

[CONTROL][1]	Open Windows
[CONTROL][2]	Close Window
[CONTROL][3]	Zoom Window
[CONTROL][4]	Next Window
[CONTROL][5]	Redraw Window
[CONTROL][6]	Swap Window
[CONTROL][7]	Tile Windows
[CONTROL][8]	Stack Windows
[CONTROL][9]	Window Cache
[CONTROL][A]	Analyze Block
[CONTROL][B]	Overwrite Block
[CONTROL][C]	Copy Block
[CONTROL][D]	Detect Pitch
[CONTROL][E]	Erase Block
[CONTROL][F]	Filter Block
[CONTROL][G]	Get Memory Info
[CONTROL][K]	MIDI Keyboard
[CONTROL][M]	Mouse Speeder
[CONTROL][N]	Negate Block
[CONTROL][O]	Optimize Block
[CONTROL][P]	Preferences
[CONTROL][Q]	Quit Editor
[CONTROL][R]	Reverse Block
[CONTROL][S]	Screen Settings
[CONTROL][T]	Time Correction
[CONTROL][U]	Erase FD-Editor
[CONTROL][V]	Insert Block
[CONTROL][W]	Call FD-Editor
[CONTROL][X]	Cut Block
[CONTROL][<]	Fade In Block
[CONTROL][>]	Fade Out Block
[CONTROL][+]	Add Block
[CONTROL][-]	Subtract Block
[CONTROL][.]	Erase Time-Correction

Reference

[SHIFT][*]	Resample Mode
[SHIFT][?]	Clipboard Info
[SHIFT][A]	Add Mode
[SHIFT][D]	Delay Mode
[SHIFT][E]	Erase/ Cut Mode
[SHIFT][F]	Filter Mode
[SHIFT][L]	Loop Mode
[SHIFT][O]	Optimize Mode
[SHIFT][R]	Replicate Mode
[SHIFT][Y]	Fade Mode
[SHIFT][CONTROL][T]	Block Time Correction

Frequency Domain Editor

[ALTERNATE][A]	About
[ALTERNATE][B]	Move Area
[ALTERNATE][C]	Copy Area
[ALTERNATE][D]	Add Area
[ALTERNATE][E]	Subtract Area
[ALTERNATE][F]	Reset F-Rubber
[ALTERNATE][G]	3D Graphic Display
[ALTERNATE][H]	Spectrogram
[ALTERNATE][I]	No Graphic Display
[ALTERNATE][K]	Peak Picker
[ALTERNATE][M]	Memorize Viewpoint
[ALTERNATE][N]	Rescale Display
[ALTERNATE][P]	Pitch Shifter
[ALTERNATE][R]	Recall Viewpoint
[ALTERNATE][S]	Spectral Shaper
[ALTERNATE][T]	Reset T-Rubber
[ALTERNATE][X]	2D Lin. Scaling
[ALTERNATE][Y]	2D Log. Scaling

[CONTROL][A]	Spectral Animation
[CONTROL][B]	Copy T > F-Rubber
[CONTROL][C]	Time Plexer

Reference

[CONTROL][D]	Spectral Dynamic
[CONTROL][E]	Enhancer
[CONTROL][F]	3D Filter
[CONTROL][G]	Get Memory Info
[CONTROL][H]	Harmonic Edit
[CONTROL][I]	Load T-Rubber
[CONTROL][J]	Save T-Rubber
[CONTROL][L]	Load Spectrum
[CONTROL][M]	Mouse Speeder
[CONTROL][N]	Load F-Rubber
[CONTROL][O]	Save F-Rubber
[CONTROL][P]	Preferences
[CONTROL][Q]	Quit Editor
[CONTROL][R]	Frequency Mixer
[CONTROL][S]	Screen Settings
[CONTROL][T]	Threshold
[CONTROL][U]	2D Use Rubberband
[CONTROL][V]	Copy F > T-Rubber
[CONTROL][W]	Save Spectrum
[CONTROL][X]	Influence
[CONTROL][Y]	Peak Limiter
[CONTROL][Z]	Frequency Selector

Synthesis Page

[ALTERNATE][A]
[ALTERNATE][E]
[ALTERNATE][L]
[ALTERNATE][S]

[CONTROL][A]
[CONTROL][C]
[CONTROL][D]
[CONTROL][E]
[CONTROL][G]
[CONTROL][I]
[CONTROL][L]
[CONTROL][M]
[CONTROL][N]
[CONTROL][O]
[CONTROL][P]
[CONTROL][Q]
[CONTROL][R]
[CONTROL][S]
[CONTROL][T]
[CONTROL][V]
[CONTROL][W]
[CONTROL][X]

About
Erase Wires/Modules
Link Modules
Auto Sample Erase

Erase Synth.Samples
Copy
Duplicate
Erase
Get Memory Info
Info
Load Connection
Mouse Speeder
Erase Connection
Open
Preferences
Quit Synthesis
Merge Connection
Screen Settings
Global Time Change
Paste
Save Connection
Cut

M-ROS

M-ROS Time Manager functions enable you to transfer status messages and commands between different programs and computers. AVALON can also be used to control time-oriented **M-ROS** programs like Cubase. Below you will find the key definitions which are available for this purpose. The keys used (on the numeric keypad) are identical for all **M-ROS** programs and all AVALON program levels.

[ENTER]	Play/ Continue	
[0] oder	1st time	- Stop
[LEERTASTE]	2nd time	- To Left Locator/To Position 1.1.0
	3rd time	- To Position 1.1.0
[()]	Rewind	
[)]	Fast Forward	
[1]	To Left Locator	
[2]	To Right Locator	
[SHIFT][1]	Store Song Position as Left Locator	
[SHIFT][2]	Store Song Position as Right Locator	
[9]	To Last Song Position	
[3]-[8]	To Cue Point 1 - 6	
[SHIFT] + [3]-[8]	Store Song Position as Cue Point 1 - 6	
[/]	Cycle on/ off	

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Appendix

In future versions of AVALON this section of the manual will contain additional explanations and data which do not readily fit under other headings. In the present version it provides a list of the samplers supported together with the relevant information needed to use the program with that sampler.

MIDI sampler settings

General

When it comes down to it, samplers are simply specialized computers for recording, processing, editing and playing back audio data - and should thus be ideally suited for communication with 'ordinary' computers. Unfortunately, however, they often behave rather stubbornly when they have to exchange data. They can refuse attempts at data transfer by issuing error messages and crashing or simply ignoring the relevant commands.

Clearly we make every effort to improve and extend the number of samplers and sampler features supported by our software. Nevertheless, it can occur that your sampler does not correspond with the machine we used to develop the relevant driver software. There can be a variety of reasons for this: undocumented production changes, new or outdated software and Sysex specifications, defects or improvements in design.

If communication between sampler and computer is not possible, we will naturally endeavour to eliminate the problem as soon as possible. If this occurs, please send us an exact description of your setup. What we particularly need to know here are

- Software version
- Serial no. and date of purchase
- Memory expansion or other hardware additions (if applicable)

The Sample Transfer Dialog Box

Send Sample Data

ATARI SY

Sample Name: CLOCK1
 Place : A/1
 Format: 16 Bit
 Length: ---109323 Samples
 Free : ---145210 Samples

SAMPLER

Device : Roland S-550
 Format : [12] Bit
 Tone : [I11]

TRANSMISSION

☐ Stereo Mode: Left Right
 MIDI Channel: 9
 MIDI Output : SMP 24 4
 MIDI Input : ATARI

SAMPLE DATA FLOW:

[RESET DEVICE] [CANCEL] [START]

The Sample Transfer dialog box appears on the screen when you transmit or receive samples. It is the only part of whose appearance changes according to the sampler currently being used.

The Sample Transfer dialog box is subdivided into three areas.

On the top left we will see information on the sample location in the computer. This data cannot be modified.

- Sample Name
- Place
The bank and sample memory location.
- Format
Resolution in bits.
- Length
In sample values.

Appendix

- Free
Remaining computer memory for samples.

When you want to receive a sample, only the destination bank and sample location is displayed.

On the top right you will find information on the sample memory location in the sampler.

- Device The name of the sampler.
- Format Resolution in bits.
For some samplers it is possible to set the resolution manually.

The appearance of the lower part of the dialog box depends on the sampler's specifications.

Note: Although for some inexplicable reason most manufacturers always seem to give the various structures and program levels of their samplers different names ('sample', 'tone', 'program', 'patch', 'layer', 'wave-sample', 'bank', 'voice', 'keygroup', etc, etc...) in the final analysis these structures are usually very similar. You will soon be able to see this for yourself in the Sample Transfer dialog. From sampler to sampler generally only two lines change - namely those with the 'lower' and 'higher' memory level of the piece of equipment concerned.

The lower part of the Sample Transfer dialog box displays information on the actual data transfer process between the sampler and computer.

- Stereo Mode
If you transmit or receive a stereo sample, this button is automatically activated. In such an event AVALON transmits or receives two samples, one after the other.
- MIDI Channel
The MIDI channel set for this sampler.
- MIDI Output
The MIDI port to which the sampler is connected.

Appendix

- MIDI Input
The MIDI In port of the computer to which the sampler's MIDI Out port is connected.
- Transmission
During data transfer a grey bar appears here that indicates how long the process will take. The transfer is complete when the grey bar reaches the right edge of the display.

The Reset Device button also allows you to initialize a connected MIDI Expander if this should become necessary, eg due to a temporary break.

MMA STANDARD

Transfer

The sample is transferred to the selected memory location number.

Stereo Transfer

The left sample is transferred to the selected memory location, the right sample to the next higher memory location.

Special features

If your sampler is not explicitly listed below, it is highly likely that data transfer can be carried out on the basis of the MMA standard. You will find more exact information in the MIDI section of your sampler manual.

AKAI S700

Transfer

If a sample is transmitted to a sampler memory location that is already occupied, the sample stored there is overwritten by the one sent from the computer.

If you select an unoccupied sampler memory location, the new sample is added to the existing ones.

Stereo Transfer

When stereo samples are transferred, the two samples are transmitted one after the other. In addition to this, a new program is created. This program contains two keygroups with full keyboard range - one allocated to the left, the other to the right output.

Special features

None.

AKAI S900

Transfer

If a sample is transmitted to a sampler memory location that is already occupied, the sample stored there is overwritten by the one sent from the computer.

If you select an unoccupied sampler memory location, the new sample is added to the existing ones.

Stereo Transfer

When stereo samples are transferred, the two samples are transmitted one after the other. In addition to this, a new program is created. This program contains two keygroups with full keyboard range - one allocated to the left, the other to the right output.

Special features

The sampler's directory can be displayed.

AKAI S950

Transfer

If a sample is transmitted to a sampler memory location that is already occupied, the sample stored there is overwritten by the one sent from the computer.

If you select an unoccupied sampler memory location, the new sample is added to the existing ones.

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Stereo Transfer

When stereo samples are transferred, the two samples are transmitted one after the other. In addition to this, a new program is created. This program contains two keygroups with full keyboard range - one allocated to the left, the other to the right output.

Special features

The sampler's directory can be displayed.

AKAI S1000

Transfer

If a sample is transmitted to a sampler memory location that is already occupied, the sample stored there is overwritten by the one sent from the computer.

If you select an unoccupied sampler memory location, the new sample is added to the existing ones.

Stereo Transfer

The two samples are transferred one after the other and also receive an '-L' or '-R' as channel identification.

Special features

AVALON can read the directory of this sampler. Samples can thus also be identified by their names.

AKAI S1100

Transfer

If a sample is transmitted to a sampler memory location that is already occupied, the sample stored there is overwritten by the one sent from the computer.

If you select an unoccupied sampler memory location, the new sample is added to the existing ones.

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Appendix

Stereo Transfer

The two samples are transferred one after the other and also receive an 'L' or 'R' as channel identification.

Special features

AVALON can read the directory of this sampler. Samples can thus also be identified by their names.

CASIO FZ 1/2

Transfer

You can select the Voice and Bank to which the sample should be transmitted.

Stereo Transfer

The left sample is sent to the selected Voice, the right sample to the next higher one. The two Voices are assigned to the two stereo outputs.

Special features

None.

DYNACORD ADS

Transfer

Dialog boxes will instruct you to switch between the MIX Page and MIDI Page, Sub-Page 5, on the ADS' front panel.

Correct data transfer can only take place if this is done.

Stereo Transfer

The two samples are transferred one after the other.

Special features

You will require Version 2.0 of the ADS operating system. It can be installed by any authorized DYNACORD dealer.

Appendix

AVALON can read the directory of this sampler. Samples can thus also be identified by their names.

However, when transmitting data to the ADS it is not possible to transfer the sample's name. Samples received by the ADS via MIDI are consecutively numbered as 'NEW 01', 'NEW 02,' etc.

Occasionally the ADS' memory management goes haywire during sample reception. If this occurs, switch over to another Mix, call Sub-Page 2 of the SAMPLE Page, turn the control wheel until the received sample is displayed and switch back to the original page.

DYNACORD ADD-2

Transfer

Dialog boxes will instruct you to switch between the MIX Page and MIDI Page, Sub-Page 5, on the ADD-2's front panel.

Correct data transfer can only take place if this is done.

Stereo Transfer

The two samples are transferred one after the other.

Special features

You will require Version 2.0 of the ADD-2 operating system. It can be installed by any authorized DYNACORD dealer.

AVALON can read the directory of this sampler. Samples can thus also be identified by their names.

However, when transmitting data to the ADD-2 it is not possible to transfer the sample's name. Samples received by the ADD-2 via MIDI are consecutively numbered as 'NEW 01', 'NEW 02,' etc.

Occasionally the ADD-2's memory management goes haywire during sample reception. If this occurs, switch over to another Mix, call Sub-Page 2 of the SAMPLE Page, turn the control wheel until the received sample is displayed and switch back to the original page.

EMU EMAX

Transfer

You are able to set the Key and the Level to which the sample is allocated. A new Preset (no. 99) is created.

Stereo Transfer

The left sample is sent to the selected Level, the righthand sample to the other.

Special features

None.

EMU EMAX II

Transfer

You are able to set the Key and the Level to which the sample is allocated. A new Preset (no. 99) is created.

Stereo Transfer

The left sample is sent to the selected Level, the righthand sample to the other.

Special features

None.

ENSONIQ EPS

Transfer

You can select Instrument, Layer and Wavesample; Instrument and Layer must already be set up in the sampler.

Stereo Transfer

The selected Layer and Wavesample positions are used for the left sample. For the right sample the selected Layer and Wavesample numbers are increased by one.

Special features

The 'System Exclusive' parameter on the EPS must be set to ON.

ENSONIQ EPS 16+

Transfer

You can select Instrument, Layer and Wavesample; Instrument and Layer must already be set up in the sampler.

Stereo Transfer

The selected Layer and Wavesample positions are used for the left sample. For the right sample the selected Layer and Wavesample numbers are increased by one.

Special features

The 'System Exclusive' parameter on the EPS must be set to ON.

PROPHET 2000

Transfer

The sample is transferred to the selected memory location number.

Stereo Transfer

The left sample is transferred to the selected memory location, the right sample to the next higher location.

Special features

None.

ROLAND S330

Transfer

The destination memory location must be occupied by a sample that is at least as large as the sample to be transferred.

Stereo Transfer

The two samples are transferred to two consecutive memory locations.

Special features

Switch to MIDI Mode and set the 'System Exclusive' parameter to 'On'.

ROLAND S50

Transfer

The destination memory location must be occupied by a sample that is at least as large as the sample to be transferred.

Stereo Transfer

The two samples are transferred to two consecutive memory locations.

Special features

Switch to MIDI Mode and set the 'System Exclusive' parameter to 'On'.

ROLAND S550

Transfer

The destination memory location must be occupied by a sample that is at least as large as the sample to be transferred.

Stereo Transfer

The two samples are transferred to two consecutive memory locations.

Special features

Switch to MIDI Mode and set the 'System Exclusive' parameter to 'On'.

YAMAHA TX 16W

Transfer

When the TX 16W's SysEx Device ID is set to 'OMNI', AVALON's SysEx Channel for this sampler must be set to 1.

Stereo Transfer

The two samples are transferred to two consecutive memory locations.

Special features

Due to a bug in the TX 16W's operating system, reading the sampler's directory can take very long. You can cancel this process by pressing [ESC].

SCSI sampler settings

General

The following SCSI samplers are supported. As there is no difference between installing them for use with SCSI and MIDI there is no need to repeat the sampler information here. Please see the relevant MIDI section above.

AKAI S1000

AKAI S1100

DYNACORD ADS

DYNACORD ADD 2

ENSONIQ EPS